ISSN 0711-6659

canadian acoustics
acoustique canadienne

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

SEPTEMBER 2019 SEPTEMBRE 2019
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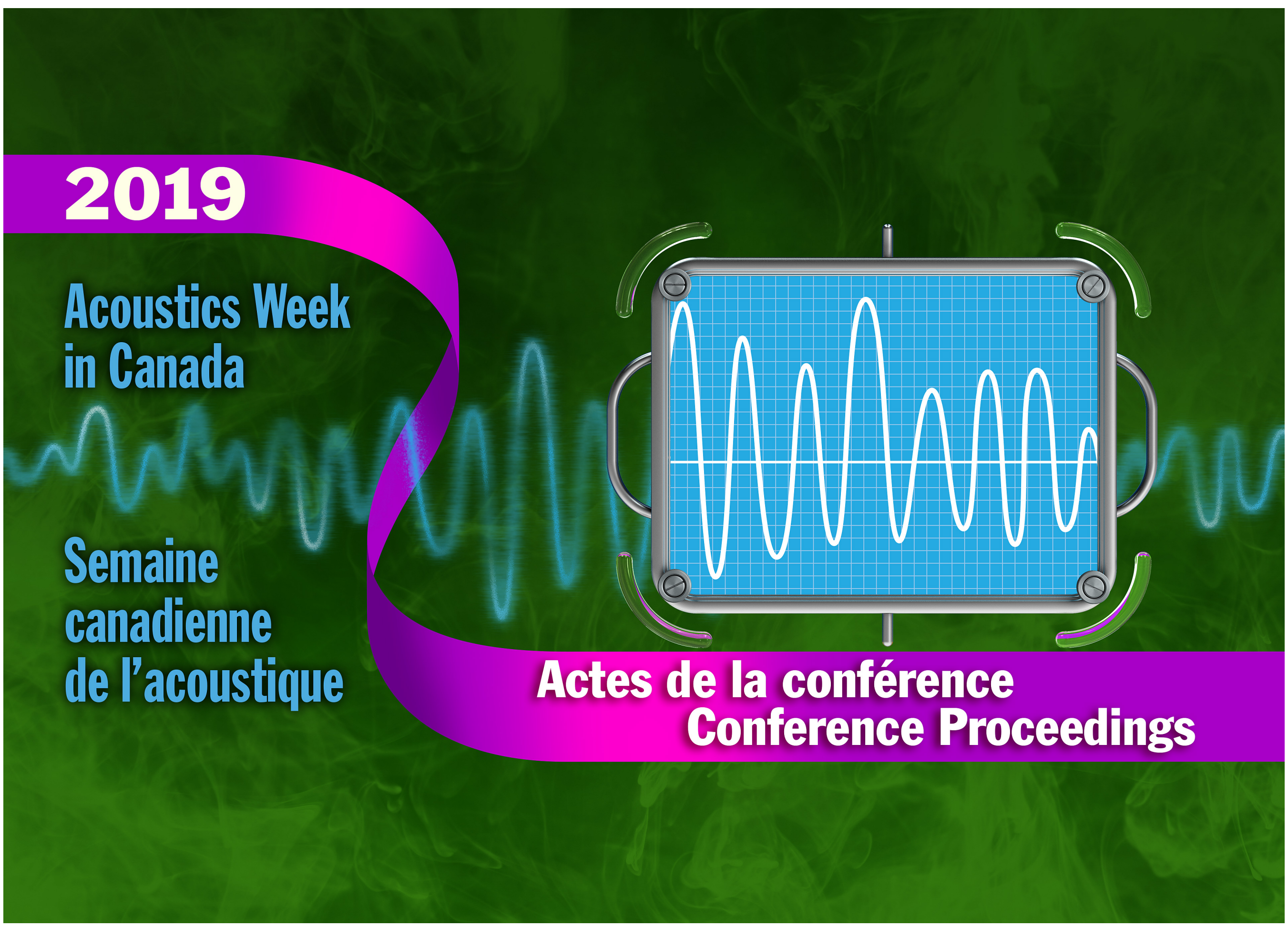

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Canadian Acoustical Association/Association Canadienne d'Acoustique P.B. 74068 Ottawa, Ontario, K1M 2H9

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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 use the application form from the CAA website 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 c/o JASCO Applied Sciences 2305–4464 Markham Street Victoria, BC V8Z 7X8 - - - secretary@caa-aca.ca - Dr. Roberto Racca

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, merci d'utiliser le formulaire d'adhésion du site de l'ACA et de retourner ce dernier avec votre chèque ou mandat au secrétaire de l'association (voir adresse ci-dessus). - Canadian Acoustical Association/Association Canadienne d'Acoustique c/o JASCO Applied Sciences 2305–4464 Markham Street Victoria, BC V8Z 7X8 - - - secretary@caa-aca.ca - Dr. Roberto Racca

CANADIAN ACOUSTICAL ASSOCIATION
ASSOCIATION CANADIENNE D'ACOUSTIQUE

ACOUSTICS WEEK IN CANADA 2019 – EDMONTON AB

Welcome to Edmonton!

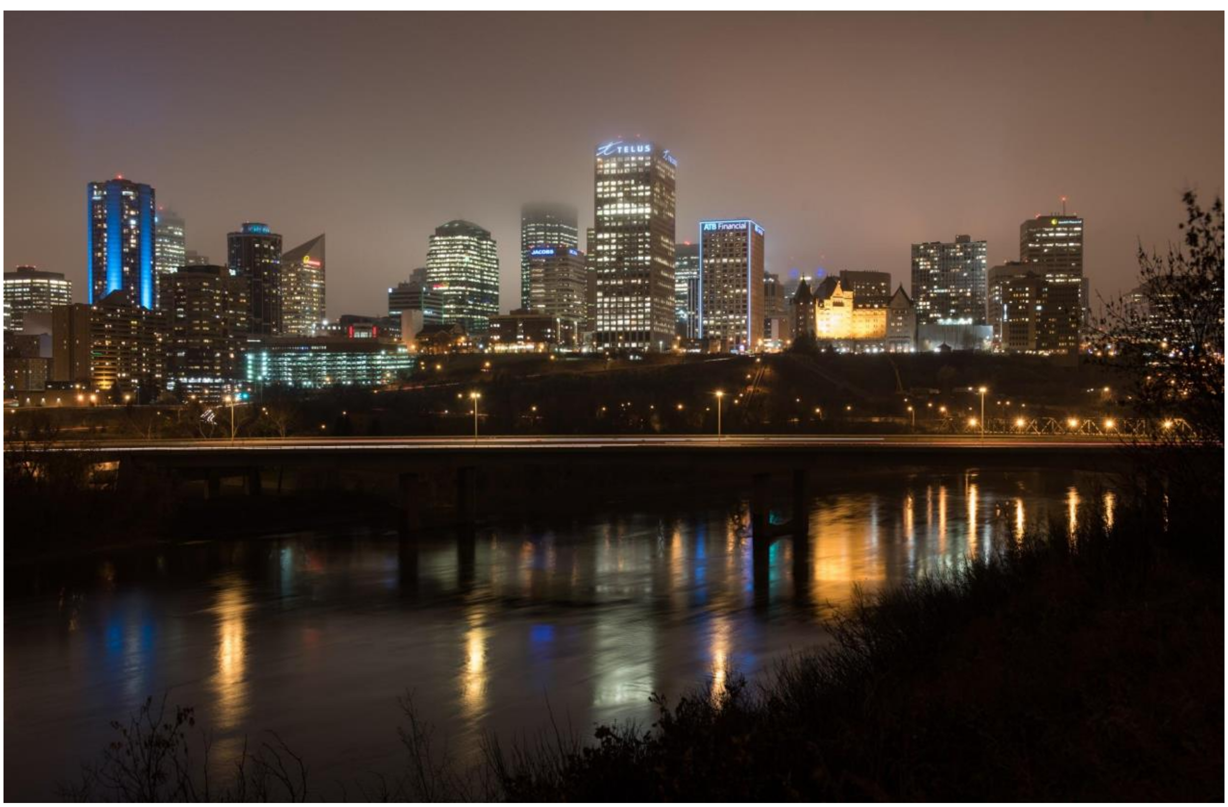
Contacts/Organizing Committee

Conference Chair:	Benjamin V. Tucker, University of Alberta
Treasurer:	Corjan Buma, University of Alberta/ACI
Technical Chairs:	Tara Vongpaisal, MacEwan University
	Daniel Aalto, University of Alberta

Exhibit/Sponsor

Coordinators:	Ellen Buchan, Alberta Government
Student Prizes and Subsidies:	Mary Ingraham, University of Alberta
Website:	Matthew Kelley, University of Alberta

Conference Website: http://awc.caa-aca.ca



Introduction

We are very happy to welcome you to Acoustics Week in Canada 2019 in Edmonton. We are grateful for the opportunity to host this year's conference. We would like to recognize that the conference will take place in Treaty Six territory, a traditional gathering place for diverse Indigenous peoples including the Cree, Blackfoot, Metis, Nakota Sioux, Iroquois, Dene, Ojibway/ Saulteaux/Anishinaabe, Inuit, and many others whose histories, languages, and cultures continue to influence our vibrant community. Acoustics Week in Canada was last held in Edmonton in 2003 and in the intervening years the city has changed dramatically and we felt it was time for the Canadian Acoustical Association to return to Edmonton. We are pleased that the conference will feature presentations from 61 researchers from around Canada with a workshop hosted by the National Research Council. In addition, 31 two-page proceedings papers were submitted, and you will find them within these conference proceedings. The papers span topics from building and room acoustics to the acoustics of indigenous languages. As an organizing committee, we would like to recognize that 2019 is the United Nations International Year of the Indigenous Language. As a celebration of the International Year of the Indigenous Language, our final plenary speaker, Sonya Bird will speak about indigenous languages. We will have two other plenary speakers: Hildegard Westerkamp will speak about soundscapes and Michelle Vigeant will speak about room acoustics.

Enjoy the conference!

Benjamin V. Tucker

SEMAINE CANADIENNE D'ACOUSTIQUE 2019 – EDMONTON AB

Bienvenue à Edmonton!

Contacts / Comité organisateur

Président :	Benjamin V. Tucker, University of Alberta
Trésorier :	Corjan Buma, University of Alberta/ACI
Directeurs scientifiques :	Tara Vongpaisal, MacEwan University
	Daniel Aalto, University of Alberta

Exposition / Commandites

Coordinateur aux commandites :	Ellen Buchan, Alberta Government
Prix étudiants et subventions :	Mary Ingraham, University of Alberta
Site de la conférence :	Matthew Kelley, Université de l'Alberta

Site web du Congrès : https://awc.caa-aca.ca/

Introduction

Nous sommes très heureux de vous accueillir à la Semaine canadienne de l'acoustique 2019 à Edmonton. Nous sommes honorés d'avoir l'occasion d'accueillir cette conférence réputée cette année. Nous aimerions souligner que la conférence aura lieu sur le territoire Treaty Six, un lieu de rassemblement traditionnel pour divers peuples autochtones, notamment les Cris, les Pieds-Noirs, les Métis, les Sioux Nakota, les Iroquois, les Dénés, les Ojibway/ Saulteaux/Anishinaabe, les Inuits et plusieurs autres, et que les histoires, langues et cultures continuent à avoir une influence sur notre communauté très active. La dernière fois que la Semaine canadienne de l'acoustique a eu lieu à Edmonton en 2003, la ville a changé radicalement et nous avons pensé qu'il était temps pour l'Association canadienne d'acoustique de retourner à Edmonton. Nous sommes enchantés que 61 chercheurs de partout au Canada y présentent des exposés et que le Conseil national de recherches du Canada (CNRC) sera l'hôte d'un atelier. De plus, 31 articles de conférence de deux pages ont été soumis, et vous les trouverez dans les actes de cette conférence. Les articles portent sur des sujets allant de l'acoustique des bâtiments et des salles à l'acoustique des langues autochtones. En tant que comité organisateur, nous aimerions souligner que 2019 est l'Année internationale des Nations Unies pour les langues autochtones. Pour célébrer l'Année internationale des langues autochtones, notre dernière conférencière plénière, Sonya Bird, parlera des langues autochtones. Nous entendrons deux autres orateurs en séance plénière : Hildegard Westerkamp parlera des paysages sonores et Michelle Vigeant de l'acoustique des salles.

Bonne conférence !

Benjamin V. Tucker

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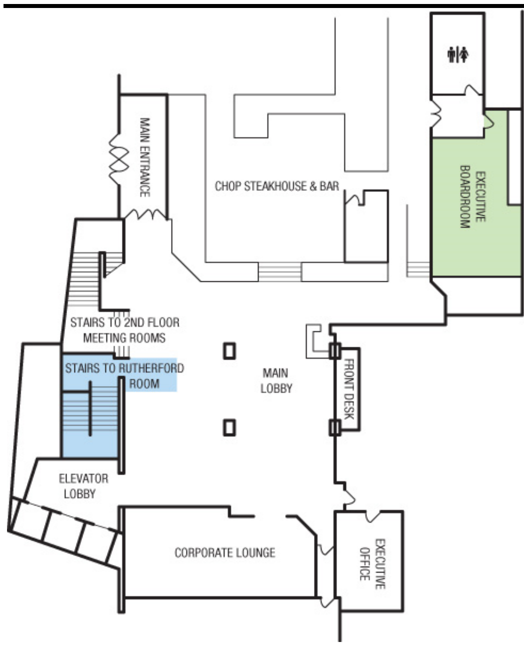
Durisol
NOISE BARRIER SYSTEMS

TUESDAY 8 OCT		
4:00-10:00	Advanced Registration (Sutton Place Hotel)	
6:30-8:30	Reception (MacEwan University - Allard Hall)	
DAY ONE	**WEDNESDAY 9 OCT (Sutton Place Hotel)**	
8:50-9:OO	Welcome	
9:00-10:00	Keynote Talk (1 of 3) - Hildegard Westerkamp (William Tomison Room)	
10:00- 10:30	COFFEE BREAK	
	Winterlake Room	**Rowand Room**
10:30-12:00	SPEECH PRODUCTION	BUILDING ACOUSTICS
12:00-1:15	LUNCH (William Tomison Room)	
1:15-2:55	SPEECH PERCEPTION AND PRODUCTION	BUILDING ACOUSTICS
2:55-3:15	COFFEE BREAK	
3:15-5:15	SPEECH PERCEPTION AND PRODUCTION	ENGINEERING AND PHYSICAL ACOUSTICS
5:30-6:30	ASTC Workshop by the National Research Council Canada on the National Building Code of Canada (NBCC)	
DAY TWO	**THURSDAY 10 OCT (Sutton Place Hotel)**	
9:00-10:00	Keynote Talk (2 of 3) - Michelle Vigeant (William Tomison Room)	
10:00- 10:20	COFFEE BREAK (Exhibits are in the foyer all day)	
10:20-11:40	SPEECH PERCEPTION AND PRODUCTION	ENGINEERING AND PHYSICAL ACOUSTICS
11:40-1:15	LUNCH (William Tomison Room)	
1:15-2:35	AUDIOLOGY	BIOACOUSTICS

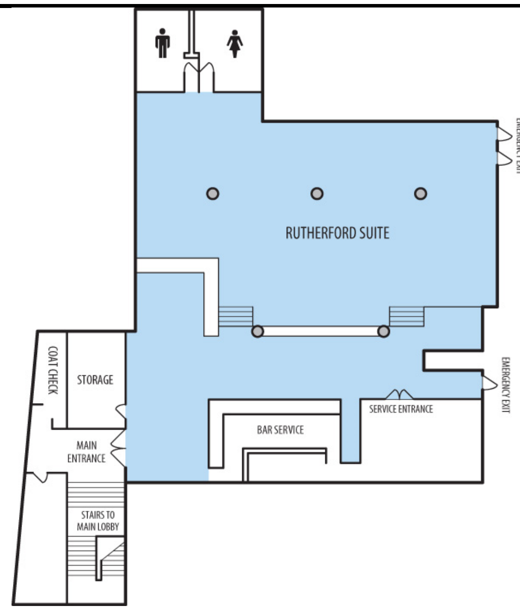
2:35-3:15	COFFEE BREAK	
3:15-4:15	SPEECH PRODUCTION	NOISE ACOUSTICS
4:30-5:30	ANNUAL GENERAL MEETING	
6:30-8:30	BANQUET (Edmonton Convention Centre)	
DAY THREE	**FRIDAY 11 OCT (Sutton Place Hotel)**	
9:20-10:20	Keynote Talk (3 of 3) - Sonya Bird (William Tomison Room)	
10:20-10:40	COFFEE BREAK	
10:40-12:00	AUDITORY PERCEPTION	NOISE ACOUSTICS
12:00-1:00	LUNCH & STUDENT PRESENTATION AWARDS (William Tomison Room)	

The Sutton Place Hotel

Main Floor



Lower Level



Second Floor

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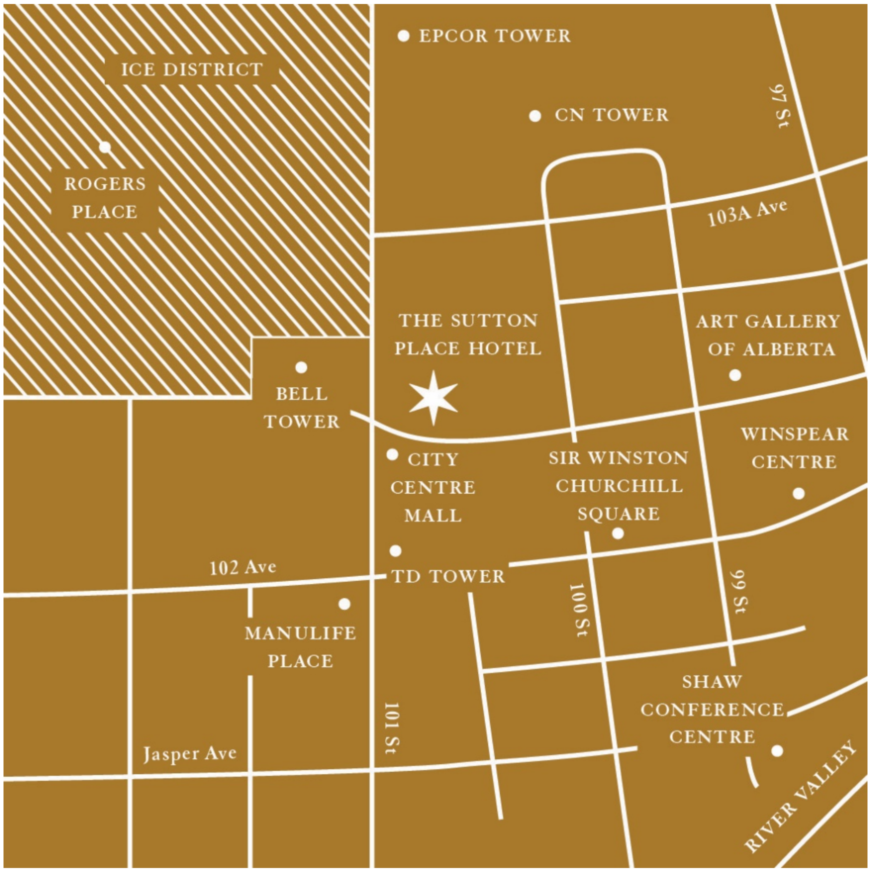
The Sutton Place Hotel

LOCATION



Airport Information:

- Edmonton International Airport: 35km – approximately 35 minutes
- Shuttle bus: Sky shuttle $18.00 one way or $30.00 return
- Taxi: Approximately $55.00, pending traffic

Parking:

Hotel parking is available at the prevailing rates of $35.00 per day for self-parking, or $45.00 per day for valet parking until 3pm the following day. Adjacent parking lots between $5-$31.

*Please note: Parking prices are subject to change without notice, as the hotel does not own the parking facilities.

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Plenary Speakers

Hildegard Westerkamp

Plenary Talk Title: **Grounded in Listening - Messages from Inside the Soundscape**

Composer Hildegard Westerkamp focuses on listening, environmental sound and acoustic ecology. At the beginning of her career she worked with R. Murray Schafer and the World Soundscape Project, is a founding and board member of the World Forum for Acoustic Ecology and was long-time editor of its journal Soundscape. She has conducted soundscape workshops, and has given concerts and lectures locally and internationally. In 2003 Vancouver New Music (VNM) invited her to coordinate and lead regular public soundwalks. This inspired the creation of The Vancouver Soundwalk Collective, whose members are now continuing the work, and are joined internationally by an astonishing number of new soundwalk initiatives. For some years now she has mentored a variety of younger composers, sound designers, soundwalk leaders and people pursuing careers in soundscape studies and acoustic ecology. Excerpts of her compositions appear in Gus van Sants' films Elephant and Last Days and more recently she collaborated on the soundtrack of Nettie Wild's film Koneline. Her newest composition Klavierklang for pianist Rachel Iwaasa had its world premiere at ISCM's World Music Days in Vancouver, November 2017. Also in 2017 Hildegard's ways of composing and listening were presented on CBC IDEAS: http://www.cbc.ca/radio/ideas/how-opening-our-ears-can-open-our-minds-hildegard-westerkamp-1.3962163

Michelle Vigeant

Plenary Talk Title: **Perceptual and Physiological Responses to Room Acoustics and Noise**

Michelle is an associate professor in the Graduate Program in Acoustics at Penn State, where she also holds a joint appointment with the Department of Architectural Engineering. She leads the Sound Perception and Room Acoustics Laboratory (SPRAL), where her research is focused on linking quantitative measures to human perception of sound. Michelle also studies the effects of noise on stress levels, task performance, and annoyance. The majority of her research studies are conducted in a dedicated, 3D virtual sound reproduction laboratory named the AUralization and Reproduction of Acoustic Sound-fields (AURAS) facility, which is housed in an anechoic chamber. Michelle has received an NSF CAREER Award and a 3M Non-tenured Faculty Award. She is currently an elected member of the Acoustical Society of America's Executive Council. Prior to joining Penn State in 2012, Dr. Vigeant was an assistant professor at the University of Hartford in the Mechanical Engineering Department within the undergraduate acoustics concentration for 5 years. She obtained her Ph.D. in Architectural Engineering, focusing on Architectural Acoustics, from the University of Nebraska-Lincoln in 2008, and she received a B.Sc. in Mechanical Engineering Co-op from the University of Alberta in 2003.

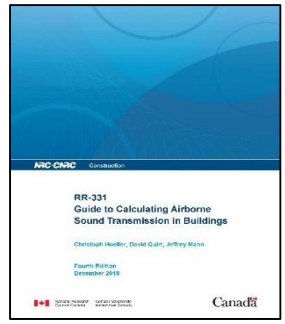
Sonya Bird

Plenary Talk Title: **Documenting the phonetic structures of Canada’s Indigenous languages**

Sonya Bird has been conducting phonetic documentation of British Columbia’s Coast Salish languages since 2002. Dr. Bird’s current research focuses on supporting adult Indigenous language learners to achieve oral proficiency and fluency, through (a) documenting pronunciation among first language speakers, (b) comparing first languages speakers’ pronunciation to that of second language speakers, in order to understand the challenges for learners, and (c) developing and assessing teaching tools/materials to overcome these challenges.

ASTC Workshop by the National Research Council Canada

In collaboration with a number of Canadian industry partners, the National Research Council Canada (NRC) has developed a number of tools to support the transition to the ASTC rating. This workshop will provide an overview of the tools that are available for the calculation of the ASTC rating to demonstrate compliance with the acoustic requirements of the code for different building constructions. The overview will be followed by question and answer session and a hands on workshop for which attendees are encouraged to bring questions and their own data to be used in the Excel spreadsheets. The Excel spreadsheets will be made available prior to the workshop and all participants are encouraged to bring their own laptops.



Ivan Sabourin

holds a Mechanical Engineering Degree from the University of Ottawa and he has been working with the Acoustic Group at the National Research Council Canada (NRC) for over 11 years. During this time, he has mainly worked on building acoustic projects: evaluating client products in floor and wall assemblies, evaluating flanking assemblies and contributing to the research of the ASTC Guide. During the last few years, Ivan has been involved in the upgrade and technical functioning of the Acoustic facilities.

Jeffrey Mahn

is a senior research officer at the National Research Council Canada specializing in flanking sound transmission in buildings. He has a masters degree from the Technical University of Denmark and a PhD from the University of Canterbury in Christchurch, New Zealand. The NRC is currently engaged is large-scale research projects relating to the transmission of structure-borne noise between dwellings. Based on the results of the research projects, the NRC has published a number of research reports to support the transition to the acoustic requirements in the National Building Code of Canada including the recently published fourth edition of RR-331 Guide to Calculating Airborne Sound Transmission in Buildings.

TUESDAY 8 OCT

Registration (MacEwan University - Allard Hall)
Reception (MacEwan University - Allard Hall)

WEDNESDAY 9 OCT (Sutton Place Hotel)

Welcome

Keynote Talk (1 of 3) - Hildegard Westerkamp (William Tomison Room)

COFFEE BREAK

Winterlake Room	Rowand Room
SPEECH PRODUCTION	BUILDING ACOUSTICS
Michael Kiefte, Terrance M. Nearey: **An Acoustic Analysis of Vowel Production in Spontaneous Speech**	Paul Marks: **Acoustical Considerations for Design-Build Mental and Behavioural Healthcare Facilitie**
Matthew C. Kelley, Daniel Aalto: **Measuring the dispersion of density in head and neck cancer patients' vowel spaces: The vowel dispersion index**	Mandy Chan, William Gastmeier: **Recent Experience with Acoustical Privacy Considerations in Academic Settings**
Arian Shamei, Sonya Bird: **Acoustic Correlates of Cannabis-Intoxicated Speech**	Kelly Kruger: **Complexities of Curtain Wall Flanking Transmission – A Case Study**
Bryan Gick, Arian Shamei: **Effects of Outer Space on Vowel Space**	Todd Busch: **Generic Third-Octave Band Spectra for Construction Equipment**
Bryce Jacob Wittrock, Benjamin V. Tucker: **Canadian prairie dialects: An exploration of Alberta and Saskatchewan vowels**	Jianhui Zhou, Behzad Vafaeian, Ying-Hei Chui: **Radiation Efficiency of Cross Laminated Timber Panel by Finite Element Modelling**

LUNCH (William Tomison Room)

SPEECH PERCEPTION AND PRODUCTION	BUILDING ACOUSTICS
Scott James Perry, Benjamin V. Tucker: **L2 Production of American English Vowels in Function Words by Spanish L1 Speakers**	Weidong Li: **Evaluation of an Indoor Open Space's Acoustical Quality – A Case Study**
Annika Nijveld, Louis ten Bosch, Mirjam Ernestus: **Dutch learners of English can identify reduced pronunciation variants in English running speech, while Spanish learners cannot**	Matthew Golden, Roderick Mackenzie: **The Case for Minimum Impact Noise Requirements in the National Building Code of Canada**
McKae Pawlenchuk, Yoichi Mukai, Benjamin V Tucker: **Reduced flaps in English processed by non-native speakers**	Matthew Golden, Wilson Byrick: **Comparison of High-Performance Floor and Ceiling Isolation Systems: Are Springs Always Better?**
Quinn Goddard, Angeliki Athanasopoulou, Darin Flynn: **Lexical Stress in Plains Cree: An Acoustic Account**	Umberto Berardi: **Sound attenuation of a wood noise barrier using a meta-material approach**
Gracellia Purnomo, Gloria Mellesmoen, Arian Shamei, Bryan Gick: **Production study of Spanish spirantization in naturalistic speech**	Ellen Buchan: **Renovation of the Northern and Southern Alberta Jubilee Auditoria: A Case Study on the Impact of Orchestra Pit Renovation on Acoustics**

COFFEE BREAK

SPEECH PERCEPTION AND PRODUCTION	ENGINEERING AND PHYSICAL ACOUSTICS
April Pereira: **Effects of Modality and Linguistic Materials on Memory in Younger and Older Adults**	Victor Krupka, Sebastian Ghinet, Eric Chen, Andrew Price, Viresh Wickramasinghe, Anant Grewal: **Investigation of Aircrew Noise Exposure Due to the Use of the Intercom System Onboard the RCAF CH-149 Helicopter**
Carolina Salinas-Marchant, Geneviève Meloni, Andrea A.N. MacLeod: **Phonetic and phonological development of consonants in Franco-Quebec children: Mixed models for fricative analysis**	Victor Krupka, Sebastian Ghinet, Viresh Wickramasinghe, Anant Grewal: **Hearing Protection Performance Evaluation of Active Noise Reduction Headsets Under High Intensity Noise Levels**
Kirsten Mulder, Alesha Reed, Benjamin V. Tucker, Carol A. Boliek: **The Effects of Transcranial Direct Current Stimulation (tDCS) on Intermuscular Coherence during Maximum Performance Tasks in Healthy Younger and Older Adults**	Maryse Lavoie: **Évaluation des nuisances sonores provenant des systèmes à traitement d'air en milieu urbain**
Stephanie L. Archer, Megan Galloway, Wenfu Bao, Hester Duffy, Sotaro Kita : **Do mothers prefer monkeys over nembees**	John Halpenny, Joana Rocha: **Minimizing sonic boom noise to meet potential regulatory limits**
Tara Vongpaisal: **Perception of Mixed Emotions from Semantic and Prosodic Cues**	Ian Matthew, Anthony Amarra: **Comparison of Results from the STEAM Rail Noise Model to Potenti Alternatives**
	Mark Salsberg: **Sound insulation and Absorption in Rail Applications**

ASTC Workshop by the National Research Council Canada on the National Building Code of Canada (NBCC)

THURSDAY 10 OCT (Sutton Place Hotel)

Keynote Talk (2 of 3) - Michelle Vigeant (William Tomison Room)

COFFEE BREAK

Exhibits all day in the Foyer

SPEECH PERCEPTION AND PRODUCTION	ENGINEERING AND PHYSICAL ACOUSTICS
Filip Nenadic, Matthew C. Kelley, Ryan G. Podlubny, Benjamin V. Tucker: **Speech perception and the role of semnatic richness in processing**	Todd Busch: **Design Of A Test Apparatus For Measuring The Low-Amplitude Vibration Sensitivity Of Large Test Articles**
Terrance Nearey, Benjamin V. Tucker: **Modeling categorization of a VCCV continuum in the presence of response lapses and dropouts**	Basim Al Tlua, Joana Rocha: **Development and Testing of an Aeroacoustic Wind Tunnel Test Section**
Yadong Liu, Pramit Saha, Bryan Gick, Sidney Fels: **Deep learning based continuous vowel space mapping from hand gesture**	Ian Bonsma, Nathan Gara, Brian Howe: **Intentional yaw Misalignment and the Effects on Wind Turbine Noise**
Dallin A Backstrom, Benjamin V Tucker, Matthew C Kelley: **Forced-alignment of the sung acoustic signal using deep neural nets**	Weidong Li: **A Comparative Evaluation of Hand-Arm Vibration Impacts**

LUNCH (William Tomison Room)	
AUDIOLOGY	BIOACOUSTICS
Kristen Elizebeth Mulderrig: **Deaf Cultural Identification, Cochlear Implants, and Life Satisfaction**	Erin Bayne: **Wiretapping the wilderness: New approaches to wildlife monitoring using autonomous recording units**
Jeremie Voix: **Hearing Protectors Fit-Testing: a Smartphone-Based Approach**	Richard William Hedley: **Distance truncation via relative sound level for surveys of birds in patchy habitat**
Alex Gascon, William Hodgetts: **Which Threshold Do We Trust? A Comparison of Threshold Measurements In Adult Bone Conduction Users**	Elly Caitlin Knight, Erin M Bayne: **Automated acoustic mark-recapture of a migratory bird using sign recognition software**
Farid Moshgelani: **On Speech Envelope Enhancement for Hearing Aid Applications**	Kimberley Ann Campbell, Stephanie Thunberg, Christopher B Sturdy: **Features of male- and female-produced song in black-capped chickadees (Poecile atricapillus) change between seasons**
COFFEE BREAK	
SPEECH PRODUCTION	NOISE ACOUSTICS
Noor Hisham Al-Zanoon, Daniel Aalto: **Investigation of tongue biomechanics in English vowels (/i/,/æ/) using virtual computer simulation**	Andy Metelka: **Wind Turbine noise relationships of tonality, infrasound, amplitude modulation and audible noise**
Terrina W Chan, Grace Purnomo, Yadong Liu, Bryan Gick: **Coarticulation in speech and smile movements: An electromyographic and video analysis**	Peter VanDelden: **The Significance of Sampling Rate in Firearms Sound Exposure Determination**
Himanshu Goyal, Pramit Saha, Bryan Gick, Sidney Fels: **EEG-to-F0: Establishing artificial neuro-muscular pathway for kinematics-based fundamental frequency control**	Andy Metelka: **Alternative measurement techniques for quantifying Motorcycle noise**
ANNUAL GENERAL MEETING	
BANQUET (Edmonton Convention Centre)	
FRIDAY 11 OCT (Sutton Place Hotel)	
Keynote Talk (3 of 3) - Sonya Bird (William Tomison Room)	
COFFEE BREAK	
AUDITORY PERCEPTION	NOISE ACOUSTICS
Alberto Behar, Joseph Rovetti, Mohammad Abdoli, Fran Copelli, Frank Russo: **Listening Effort in Eateries**	Corjan Buma: **Overview of recent project work at the MEANU**
Cory McKenzie, Amberley Ostevik, Bill Hodgetts, Jacqueline Cummine, Daniel Aalto: **Associations Between Musical Experience and Auditory Discrimination**	Mark Bliss: **Developing the Port of Vancouver's Port Noise Rating Methodology**
Katherine Marcoux, Benjamin V. Tucker, Mirjam Ernestus: **Non-native Lombard speech intelligibility for native and non-native listeners**	Amir A. Iravani, Lucas Arnold: **Innovative and Feasible Noise Mitigation Planning**
William Hodgetts, Daniel Aalto, Jacqueline Cummine: **Is Apple's Virtual Assistant Siri Familiar Enough to Lead to Better Understanding and Easier Listening Than an Unfamiliar Voice?**	Jean-Philippe Migneron, Jean-François Hardy, André Potvin, Jean-Gabriel Migneron: **Studying noise assessment and policies to influence noise management in Quebec**
LUNCH & STUDENT PRESENTATION AWARDS (William Tomison Room)	

AUDIOLOGY - AUDIOLOGIE

Deaf Cultural Identification, Cochlear Implants, And Life Satisfaction
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DEAF CULTURAL IDENTIFICATION, COCHLEAR IMPLANTS, AND LIFE SATISFACTION

Kristen Mulderrig *[1], **Dr. Sean Rogers** †[2]
MacEwan University

1 Introduction

Cultural identification within the Deaf community has 4 distinct subgroups: deaf (immersion), hearing, bicultural, and marginal [1, 2]. Each of these labels represent a specific internal stance about an individual's Deaf identity, and how they impact their overall well-being [1, 3]. Culturally deaf describes a profound, positive stance about their deafness, hearing describes the typical stance that the hearing world shares about Deaf culture, marginal describes a shifting loyalty between the hearing and Deaf worlds and culturally bicultural describes those who have integrated their Deaf pride in a way that is balanced with the hearing world [1].

This leads to wondering where Cochlear Implants (CI) users fit into the model for Deaf acculturation. Since CI's are electrical devices that create artificial sound processing, their perception of auditory signals is at a substantially lower rate than normal hearing (NH) individuals and an overall lower life satisfaction has been found [4, 5].

The present study seeks to address the problem of lower life satisfaction found in the Deaf community and seeks to identify how CI users fit into the community. The researchers hypothesized that Deaf individuals who identify with both the hearing and Deaf communities, will have the highest overall life satisfaction. Individuals identifying with either the Deaf or hearing community, will have high overall life satisfaction, however, individuals not identifying with either communities will have the lowest life satisfaction. In addition, Deaf individuals who have CI's will have higher overall life satisfaction than those without, and even higher life satisfaction with strong cultural identification.

2 Method

2.1 Participants

Thirty-three participants responded to the online survey, however, 20 scorable participants were used. The participants were recruited from the Edmonton area Deaf community through: The Connect Society, Facebook pages, and MacEwan University. The participants were 40% males and 60% females. Of those, 80% were 19 or older and 70% of the respondents reported English as their first language. As well, 50% indicated they were born deaf, 25% said they have CI's, and 95% reported they are Deaf or hard of hearing.

2.2 Measures

A total of 3 scales were used in this study: the Deaf Identity Development Scale (DIDS), the Deaf Acculturation Scale (DAS), and the Satisfaction with Life Scale (SWLS). The DIDS asked questions revolving around one's preference for Deaf or hearing culture and resulted in the participant being categorized into either immersion, hearing, bicultural, or marginal cultural identification. The DAS is an extensive revision of the DIDS, asking questions centered around activities and preferences. Lastly, The SWLS asks questions about how the participant feels about their life.

2.3 Design

An online survey was used to examine how CI's impact cultural identification and life satisfaction. The predictor variables were whether the participant has CI's or not and what their cultural identification looked like. The outcome variable was the degree of life satisfaction reported.

2.4 Procedure

The study took place online where participants found a link to the study and began the survey. Once consent was given, they agreed to participate and completed personal demographic information, they filled out the 3 sections previously described and were then debriefed. Each session took around 30 minutes to complete and no verbal instructions were given. If MacEwan students completed the study, they received the same instructions but then were granted an additional 1% to their final grade in their introductory Psychology course for participating.

3 Results

3.1 Life Satisfaction

A total sample of 20 participants (N = 20) revealed low confidence in the results. The mean life satisfaction for all groups was 26.0 (SD = 8.62), with bicultural having the highest life satisfaction at 49.05 (SD = 9.92, $p > .05$), followed by the immersion group with a life satisfaction score of 28.05 (SD = 7.619, $p > .05$), the hearing group scored 21.15 (SD = 8.216, $p > .05$) on life satisfaction, and the marginal group scored a 24.75 (SD = 8.09, $p > .05$). The correlation between life satisfaction and hearing was -.349 ($p > .05$), life satisfaction correlated with immersion at .481 ($p > .05$), marginal scored a correlation of -.699 ($p < .05$), and bicultural correlated with life satisfaction at .434 ($p > .05$).

3.2 Regression Analysis

A regression analysis indicated an R^2 of .586, an adjusted R^2 of .475, and showed the model was statistically significant F5.305(4,15), $p = 0.007$ (see Table 2). CI's were not significantly correlated with life satisfaction ($r = -.025$, $p < .05$). CI's and hearing identification were significantly related

* mulderrigk@mymacewan.ca
† RogersS10@macewan.ca

(r = .025, p < .05), immersion and CI's scored .481, marginal scored -.699 and bicultural and CI's scored at a .434.

Table 1: Correlations with Life Satisfaction

Hearing	Immersion	Bicultural	Marginal
-.349	.481	.434	-.699

The hearing group consisted of 10% of the overall participants, 5% were marginal, 85% bicultural, and no participants fell into the immersion group. Culturally marginal was the only statistically significant group with an R^2 of .488, F17.174(1,18), p = 0.01. Adding hearing identification resulted in a p-value of .067, bicultural lead to a p-value of .724, and adding deaf acculturation resulted in a p-value of .980.

Table 2: Model Summary

R	R Square	Adjusted R Square	Sig. Change	F
.765	.586	.475	.007	

4 Discussion

The present study hypothesized that Deaf individuals who identify as bicultural will have the highest life satisfaction and those who do not identify with either the Deaf or hearing community will have the lowest life satisfaction. While the data gathered failed to provide significant support for this hypothesis, it is interesting that the trends match the pattern of this hypothesis. Results could not be drawn based on CI's due to the small sample size. This study found that 62% of Deaf individuals reported neutral life satisfaction, meaning the majority of the sample are relatively happy with their life. In comparison to the NH population, a world poll indicated 74% of over a million respondents report they are happy [6]. This shows the Deaf community does have lower than normal life satisfaction and additional research should be conducted to promote greater well-being in the Deaf.

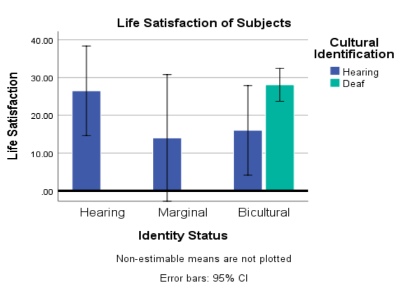

Figure 1: Life Satisfaction of Subjects

The small sample of CI users (N=5) indicated conclusions could not be drawn confidently and the study rejected the null hypothesis. Although the findings do show a trend toward the marginal group having a lower life satisfaction in relation to the implants, a larger sample size is necessary. Yet, life satisfaction did significantly correlate with cultural identification as hypothesized. Meaning that those who had a strong cultural identification had a higher overall life satisfaction than those who did not.

The total life satisfaction of the sample could vary from 5-9 (extremely dissatisfied) to 31-35 (extremely satisfied). This means the bicultural group scored the highest life satisfaction, followed by the immersion, marginal, and then hearing. This is relatively consistent with the researcher's hypothesis as they predicted that the bicultural group would have the highest life satisfaction and marginal would have the lowest. The correlation between hearing and life satisfaction was negative indicating that those who identified as hearing had a lower life satisfaction due to it. As well, marginal showed the same trend but with an even lower negative correlation.

5 Conclusion

The present study suggests that there is a link between how individuals feel overall and how strongly they identify with a culture. Because those who had stronger life satisfaction also had higher cultural identification, it can be concluded that Deaf cultural involvement has implications for overall well-being whether it be in both the Deaf and hearing communities [4, 7-8]. No correlation between CI's and cultural identification or life satisfaction was found, further research should be conducted to see if there is a relationship between these variables. Overall, this research has advanced the larger body of knowledge by showing there is a strong correlation between the Deaf community's well-being and how they fit into the Deaf or hearing communities.

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Hearing Protectors Fit-Testing Using Smartphones: Preliminary Data

Jérémie Voix *[1]

[1] École de technologie supérieure (Université du Québec), Montréal, Québec, H3C 1K3

1 Introduction

Hearing protection devices (HPD) are -too- often the first line of protection for industrial workers against the risk of noise-induced hearing loss. But the efficiency of HPD is dramatically affected by workers' ability to properly fit them and get a sufficient amount of attenuation over the total duration of their exposure to high levels of noise [1]. Fit-testing systems have been developed over the years to assist workers in fitting their HPD and to estimate the amount of attenuation offered by such HPDs [2]. These systems are often expensive, require trained personnel and end-up being used only once in a while to assess if a given worker is properly protected. A possible fast and lightweight alternative is explored here where the user can self administer a simple auditory test using a smartphone or tablet.

2 Proposed Fit-Testing Method

To quickly estimate the amount of attenuation provided by any given HPD on any given individual, a tablet/smartphone app has been developed. The app features an audio stimulus generator to generate loud tones over the tablet embedded loudspeakers, a graphical user interface where the user can report the count of audio stimuli perceived, and an attenuation prediction algorithm able to estimate the overall attenuation of the HPD under test, only knowing its type. The proposed approach relies on a supra-threshold method where sequence of 1 kHz-centered narrowband stimuli are played in decreasing levels with steps of 5 dB (see Fig. 1 for timeplot of the exact stimuli used). The subject simply counts the number of bursts perceived before stimuli become inaudible in two conditions: with open ears and with both ears occluded with the HPD under evaluation

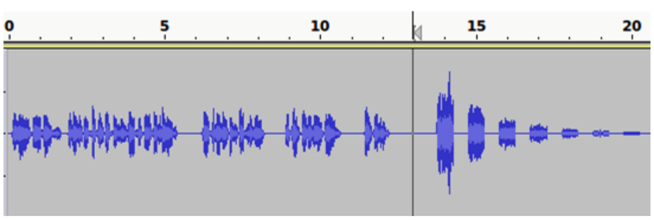

Figure 1: Timeplot of the audio stimuli consisting in a 12-second explanation followed by the test sequence (only 7 of the 15 bursts are visible here).

2.1 Experimental Setup

The proposed method has been developped in laboratory on a dozen test subjects with normal hearing. Each test subject is seated in an audiometric booth equipped for REAT measurement following ANSI S12.6 standard [3]. The tablet running the app and generating the stimuli is held in a fixed location at ear level with the 2 miniature speakers of the tablet facing the subject, as seen in Fig. 2, at a sound pressure level -not controlled but fixed in both open and occluded condition- that reaches up to 75 dB. Four experimental steps are then performed: First, the REAT open threshold is measured, using the Bekesy method with audiometric stimuli presented in a so-called "free field" condition through the booth loudspeakers. Second, the tablet is used to generate the 1 kHz audio stimuli and the subject counts the number of bursts perceived before becoming inaudible. The earplugs under evaluation are then fitted on both ears by the subjects themselves with only the help of a broadband "fitting noise". Third, the REAT occluded thresholds are measured. Fourth, the tablet generates the same audio sequence as in the open condition but increased by 5 dB to slightly overcome the earplug attenuation; that offset value beeing later accounted for in the calculation of the tablet measured attenuation. This whole sequence is repeated a second time for every subject, with a total of 6 subjects tested for the premolded earplugs and 11 subjects tested for the roll-down foamplug.

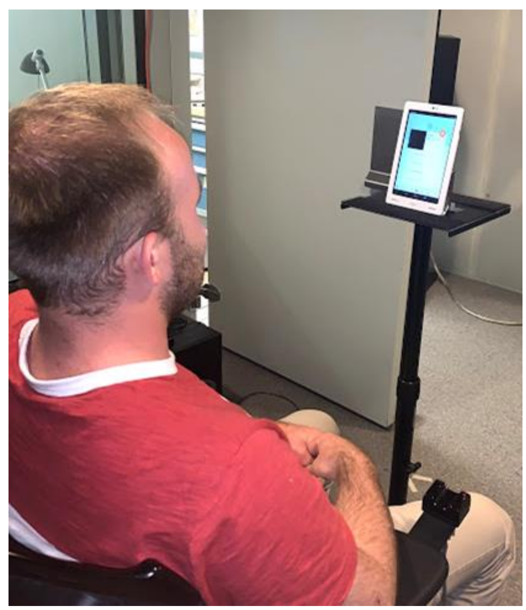

Figure 2: Photography of the test subject seated in the audiometric booth equipped for REAT measurements, clearly showing the tablet used to generate the audio stimuli.

2.2 Computation of Attenuation Values

The two counts values are entered within the app to compute an estimate of the overall HPD attenuation at 1 kHz, by multiplying by 5 dB the difference between the counted stimuli in open and occluded conditions and adding the 5 dB offset. This resulting value is then compared to the binaural

* jeremie.voix@etsmtl.ca

REAT attenuation. The REAT is reported at every octave-band form 125 Hz to 8 kHz and an overall attenuation can also be reorted. This overall attenuation is computed using the "octave band method" [1] as an C-A overall attenuation, hence represents the average difference between the C-weighted overall sound pressure level and the A-weighted overall sound pressure level attenuated by the HPD using a broadband pink noise spectrum.

3 Preliminary Results

3.1 Comparison between REAT and Tablet Measured Attenuation Values

The measured attenuation values using REAT and using the tablet are compared in Fig. 3 and 4. for both the roll-down foamplug and premolded earplug. They show that while the REAT values obtained at 1 kHz (red diamonds) are poorly predicted, the overall REAT attenuation values (blue squares) are better correlated with the attenuation measured at 1 kHz with the tablet.

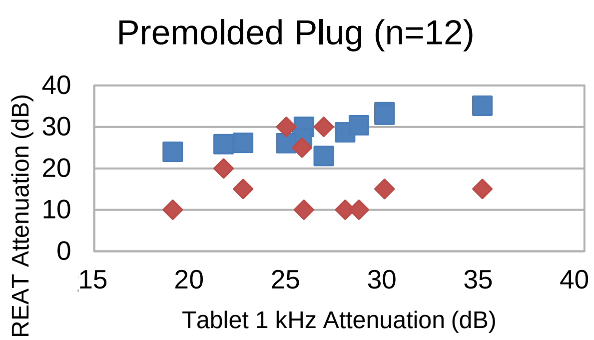

Figure 3: Scaterplot of REAT values obtained at 1 kHz (red diamonds) and overall (blue squares) as a function of attenuation measured at 1 kHz with the tablet for 2 trials of premolded plugs on 6 subjects.

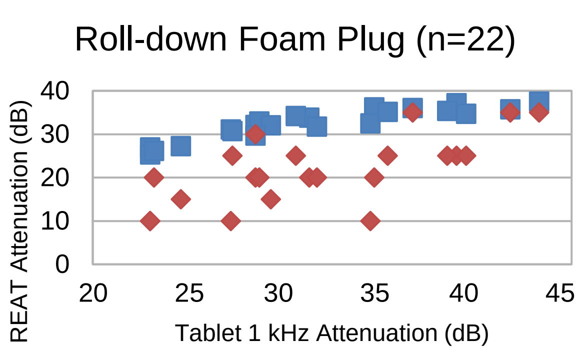

Figure 4: Scaterplot of REAT values obtained at 1 kHz (red diamonds) and overall (blue squares) as a function of attenuation measured at 1 kHz with the tablet for 2 trials of roll-down foam plugs on 11 subjects.

By comparing Fig. 3 and Fig. 4, it can also be seen that the overall REAT attenuation appears to be better correlated with 1 kHz tablet measurement for the roll-down foam plug than for the premolded earplug

4 Discussion

The data obtained suggest that a polynomial regression, using the best fit (in a least-squares sense) method, could be used to link the overall C-A attenuation measured using REAT to the attenuation measured with the tablet at 1 kHz. This regression could later be used as a predictive model, and should probably be a function of the type of HPD tested (roll-down foam, premolded, formable, push-to-fit, custom molded, and even semi-inserts and earmuffs). The reasons why the regression obtained is not a linear should also be investigated, exploring dynamic range limitation, noise floor ceilings, etc. It could also be tempting to add extra tones at lower frequencies in the tablet stimuli burst, in order to better predict the overall attenuation [4]. Nevertheless, it should also be remembered that the level of the stimuli being uncontrolled (but kept identical in both open and occluded condition) only the 1 kHz band will remain unaffected by the dynamic compression of the human hearing as expressed by the equal-loudness contours.

5 Conclusion

This very preliminary study suggests that a simple measurement made using only a tablet or smartphone generating 1 kHz tonal bursts could possibly predict overall attenuation of foamplugs with a limited degree of accuracy but sufficient for general use. Further research is now needed to validate a robust polynomial regression model using a large number of subjects and HPD models.

Acknowledgments

The technical support offered by the laboratory of the NSERC-EERS Industrial Research Chair in In-Ear Technologies (CRITIAS) is greatly acknowledged. The author would also like to thank Mrs. Marine Laplace for having kindly included these preliminary measurements as part of the product validation conducted for EERS Global Technologies.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Which Threshold Do We Trust? A Comparison Of Threshold Measurements In Adult Bone Conduction Users

Alex Gascon, William Hodgetts

Advancements in prescription formulae and fitting protocols for bone anchored hearing devices (BAHD) have highlighted the importance of measuring in-situ bone-conduction hearing thresholds (BCHT). In-situ BCHT are assessed routinely during fittings of percutaneous BAHD and BAHD worn on a soft-band. In-situ measurements are performed within the manufacturer's software, using the patient's BAHD device as a transducer. Uncertainty remains with respect to the relationship between transcutaneous (soft-band) in-situ, percutaneous (abutment) in-situ and traditionally measured (B71 transducer) BCHT; these differences have not been previously studied. In this study, we are measuring each participant's BCHT in three conditions; (1) in-situ with a percutaneous BAHD, (2) in-situ with the BAHD worn on a soft-band and (3) with the B71 transducer using standard audiometry. This cross-sectional study includes patients from the Institute for Reconstructive Sciences in Medicine (iRSM) over 18 years of age, with conductive or mixed hearing loss that use a BAHD. In-situ hearing thresholds are measured with two different BAHD, the Oticon Medical Ponto 3 Super Power and the Ponto 4. By exploring these potential differences, we aim to contribute to the development of more precise prescription and fitting protocols for soft-band BAHD users. The potential to develop and use correction factors to better estimate in-situ BCHT from thresholds measured on a B71 would be especially valuable for pediatric patients who are not always able to complete standard or play audiometry. The results of this study will contribute to future research with a pediatric population.

On Speech Envelope Enhancement For Hearing Aid Applications

Farid Moshgelani, Vijay Parsa, Chris Allan, Sangamanatha Ankmnal Veeranna, Prudence Allen

Children with Auditory Processing Disorder (APD) experience reduced speech intelligibility in the presence of background noise due to plausible deficits in their temporal, spectral, and/or binaural processing. Envelope enhancement (EE) algorithms, which accentuate temporal modulation cues in the speech signal, may improve speech intelligibility scores for children with APD, although a comprehensive evaluation of EE algorithms is currently lacking. In the present study, speech intelligibility was assessed for unprocessed and envelope-enhanced speech in the presence of stationary and non-stationary background noise at four (3, 0, -3, and -6 dB) signal to noise ratios (SNRs) with children with APD, both with and without noise reduction (NR) algorithm as a front-end to the EE algorithm. Furthermore, intrusive and non-intrusive objective models were explored to predict the subjective speech intelligibility data. Results from the subjective experiments demonstrated that the EE algorithm was mainly beneficial for improving the speech intelligibility scores at SNRs < 0 dB, only when the NR algorithm was incorporated as a front-end to the EE algorithm and the background noise was stationary. For the non-stationary background noise, the chosen EE algorithm was less effective for children with APD in improving their speech intelligibility, even with the addition of an NR algorithm as a front-end. On the objective side, proposed estimators exhibited good correlation with the subjective data, with the intrusive estimator demonstrating better generalization capabilities across different subjective databases. Results from this study can potentially help in developing and fine-tuning new EE algorithms, and in guiding the choice and activation of EE algorithms in hearing aid applications.


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AUDITORY PERCEPTION - PERCEPTION AUDITIVE

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LISTENING EFFORT IN EATERIES

Joseph Rovetti *[1], Alberto Behar †[1], Mohammad Abdoli-Eramaki ‡[1], Fran Copelli ♦[1], and Frank Russo #[1]
[1]Ryerson University, Canada

1 Introduction

Eateries such as restaurants, cafeterias, and food courts can be noisy. For instance, they tend to have large crowds of people talking, and often they also have music playing. Most eateries also lack sound-absorbing treatments, such as curtains or carpet, further contributing to the high level of background noise. These factors force people in eateries to raise their voices and increase the effort that they exert during listening, known as listening effort. Despite the relevance of this issue, studies of people's experience of listening in eateries have been limited to the use of subjective measures [1]. In addition, studies of the acoustical environment of eateries have been limited to sound level measurements [2]. Thus, the current study has three broad aims. The first aim is to objectively measure the effort that people exert while listening in eateries, and the second aim is to determine how this effort changes with the quantity and quality of background noise. Finally the study will acoustically characterize the associated noise profiles.

2 Method

2.1 Noise measurements

A stereophonic headset and a digital recorder will be used to collect 10-minute samples of ambient noise in four restaurants, two "quiet" and two "noisy". Collecting samples from "quiet" and "noisy" restaurants will allow the study to determine whether noise in these environments, in addition to their quantitative differences (i.e., the sound level), also differ qualitatively (e.g., the frequency content). The specific noise qualities that will be considered include LAeq, L10, L90, LPeak, as well as 1/8 and 1/3 band spectra

2.2 Speech perception task

Fifteen younger adults will participate in the study. They will complete a task that requires them to listen to sentences, which will be presented among the background noise recorded from restaurants, and repeat back the final word. These sentences will always be presented at the same level, but two variables will be manipulated. The first manipulation will be the type of restaurant from which the background noise was recorded: "quiet" or "noisy". The second will be the level of background noise: no background noise, signal-to-noise ratio (SNR) = +4, SNR = +1, and SNR = -2. To avoid interference noise, participants will complete the task in a double-walled sound-attenuated chamber, with the stimuli presented via insert headphones.

2.3 Functional near-infrared spectroscopy

As participants complete the task, the listening effort that they exert will be measured objectively using functional near-infrared spectroscopy, a non-invasive optical brain imaging method. This method requires the use of a sensor pad affixed to participants' foreheads, which contains four light sources and 10 light detectors. The light sources emit near-infrared light into the scalp above the prefrontal cortex (PFC), and the light detectors measure the amount of light that returns to the sensor pad. Based on the amount of light that results, the concentration of oxygen in the PFC will be calculated. This will serve as an objective measure of the listening effort.

3 Expected results

It is predicted that as SNR decreases (i.e., as the level of background noise increases), the listening effort that participants exert, as indexed by PFC oxygenation, will increase [3]. It is also predicted that oxygenation of the left lateral region of the PFC, including the inferior frontal gyrus, will be most sensitive to changes in SNR. There is no prediction as to whether listening effort will differ across "quiet" and "noisy" restaurants, nor is there a prediction as to whether and how noise will differ qualitatively across these environments.

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* joseph.rovetti@ryerson.ca
† albehar31@gmail.com
‡ m.abdoli@ryerson.ca
♦ fcopelli@ryerson.ca
russo@ryerson.ca

ASSOCIATIONS BETWEEN MUSICAL EXPERIENCE AND AUDITORY DISCRIMINATION

Cory McKenzie [*1], Amberley V. Ostevik [†2], Bill Hodgetts [‡2,3], Jacqueline Cummine [♦2,4]. Daniel Aalto [♯2,3]
[1]Faculty of Science, University of Alberta,
[2]Faculty of Rehabilitation Medicine, University of Alberta,
[3]Institute for Reconstructive Sciences in Medicine, Covenant Health,
[4]Neuroscience and Mental Health Institute, University of Alberta,

1 Introduction

Auditory processing is affected by both musical experience and native language. However, there is uncertainty as to which aspects of auditory perception are influenced by musical experience and whether this varies as a function of native language. Auditory discrimination (the ability to tell apart similar sounds) is an important part of auditory perception. Better pitch discrimination has a consistent relationship with more musical experience, regardless of native language. However, duration discrimination is positively related to musical experience only in some language groups. A previous study with Finnish speakers found no relationship between duration discrimination and musical experience [1], while another study found Mandarin musicians had enhanced duration discrimination when compared to non-musicians [2].

Here, we set out to study the relationship between musical experience and various aspects of auditory discrimination in a sample of English-speaking participants. The first objective was to determine if musical experience was related to auditory discrimination. Second was to determine if individual performance differences between simple and complex duration discrimination, were related to musical experience. Then finally to assess if the degree of modulation of frequency and intensity, on duration perception, is related to musical experience.

2 Methods

2.1 Participants

Eighty-two participants completed the experiment, all were right-handed, 18-33 years old (49% female, mean age of 22 years old), and undergraduate students. They self-reported speaking English natively. 78 participants reported normal hearing and four participants were unsure. This experiment was conducted according to the ethical guidelines of the Declaration of Helsinki and was approved by the Human Research Ethics Board at the University of Alberta (Pro00082110).

* ccmckenz@ualberta.ca
† aostevik@gmail.com
‡ bill.hodgetts@ualberta.ca
♦ jcummine@ualberta.ca
♯ aalto@ualberta.ca

2.2 Procedure

Musical sophistication was scored using the self-report questionnaire of the Goldsmiths Musical Sophistication Index (Gold-MSI) [3]. The questionnaire evaluates musical sophistication based on active engagement, perceptual abilities, musical training, singing abilities, and emotional engagement. These five factors can be scored individually or scored together to create a generalized musical sophistication score. This index is particularly useful for comparing participants with a wide variety of both formal and casual musical experience.

The behavioural task included three tests. All three tests consisted of a forced-choice decision between two different sounds. In the first test, participants were asked which of the two sounds was a higher pitch; in the second, they were asked which sound had a longer duration; and in the third test, they were asked which of two complex sounds was longer. The complex sounds varied in pitch, duration, and intensity. White noise was included in the complex task to mask any potential distortion patterns. The behavioural task was completed on a laptop that was running custom Matlab scripts, and circumaural headphones calibrated to 73 dB were used.

Stimuli

All behavioural task stimuli were in the range of normal human speech. All sounds varied from a standard of 300ms long, 73 dB, and 150 Hz. In the simple pitch test, the frequency varied adaptively while the length and intensity were held constant. In the simple duration test, the length of the sound varied and the pitch and intensity did not change. In the complex task, all three features varied randomly. For duration, the sounds had a standard deviation of 75ms and were between 150ms and 450ms. The fundamental frequency of the sounds had a standard deviation of four semitones, and there was a pitch glide which also had a standard deviation of four semitones. The intensity had a standard deviation of 1 dB.

3 Results

Six correlations were used to compare behavioural data to Gold-MSI scores (Table 1). At a Bonferroni corrected alpha value (α=0.0083), Gold-MSI scores are only significantly correlated with simple pitch discrimination (rs=0.297, p=0.008). At an uncorrected alpha, Gold-MSI is also correlated to complex duration discrimination (rs=0.218, p=0.049) and intensity modulation (rs=-0.283, p=0.010). Simple duration discrimination, duration ratio, and pitch modulation have no significant relationship with Gold-MSI scores.

Musical experience is related to some aspects of auditory discrimination, but not all. Greater pitch discrimination (a more negative slope) is associated with more musical experience (Table 1). Degree of musical experience is not associated with duration discrimination ability (Table 1). Musical experience was not a factor in how much changes in frequency or intensity affected duration discrimination (Table 1). There is also not a relationship between musical experience and how much performance changed from the simple to complex duration discrimination tasks (Table 1).

Table 1: Correlations performed on behavioral data (Spearman's rho). Scores on the Goldsmith Musical Sophistication Index questionnaire (MSI) are correlated to various scores from behavioral discrimination tasks.

Correlation	Spearman's Rho	p value
MSI & Simple Duration Discrimination	0.185	0.100
MSI & Pitch Discrimination	-0.297*	0.008
MSI & Complex Duration Discrimination	0.218	0.049
MSI & Duration Ratio	-0.054	0.0635
MSI & Frequency Modulation	0.014	0.904
MSI & Intensity Modulation	-0.283	0.010

*Indicates significant p-values (α=0.0083).

Duration, intensity, frequency, and frequency range all significantly impacted duration discrimination in the complex duration discrimination task (Table 2). This confirms that when ignoring musical experience, there are modulation effects. There is also a significant difference in performance between the simple and complex duration tasks ($t=9.303$, $p=2.22e^{-14}$).

4 Discussion

Musical experience was expected to have a positive relationship with both pitch and duration discrimination in a population of English speakers, however it was related only to pitch. These results support previous findings that pitch discrimination is related to musical experience regardless of language. However, the lack of a relationship between musical experience and duration discrimination suggests that there is still more research needed to determine how native language impacts if musical experience improves duration discrimination. Previously, it was proposed that speaking a quantity language (in which duration plays an important role in phonetics) increased duration discrimination to a maximum ceiling, and that musical expe-rience could not improve it further [1]. This is not supported by the present study.

This study highlights many areas of potential future study. First and foremost, in understanding how language plays a role in duration perception. The findings that

Table 2: Means of the coefficients of logistic regression models fitted to each participant's duration responses separately with frequency range difference calculated as the difference between the absolute values of the dynamic f_0 ranges.

Effect	Mean	95% CI	t value	p value	Cohen's d
intercepts	0.23	(0.14, 0.32)	5.0	$p=3e^{-6}$	1.411
Duration Difference	20	(18, 22)	21.0	$p<2e^{-16}$	23.191
Intensity Difference	0.1	(0.08, 0.12)	11.0	$p<2e^{-16}$	6.075
Frequency Difference	0.068	(0.036, 0.100)	4.2	$p=6e^{-5}$	0.986
Frequency Range Difference	0.035	(0.022, 0.048)	5.4	$p=6e^{-7}$	3.470

musical experience was 1) not related to individual performance differences between simple and complex duration discrimination, or 2) to the degree of modulation of frequency and intensity on duration perception, require more research to properly interpret. The impact of other factors, such as bilingualism, culture, and cognitive abilities on the relationship between musical experience and auditory discrimination remains unstudied.

5 Conclusion

Scores from a self-report measure of both formal and casual musical experience (Gold-MSI) are correlated with pitch discrimination in a population of adult English speakers. However, 1) both simple and complex duration discri-mination are not related to musical experience, 2) there is no correlation between musical experience and the degree of decrement in duration discrimination performance on a complex task, compared to a simple single-feature task, and 3) duration perception effects, as a function of changes in auditory features (i.e., frequency and intensity), are not significantly related to musical experience in English spea-kers. Overall, these findings suggest that musical experience does not enhance all aspects of auditory processing equally, and that any enhancements may be dependent on the particular features of the individual's language background.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Is Apple's Virtual Assistant Siri Familiar Enough To Lead To Better Understanding And Easier Listening Than An Unfamiliar Voice?

William Hodgetts, Daniel Aalto, Jacqueline Cummine

Background: Previous work has shown that familiar voices are easier to understand in the presence of an unfamiliar competing masker. Interestingly, familiar voices are also easier to ignore when they are the masker and the target is an unfamiliar voice (Johnsrude et al., 2013). In this work we are looking to extend these findings to a virtual assistant, Siri, that is available on Apple devices. Methods: Thirty adult listeners with normal hearing are being recruited. Participants will perform a complex hearing task called a coordinate response measure. Participants will be listening through speakers to a target (identified as Baron) saying a coloured number (one out of 32 options; 4 rows of differently-coloured numbers from 1 to 8) while simultaneously listening to a masker saying another call sign. Participants are asked to identify the target coloured number via a mouse. In block 1, participants will be listening to a familiar target voice (i.e., Siri) and an unfamiliar masker voice (i.e., alternate Apple voice). In block 2, participants will be listening to two unfamiliar voices. If the participant chooses the right call sign, the level of the next target drops by 1dB. If the participant chooses the incorrect call sign, the level of the next target goes up by 3dB. This holds the participant performance at a 75% accuracy. At the conclusion of the experiment, the target to masker ratio is obtained. Our expectation is that the familiar target voice will be easier to hear, resulting in a lower target to masker ratio.

Non-Native Lombard Speech Intelligibility For Native And Non-Native Listeners

Katherine Pearl Marcoux, Benjamin V. Tucker, Mirjam Ernestus

Lombard speech production, speech produced in noise, is well studied in native speakers but not in non-natives. This research aims to fill this gap by investigating whether the Lombard intelligibility benefit, where Lombard speech is more intelligible than plain speech when heard in noise, is comparable for two different non-native listener groups. Native Dutch and native Spanish/Basque listeners completed an intelligibility test using English stimuli from the Dutch English Lombard Native Non-Native corpus (Marcoux & Ernestus, 2019). For our stimuli, we used nouns from plain and Lombard speech produced by native (American-English) and by non-native (Dutch) speakers. We expect all listener groups to show the Lombard intelligibility benefit for both native and non-native speech, with a larger benefit for native than for non-native speech. When the English speech is produced by Dutch natives, we expect the Lombard intelligibility benefit to be greater for the Dutch listeners than for the Spanish/Basque listeners, since they share the same native language. In the future, we will additionally collect data from native English listeners as a baseline. By examining the perception of native and non-native Lombard speech, we hope to gain insight into the characteristics of non-native Lombard speech.

BIOACOUSTICS - BIOACOUSTIQUE

Features Of Male- And Female-Produced Song In Black-Capped Chickadees (Poecile Atricapillus) Change Between Seasons
Kimberley Ann Campbell, Stephanie Thunberg, Christopher B Sturdy 36

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Features of Male- and Female-Produced Song in Black-Capped Chickadees (*Poecile Atricapillus*) Change Between Seasons

Kimberley A. Campbell *[1], Stephanie Thunberg †[2], and Christopher B. Sturdy ‡[1,3]
[1] Department of Psychology, University of Alberta
[2] Department of Biology, University of Alberta
[3] Centre for Neuroscience, University of Alberta

1 Introduction

Of the 10 000 or so known avian species, nearly half belong to the clade Passeri [1]. Also called Oscines, or songbirds, birds in this group are unique in that they possess a morphologically complex vocal organ, the syrinx, which facilitates the production of complex and diverse vocalizations, including their song [2]. Songbird song is often learned and functions in both territory defense and mate attraction [3].

Though songbird song has been extensively studied, most research has focused on temperate species in which males tend to sing long, complex songs and females have been thought not to sing [4]. Recently, increasing accounts of female song in temperate species have been documented, suggesting that both males and females produce song [5]. Despite this, quantitative evaluations of the acoustic structure of female song, especially as compared to male song, are lacking (though see [6 - 8]).

The production and acoustic structure of female song was recently described in black-capped chickadees (*Poecile atricapillus*) [8]. These small, North American songbirds produce an acoustically simple song, the *fee-bee*, as compared to most other songbird species, consisting of only two notes [9] (see Figure 1). Though we now know that both sexes produce *fee-bee* song, the function of female song is not yet well understood. In this experiment, we sought to investigate how male- and female-produced black-capped chickadee *fee-bee* song changed in acoustic structure throughout the year.

2 Method

2.1 Subjects

Recordings of *fee-bee* songs were used from eight adult black-capped chickadees (five male, three female). Birds were captured as adults in Edmonton, Alberta (53.53°N, 113.53°W) and Stony Plain, Alberta (53.45°N, 114.01°W) between 24 December 2010 and 20 February 2017. Sex was determined by deoxyribonucleic acid (DNA) analysis [10].

2.2 Recordings

Birds were recorded during two seasons, spring and fall, between March 20, 2012 and June 20, 2016. Each recording session lasted for one hour from 8:00 – 9:00. Birds were housed, transported, and recorded in individual 30 cm wide×40 cm high×40 cm deep Jupiter Parakeet cages (Rolf C. Hagen, Inc., Montreal, QC). Individual birds were recorded using a Marantz PMD670 (Marantz America, Mahwah, NJ) digital recorder set to a 16 bit, 44 100Hz sampling rate, and an AKG C 1000S (AKG Acoustics, Vienna, Austria) microphone set up in 1.7m×0.84m×0.58m sound attenuating chambers (Industrial Acoustics Company, Bronx, NY). The microphone was positioned 30 cm above the back center of the top of the cage.

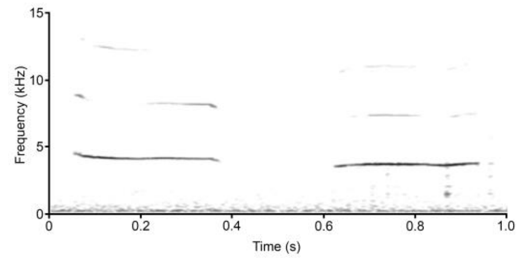

Figure 1: Sound spectrogram of a black-capped chickadee *fee-bee* song (window size = 256 points, time resolution = 5.8 ms).

2.3 Measures

Each bird produced at least one *fee-bee* song during each season of interest, spring and fall. Songs were chosen randomly from available songs.

A series of 13 frequency measures were obtained using the 'specan' function in the R package, warbler [11] (mean frequency, standard deviation, frequency median, frequency of the 25th quartile, frequency interquartile range, frequency of the 75th quartile, mean dominant frequency, maximum dominant frequency, dominant frequency range, start of dominant frequency, end of dominant frequency, slope of dominant frequency, mean frequency peak). Similarly, three amplitude measures were obtained using the 'analyzeFolder' function in the R package, soundgen [12] (mean amplitude, amplitude median, standard deviation of amplitude).

2.4 Statistical Analyses

We conducted a principle components analysis (PCA) on frequency and amplitude measures in order to reduce them to a smaller subset of independent measures. Once these principle components were calculated, we conducted a repeated measures analysis of variance (ANOVA) with sex and season as independent variables and the principle components as dependent variables. All statistical analyses were conducted using R, version 3.3.2 [13].

* ki6@ualberta.ca
† sditlof@ualberta.ca
‡ csturdy@ualberta.ca

3 Results

The first principle component (PC1) explained 63.2% of the variance in frequency measures, while the second (PC2) explained an additional 20.7% of variance. Maximum dominant frequency and dominant frequency range contributed the most to PC1. Start of dominant frequency and dominant frequency slope contributed the most to PC2. We chose to continue with these two principle components as they explained a cumulative 83.9% of variance. The first two principle components for amplitude measures explained 95.3% and 4.6% of variance, respectively. Amplitude mean and amplitude median contributed the most to both PC1 and PC2. These two principle components explained a cumulative variance of 99.9%.

We conducted a repeated measures ANOVA to determine how the acoustic measures represented by the principle components differed between the sexes and the seasons. There was a significant main effect of PC1 for frequency measures such that songs produced in the spring had higher scores than those produced in the fall ($p = 0.001$, $F_{1,6} = 37.191$). All other main effects and interactions were not significant ($p > 0.139$).

4 Discussion

Variation in song structure has not yet been described in black-capped chickadees. This experiment was designed to evaluate acoustic differences in *fee-bee* song structure produced by males or females, as well as between two seasons, spring and fall.

We found that the structure of fee-bee songs changes between the seasons. This could be due to the critical biological functions, such as mate attraction, territory defense, and solicitation of extrapair copulations, which are more critical in spring, which corresponds to the black-capped chickadee breeding season, than they are in the fall [14]. This parallels our findings that songs produced by both sexes show less acoustic variation in spring-produced songs than in fall-produced songs.

5 Conclusion

Overall, our results have demonstrated that the songs of both sexes differ in acoustic structure between the fall and the spring. Future studies should further evaluate the nature of the acoustic change in songs across seasons, as well as evaluating whether the method or degree of change differs between the sexes. Learning more about the changes in acoustic structure of the *fee-bee* song may aid in better understanding the use and function of the vocalization, especially in female black-capped chickadees, whose use of the *fee-bee* song is not yet well studied.

Acknowledgments

The studies described here were conducted in accordance with the Canadian Council on Animal Care Guidelines and Policies with approval from the Animal Care and Use Committee for Biosciences for the University of Alberta (AUP 108). Capture occurred under Environment Canada Canadian Wildlife Service Scientific permit, Alberta Fish and Wildlife Capture and Research permits, and City of Edmonton Parks Permit. This research was supported by a Natural Sciences and Engineering Research Council of Canada Discovery Grant and Discovery Accelerator Supplement, an Alberta Ingenuity Fund New Faculty Grant, a Canada Foundation for Innovation New Opportunities Fund and Infrastructure Operating Fund grants along with start-up funding and CFI partner funding from the University of Alberta to CBS. KAC was supported by an Alexander Graham Bell Canada Graduate Scholarship.

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Wiretapping The Wilderness: New Approaches To Wildlife Monitoring Using Autonomous Recording Units

Erin Bayne

Soundscapes provide a diverse mixture of wildlife, human, and other natural sounds that can be used for documenting status, trend, and impacts. Historically, many animal populations have been monitored by human observers visiting a location for a short period of time once a year and recording the species heard. These approaches have been uncoordinated and poorly standardized, which has prevented large-scale assessment of the cumulative effects of different types of human development in space and time. Ineffective decision making has been the end result. Automated recording units (ARUs) provide a new approach to environmental monitoring that relies on a physical record of soundscapes that can be stored and used by various partners to make more informed decisions about the environmental impacts in a consistent and repeatable way. The Bioacoustics Unit at the University of Alberta – Alberta Biodiversity Monitoring Institute is a collaborative partnership working in Alberta's oilsands region that has standardized approaches to the collection and processing of sound information using ARUs. We have created a platform called WildTrax that integrates advances in computer-based species recognition, proper handling of species detection error, and ARU sampling design. The success of the approach in coordinating government, non-profit, and industry in an effort to assess and mitigate the effects of industrial noise and light on amphibians and owls in the oilsands region of northern Alberta will be described as case studies of how acoustic technology can improve environmental monitoring performance.

Distance Truncation Via Relative Sound Level For Surveys Of Birds In Patchy Habitat

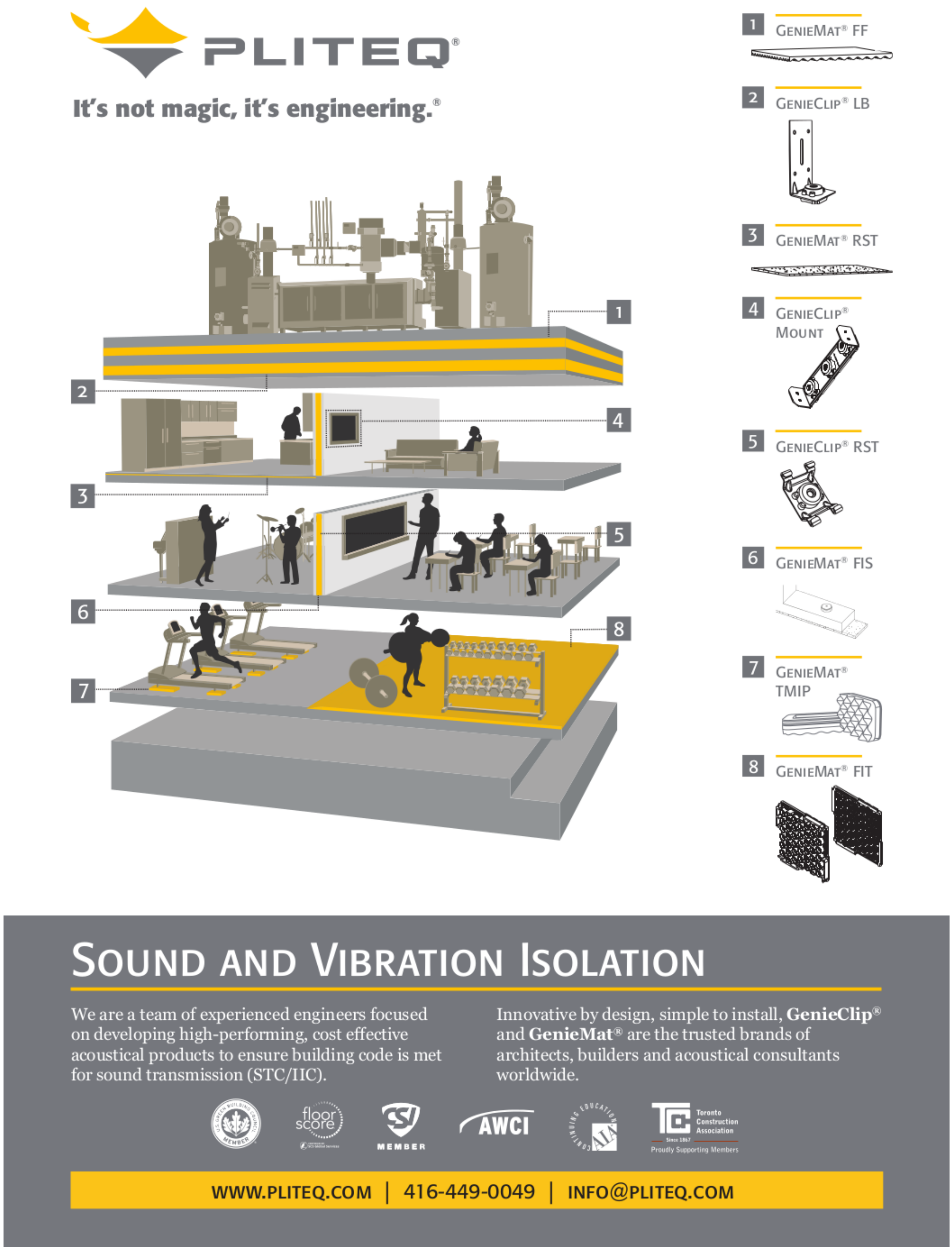
Richard William Hedley

Acoustic surveys have emerged as a primary technique for surveying vocal animals and assessing population trends. Acoustic surveys typically follow roughly the same survey protocols as traditional counts made by humans in the field: humans listen to and identify all species over a brief period of time (e.g. 3 minutes). One disadvantage of this type of survey is that most birds are detected at great distances from the survey point (often >100m). Since the bearing to the detected bird is unknown, there is great uncertainty about the actual location of the bird and the habitat from which they are vocalizing. Often, the bird is using habitats that are very different from the habitat at the point (e.g. wetland vs forest), posing problems for modeling habitat associations. We present an alternative survey method to reduce this issue. Rather than listen to short recordings, we automatically extract vocalizations exceeding a sound level threshold from longer recordings, inspired by previous results relating the measured sound level with the distance of the sound source. Birds whose vocalizations exceed the sound level threshold are expected to be very near the microphone, and likely in the same habitat as the microphone. We present a comparison between this method and traditional listening surveys, applied to the question of surveying birds on small (1 hectare) well sites in northern Alberta. We found listening surveys to be unsuitable for this application, because most detected birds were singing from outside the target survey area (the well site). Preliminary results suggest our approach detects fewer birds, a much higher proportion of which were within the target survey areas, making this method more suitable for this application. We propose that this method will be generally useful for surveying patchy habitat, or when greater certainty about source location is desired.

Automated Acoustic Mark-Recapture Of A Migratory Bird Using Signal Recognition Software

Elly Caitlin Knight, Erin M Bayne

Many species of acoustic animals have been shown to have individual-specific vocal characteristics and so monitoring wildlife populations via those vocalizations has been proposed as a non-invasive alternative to traditional capture-mark-recapture studies. In particular, individual-specific vocalizations can be used to determine whether individuals return to the same breeding area between years. Current approaches to acoustic mark-recapture involve hand-recording specific individuals in the field and hand measurement of vocal characteristics from the resultant recordings, both of which are time intensive. Here, we use a nightjar species, the common nighthawk (Chordeiles minor) to test whether inter-annual acoustic mark-recapture can be achieved with passive acoustic monitoring and signal recognition. The nightjar family (Caprimulgidae), is a group of migratory, nocturnal birds who rely heavily on acoustic communication and whose vocalizations are particularly well-suited for signal recognition. We used autonomous recording units (ARUs) to record 'mark' clips of the vocalizations of 12 male common nighthawks during the breeding season of 2016 north of Ft. McMurray, Alberta. We used those 'mark' clips to train individual-specific recognizers for each bird using Song Scope signal recognition software. We then used each recognizer to scan a set of 'recapture' recordings that were collected with ARUs the following breeding season in 2017. We predicted that if an individual returned to their territory, that there would be more detections with a higher classification probability than if the individual had not returned. We validated our approach with known returns of six of the individuals. Preliminary results indicate a potential for false negatives if neighbouring individuals have a high degree of vocal similarity. False negatives may also be caused by the mobility and atypical territorial behaviour of common nighthawks.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BUILDING ACOUSTICS - ACOUSTIQUE DU BÂTIMENT

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GENERIC THIRD-OCTAVE BAND VIBRATION SPECTRA FOR CONSTRUCTION EQUIPMENT

Todd Busch *†1

[1]Todd Busch Consulting, Mississauga, Ontario, Canada

1 Introduction

A review of the literature on assessments of the impact of construction vibration on sensitive buildings and activities suggests that there is a reliance on total vibration levels without consideration of the spectra of different equipment. The author has reviewed several sources of information, including measurement reports, textbooks [1,2], conference publications [3 - 5], and regulatory guideline documents [6] to arrive at generic spectra for nineteen different types of construction equipment.

The methodology relied upon the available information about total vibration levels and assumptions regarding different frequency regimes. At low frequency, a constant displacement was assumed that transitions at mid-frequencies to constant velocity, followed at high frequencies by constant acceleration. The resulting spectra from 1 Hz to 250 Hz correspond to the information available from other sources in a generic sense. The application of these construction-equipment spectra requires further information about the propagation losses of vibration from a construction site to a sensitive building or activity. Methods for documenting propagation losses are also addressed [7, 8].

2 Development of Vibration Velocity Spectra

The procedure for creating generic spectra relies, first and foremost, on the available peak particle velocity (PPV) of representative construction equipment, typically expressed as a total amplitude in the time domain. This was done through the review of literature described above. With reference to available spectral measurement data, an assessment was made of the number of 1/3-octave bands that were effectively plateauing in the mid-frequency range where constant vibration velocity was assumed. The percentage of energy in these plateaud 1/3-octave bands was determined and then the amplitudes/levels in the lower and higher frequency ranges were determined using adjustments for constant acceleration and constant displacement, respectively. The rationale for assuming three regimes is based on inspection of published measurement data that was reviewed as part of the analysis done to support development of this publication. The following equation was used to determine the 1/3-octave band levels in the constant velocity region of the frequency domain using the information shown in Table 1:

$$\sqrt{\text{Highest FrequencyDomain PPV}^2 * \frac{\%\ \text{Energy}}{\text{Number 1/3 Octaves}}} \qquad (1)$$

Table 1: Summary of Assumed Construction Equipment Spectral Characteristics.

Equipment Class	Highest 1/3-Octave Band, Hz	% Energy	Number 1/3-Octaves	Highest Frequency Domain PPV @ 25 ft, in/sec
Excavator (Auger Drill)	10	80	5	0.008
Compactor, Vibratory	31.5	80	1	0.00577
Concrete Mixer	40	80	5	0.00165
Truck, Concrete Pump	16	80	5	0.00656
Truck, Concrete	16	80	5	0.00656
Concrete Vibrator	20	80	5	0.00060
Crane (Material Handling)	16	80	5	0.00656
Drill, Rock + Compressor	31.5	80	3	0.00268
Truck, Dump	16	80	5	0.00656
Excavator / Loader / Backhoe (Bucket)	12.5	80	5	0.00769
Forklift	25	100	5	0.00008
Grader	8	80	7	0.00099
Hammer, Riveting + Compressor	20	80	5	0.00151
Jackhammer + Compressor	20	80	4	0.00241
Pneumatic Tools + Compressor	20	80	5	0.00030
Saw, Concrete + Generator	63	80	3	0.00027
Truck, Flat-Bed	16	80	5	0.00656
Truck, Medium	20	80	5	0.00328
Welder	80	100	9	0.00001

The results are shown in Figure 1 from 1 Hz to 250 Hz for each of the nineteen equipment classes. As can be seen, a vibratory compactor is expected to have regimes of constant acceleration and constant displacement but no central region in the frequency domain where constant velocity is evident.

* toddbuschconsulting@shaw.ca
† toddbusch@hotmail.com

A jackhammer, being an impulsive vibration source, is expected to have more of a broadband characteristic, although the results shown also assume that there are harmonics of the fundamental impact frequency that come into play. Based on prior measurement experience, harmonics may also be evident for a vibratory compactor.

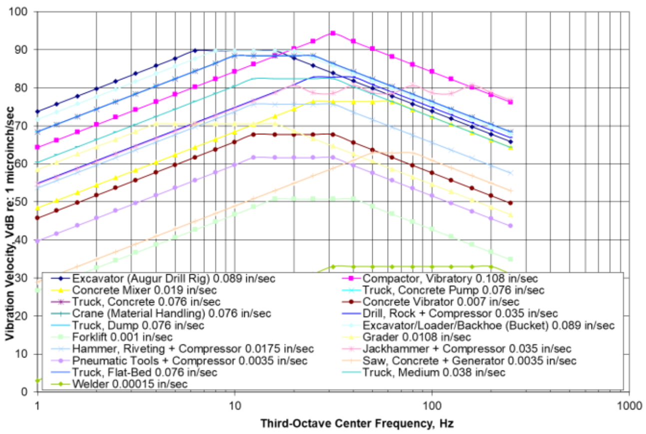

Figure 1: Generic Construction Equipment Vibration Velocity Level Spectra @ 25 ft.

3 Attenuation Law

Reference [8] provides a good framework for addressing the propagation of vibration from construction equipment through the ground. The attenuation law is given by :

$$V/Vi = D^*(Ri/R)^n \quad (2)$$

V = Vibration amplitude at distance R.
Vi = Vibration amplitude at reference distance Ri.
n = Exponent of the attenuation law.
D = Factor taking account of the transmitting medium.

The appropriate exponent n for amplitude reduction depends on the following factors (see Table 2):

- The geometry of the vibration source (e.g., point, line).
- The predominant type of wave (e.g., Rayleigh, body).
- The type of excitation (e.g., steady state, impulsive).

Table 2: Factor n for Construction Equipment Point Sources

Excitation	Rayleigh	Body
Steady State	0.5	1.0
Impulsive	1.0	1.5

To apply the values in Table 2 it is necessary to assume that the vibration energy is apportioned between Rayleigh (i.e., surface waves) and/or body waves that are dominant at depth below a ground surface [9].

The material term D is given by $D = \exp([-\alpha(R-Ri)])$ with the attenuation coefficient $\alpha = 2\pi\varsigma/\lambda$ where ς is the damping ratio of the transmitting medium and λ is the wavelength. For loose soils, ς can be assumed to be 0.01.

Reference [7] provides an alternative formulation that relies on time-domain exponential absorption coefficients to determine the rate of propagation loss through a variety of soil types. These are catalogued for illustration. The attenuation law of reference [8] can potentially be applied if the damping ratio of soil is known or can otherwise be determined through measurement. Ideally, frequency dependent.

4 Conclusion

The information provided above provides a framework for the generic estimation of construction equipment vibration levels at receptors of concern. The results can be improved if measurements are conducted of actual construction activity to determine vibration levels at a typical reference distance and/or the propagation loss from a construction site to a sensitive receptor property.

Acknowledgments

The author would like to thank the many people that have contributed over the years to refinement of techniques for assessing vibration from construction equipment.

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Radiation Efficiency Of Cross Laminated Timber Panels By Finite Element Modelling

Jianhui Zhou *1, Behzad Vafaeian†2, Ying Hei Chui ‡2
1Wood Engineering, University of Northern British Columbia
2Department of Civil and Environmental Engineering, University of Alberta

1 Introduction

Tall wood buildings are gaining an increasing interest in North America due to the recent development of mass timber panels and construction techniques. Cross laminated timber (CLT) is one of the most popular mass timber products, which consists of an odd number of orthogonal layers of dimensional lumber planks glued together by a structural adhesive. The cross-laminated layered structure in CLT makes it a dimensional stable engineered wood panel, but also an orthotropic plate in terms of its elastic properties. Its relative high strength/stiffness to density ratio is beneficial for structural seismic resilience, however, also causes challenges in meeting sound insulation requirements. Both the airborne and impact sound insulation performance of CLT panels have been tested to be low [1].

The sound insulation of a building element depends on its dynamic response to the actual excitation of either acoustic field or mechanical force and the efficiency as a sound radiator given the actual response pattern [2]. As an orthotropic material, the sound radiation behavior of CLT plates is not well understood. This study utilized a finite element (FE) modelling approach using ABAQUS FE software to investigate the sound radiation efficiency (RE) of typical Canadian CLT panels with experimental elastic constants.

2 Method

2.1 Theory

The RE of a given vibrating plate in a fluid is defined by:

$$\sigma = \frac{\bar{W}}{\rho_o c_o S \overline{\langle v^2 \rangle}} \text{ or equivalently } \sigma = \frac{\langle \bar{I} \rangle}{\rho_o c_o \overline{\langle v^2 \rangle}} \qquad (1)$$

where $\bar{W}$ the time-average radiated sound power, ρ_o and c_o are the fluid's density and sound speed, S is the plate surface area, and $\overline{\langle v^2 \rangle}$ is the spatially averaged (temporal) mean-square velocity component normal to the plate surface. The term $\langle \bar{I} \rangle$ is the spatially averaged time-average sound intensity component normal to the vibrating plate.

2.2 Finite element modelling

To investigate the RE of a vibrating plate, we considered a plate with specific boundary conditions (BCs) at its middle surface situated above a semi-infinite (rigid baffle) acoustic medium (air) under forced vibration by harmonic loads. The plate was modeled as thick shell (4-node elements full integration) being coupled with the semi-infinite acoustic medium through solid-acoustic interaction. The acoustic medium contained three regions: a cubic region of fine mesh right below the plate (fine mesh assured accurate output), a spherical-segment region of coarse mesh (this region directed waves to the virtual extension of the medium), and the virtual extension of the medium to the infinity. The first two regions were modeled by tetrahedral acoustic elements (4-node) and the virtual region by acoustic infinite elements. Fig. 1 demonstrates the configuration of the FE model. The model was analysed under steady-state harmonic conditions. The loads were harmonic point loads at certain locations, and with the frequencies in the range of 10-1000Hz with a 3-Hz step. Loads with different locations were separately modeled and their resultant RE values were averaged at each frequency.

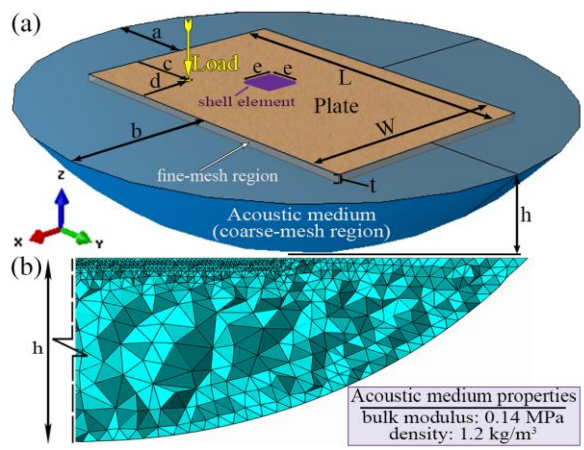

Figure 1: (a) the configuration of the FE model, (b) typical element size transition in the models.

The direct outputs of the modeling software were the complex-valued acoustic intensity vectors of the elements (at their centroid) below the plate, and the complex-valued nodal velocity vectors of the plate. The real part of the normal (to the plate) component of the intensity vector is equal to the time-average intensity ($\bar{I}$) as in Eq. 1. The half of the squared magnitude of the complex normal component of the velocity vector equals to $\overline{v^2}$ in Eq. 1.

The FE element size and the size of the explicitly modeled acoustic medium determine the accuracy of the FE model. We chose the largest element size of each medium equal to 0.1 of the smallest wavelength (λ) of propagating waves in the medium; but for the coarse-mesh region, it contained element size transition from 0.1λ to 0.3λ (Fig. 1b). Mesh sensitivity analyses on the element and acoustic

*jianhui.zhou@unbc.ca
† vafaeian@ualberta.ca
‡ yhc@ualberta.ca

medium sizes were performed and the accuracy of the results, for the purpose of model verification, was assessed by comparing the results with an analytical model regarding RE of an isotropic aluminum plate [3]. Subsequently, CLT panels were modeled as orthotropic plates coupled with the acoustic medium. The dimensions, material properties, and load cases of the plate are listed in Table 1.

Table 1: Properties of the models for verification of the FE models.

Model: 80mm CLT plate [3]
BCs1: all edges simply supported; BCs2: all edges fully clamped
$E_X = 10800MPa$, $E_Y = 734MPa$, $G_{XY} = 600MPa$, $G_{XZ} = 240MPa$, $G_{YZ} = 572MPa$, $\nu_{XY} = 0.02$, $\rho = 484\,kg/m^3$, $\mu = 0.06$
$W = 2900mm$, $L = 4200mm$, $a = 1220mm$, $b = 1870mm$, $t = 90mm$, $h = 1660mm$, $e \approx 30mm$
Load case 1: $c = 500mm$, $d = 800mm$
Load case 2: $c = 3600mm$, $d = 900mm$
E, G: Young's and shear moduli, ν: Poisson's ratio, ρ: density, μ: loss factor.

Following the verification of the results of the 80mm CLT plate, we modeled Canadian CLT plates (Table 2). The acoustic medium was scaled to fit the plates.

Table 2: Properties of the Canadian CLT plates' models.

Canadian CLT plates ($W = 3660mm$, $L = 2440mm$) BCs: all edges simply supported						
Model	E_X (MPa)	E_Y (MPa)	G_{XY} (MPa)	G_{XZ} (MPa)	G_{YZ} (MPa)	t (mm)
3ply	10393	711	518	240	300	105
5ply-1	9657	2293	542	110	360	131
5ply-2	9368	2201	543	200	350	175
7ply	9873	2488	338	300	400	220
$\nu_{XY} = 0.02$, $\rho = 520\,kg/m^3$, $\mu = 0.08$, $e \approx 30mm$; Load case 1: $c = 1830mm$, $d = 1220mm$; Load case 2: $c = 915mm$, $d = 610mm$						

3 Results and Discussion

3.1 Verification of CLT plate with literature

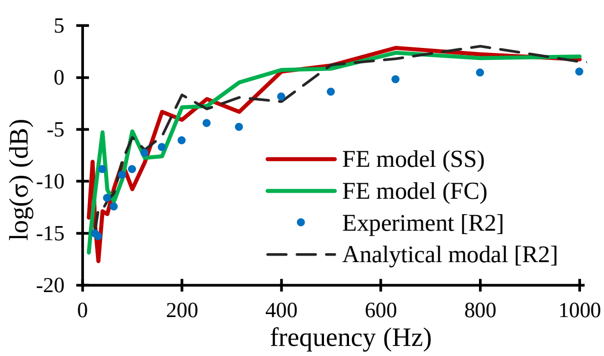

Figure 2: Comparison of the simulated CLT plate RE (σ) with the experimental and analytical values [4]. SS: simply supported, FC: fully clamped.

A general agreement can be found among the results in Fig. 2, but experimental values are always lower than modelled values with frequency higher than 200 Hz. The boundary conditions affect the RE values at certain frequencies.

3.2 Application to Canadian CLT plates

As shown in Fig.3, the RE values of different CLT plates varies under the critical frequencies, which is around 400 Hz for the 5 and 7-ply CLT plates. The 3-ply CLT plate has lower RE values than the others under the same frequency band with higher critical frequencies around 600 Hz. Due to the difference in layup and thickness, the elastic constants and area density varies among these CLT plates. The 3-ply CLT plate has less orthotropy than the other three plates. The other three plate have closer elastic constants and especially higher E_Y values than that of the 3-ply CLT. The effect of area density plays a major role in the difference among 5 and 7-ply CLT plates.

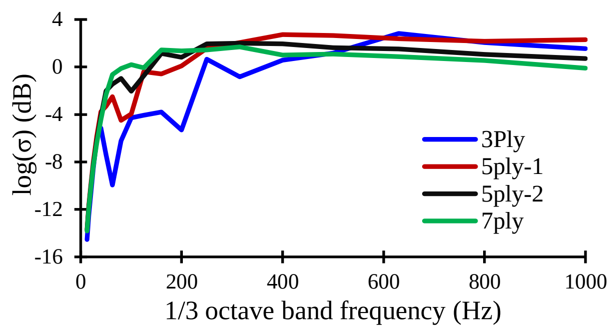

Figure 3: Simulated RE of Canadian CLT plates.

4 Conclusion

The proposed FEM approach is effective in evaluating the radiation efficiencies of different CLT panels. The elastic constants and boundary conditions are critical to the accuracy of the model. Further study will focus on the experimental verification and characterization of these effects.

Acknowledgments

The financial supports from NSERC Industrial Research Chair in Engineered Wood and Building Systems and Forest Innovation Investment-Wood First program are appreciated.

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ACOUSTICAL CONSIDERATIONS FOR DESIGN-BUILD MENTAL AND BEHAVIOURAL HEALTHCARE FACILITIES

Paul E Marks*
BKL Consultants Ltd., #308-1200 Lynn Valley Road, North Vancouver, British Columbia V7J 2A2

1 Introduction

The links between acoustics and the health and wellbeing of the general population in hospital are well known. Evidence suggests that acoustical conditions play an important role in patient health, such that positive "soundscapes" assist with the recovery process (healing and therapeutic) while poor acoustical conditions can cause irritability, anxiety, stress-related conditions as well as lower immune systems.

However, mental or behavioural healthcare has a varied occurrence, with patients that exhibit a range of conditions, capabilities, healthcare requirements and experiences usually not seen in mainstream healthcare. These symptoms do not always necessarily equate to the "norm" and the patient population is typically very vulnerable, with a low tolerance to stressors. Depression, anxiety, behavioral or personality disorders as well as abusive or chaotic tendencies factor into their healthcare requirements. In some cases, noise or even particular sounds may play some role or act as a trigger to some symptoms.

At this time, typical healthcare acoustic design standards for more mainstream facilities in Canada include CSA Z8000 [1], the FGI Guidelines for Health Care Facilities [2] or the ASHRAE Handbook [3]. These design standards or guidelines establish requirements for sound isolation, both for respite care and patient privacy, acoustical comfort and the underlying background sound levels. In general, the design standards can be considered appropriate for the promotion of health and well-being in mainstream healthcare buildings, ensuring patients can recover from or be treated for any serious illnesses without the risk of undue disturbance.

However, within mental or behavioural healthcare facilities, unduly quiet acoustical conditions can lead to a feeling of isolation or loneliness. Similarly, high levels of sound isolation can lead to patient agitation or healthcare staff not being able to adequately hear patients who are in a state of distress or self-harming. Patient responses within mental or behavioural healthcare facilities can be very different to those in mainstream healthcare facilities and the potential heightened response to various triggers, as well as the low tolerance of external conditions can be challenging to the usual well-rehearsed acoustical treatments, acousticians are accustomed to.

2 Treatment Spaces and Acoustical Needs

The acoustical needs for mental or behavioural healthcare facilities need to include a wide range of acoustical treatments, all of which have different solutions. Examples of the various units and treatments include the following:

2.1 Emergency or Crisis Treatment Units

Emergency psychiatric units are inherently noisy and active places. They often include high stress situations with high intensity activities and raised voices. The requirement for abuse-resistant finishes within common and private spaces is an over-arching design feature, but the units also typically include secure-rooms, de-escalation spaces, nurse care stations and consultation rooms, where the control of reverberant sound enables a more calming acoustic environment and patient privacy.

The surface weight of abuse-resistant (or high-core density) drywall assemblies often results in the need for loadbearing steel stud arrangements. While 18-20ga steel studs offer increased load capabilities, the reduction in sound isolation characteristics can result in both increased costs and loss of floor space in order to accommodate additional gypsum board linings or the inclusion of resilient channels to achieve acoustical separation. However, the use of wide flange or stiffened ridge stud systems, such as the Bailey Platinum steel stud systems, can offer increased load capacities without unduly compromising the sound isolation characteristics offered by non-loadbearing studs.

The control reverberant sound, both in open and cellular spaces relies on durability as well as absorption properties. Slatted-wood or wood-wool panels with concealed fibreglass or non-fibrous duct liners offer cost-effective alternatives to backloaded or security-locked drop-in ceiling panels. The use of recycled PET felt panels are also a cost-affective alternate to absorptive compressed fibreglass wall panels.

The use of secure rooms is routine in such units and maintaining high levels of sound isolation is often a requirement, so that other patients do not become disrupted by the noises emanating from the room. The use of double-leaf masonry walls or concrete wall constructions with an independently furred drywall lining with batt insulation, resilient floors and ceiling assemblies, with padded wall linings are commonplace, with access through a lobbied door anteroom with a fully gasketed outer door to maintain acoustical privacy to adjacencies and corridors.

2.2 Child and Adolescent Mental Health Units

Child and Adolescent units consist of many different uses, from individual to group counselling rooms, recreational and sporting activities, study and work rooms as well as residential bedrooms and welfare spaces. The acoustic requirements often dictate enhanced levels of sound isolation, promotion of calming spaces and a careful mix of low

* marks@bkl.ca

background levels in some spaces with higher sound levels in other areas to maintain speech privacy by sound masking.

During a recent project, healthcare staff within a proposed adolescent unit indicated that the main safety concerns related to both concealment, either of prescription medication or items that could be used for self-harming, and patients barricading themselves into room-spaces.

While a number of the acoustical treatments are readily replicated from Emergency and Crisis units, the design of residential bedrooms, nurse stations and activity rooms require further consideration to avoid the safety concerns.

Project specifications typically require standard speech privacy between circulation spaces and residential bedrooms. Additionally, bedrooms have to incorporate a 2-way door mechanism to prevent barricading with ligature-free fixtures. While "*wicket door*" systems offer many advantages, the relative cost per door is high. Also, some Health Authorities express a desire to have secondary narrow door mounted within the same frame in a double-egress configuration to allow for two-person intervention entries into bedrooms.

This configuration allows for a door stop so that standard bulb gasketing, albeit ligature-free with tear perforations, can be employed on the main door slab. However, due to issues with durability, door bottoms are not available with ligature free perforations and, therefore, traditional concealed mortised drop seals are required to maintain speech privacy.

While the ceilings will necessarily be formed from non-accessible gypsum, acoustical absorption can be provided by perforated gypsum panel systems with a textured spray finish, such as the CGC Ensemble, or from recycled PET felt panels, mounted as tack-boards as required.

Protective glazed nurse stations require a delicate balance of speech privacy so staff can discuss patient treatments or interventions, while allowing both visual and audio monitoring of the adjacencies. Wall mounted concealed microphones, can offer electronic audio connection to the adjacencies without adversely affecting staff privacy or unduly altering patients to the surveillance.

2.3 Continuing Care for Seniors

Residential care facilities for seniors often include integrated units for patients with dementia related symptoms, such as anxiety, confusion and disorientation, difficulty with communication and paranoia.

Sound, along with light, perception is known to be affected by dementia and can impair a patient's ability to orientate themselves in any environment. High noise levels readily lead to stress reactions including confusion, increased heart rate and fatigue from over-stimulation. While, natural sounds, such as birdsong or water sounds, or familiar music can provide a helpful stimulation and facilitate recall.

In general, the control of background noise levels in Continuing Care facilities is important with an emphasis placed on minimizing the impact from medical emergency alarms, paging systems and mechanical systems. Water features within entry lobbies or centralized gardens off bedrooms can be used to offer useful masking sounds.

Acoustical design for seniors within the general population often underplays issues related to presbycusis, the cumulative effect of aging on hearing, which can change the patient's ability to understand speech. However, it is even more important in dementia facilities, where reducing the impact of intruding noise allows for better comprehension, thus aiding understanding, and also improving privacy during treatments.

Common or dayrooms are often used for group activities that encourage participation with others through activities such as keep-fit, arts and crafts, group singing or dancing, and movie presentations. Reverberation control using both ceiling and wall treatments should be encouraged to achieve a reverberation time of less than a second wherever practicable in order to improve speech intelligibility and, thereby, inclusion into the group.

Patient bedrooms may also require specialist equipment for palliative or bariatric care. In such cases, the rooms typically require a higher density of penetrations for building or piped services, which can unduly affect the integrity of sound isolating assemblies or limit the incorporation of absorptive treatments. Careful detailing and selection of patient hoists is required to minimize noisy events which may agitate patients both in the room and within adjoining spaces.

As with adolescent care, double door-sets are often required in Continuing Care facilities for patient interventions. Such doors limit the potential for gasketing to provide adequate sound privacy. However, slightly increasing the sound levels within the corridors with colored noise to maintain steady background sound conditions helps to mask noisy activities and, thereby, minimize awakenings.

3 Conclusion

While solutions for acoustical treatment in mainstream healthcare are well-rehearsed and typically adequate for the patient population, the treatment of patients with mental or behavioural conditions often requires greater care and detailed consideration.

The welfare and safety of both patients and healthcare providers is critical in such facilities and treatments for sound isolation or reverberation control have to be selected not only for their acoustic adequacy but also to reflect patient needs or characteristics. Acousticians and other design professionals must engage with the healthcare providers to understand the needs and controls that are key to the success of a mental or behavioural healthcare facility so that they may include innovative and robust treatments to the benefit of the patients.

References

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[3] The American Society of Heating, Refrigerating and Air-Conditioning Engineers (ASHRAE) Handbook 2019 – HVAC Applications, Chapter 49: Noise and Vibration Control.

EVALUATION OF AN INDOOR OPEN SPACE ACOUSTICAL QUALITY – A CASE STUDY

Weidong Li *, Linda Drisdelle †
Pinchin Ltd., 2470 Milltower Court, Mississauga ON L5N 7W5 Canada

1 Introduction

The occupant's comfort in an indoor space depends on several factors including temperature, humidity, air quality, lighting, views, and acoustics. Of these factors, acoustical comfort is a key contributor to employees' performance and well-being in the workspace. Acoustical comfort is achieved when the workplace provides appropriate acoustical support for interaction, confidentiality and concentrative work [1].

As a result, the testing and evaluation of HVAC attributable background noise levels have been recognized by various organizations [2-5]. In addition, speech intelligibility is also an important factor for office environments [6, 7].

In this paper, an acoustic model was developed for an existing space. The acoustic model was first calibrated and then used to predict the benefits of applying suggested acoustical improvements in the space.

2 Acoustic Testing

To verify the modelling results, acoustic testing was conducted in a warehouse space, as shown in Figure 1.

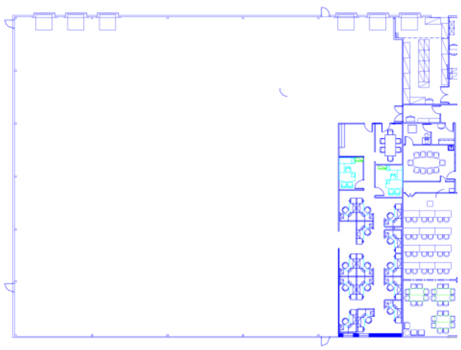

Figure 1: The Floor Plan of the Warehouse Space

A series of acoustic tests were conducted in the space. First, the HVAC attributable background noise was measured in the space. The average background noise levels without and with the unit heater operations were 44 dBA and 58 dBA, respectively.

Second, the sound pressure levels in the space were measured in the space. The measurements were used to calibrate the acoustic model. Third, impulse measurements were conducted in the space. From the impulse measurements, parameters like reverberation time and STI can be determined. The measured average reverberation time and STI were 2 s and 0.42 - 0.56, respectively.

The above measurements show that both the HVAC related background noise levels and reverberation times were significantly high for the use purposes of the space.

3 Acoustic Modelling of Existing Conditions

In this study, an acoustic model using CadnaR was built for part of the space within the warehouse, as shown in Figure 2.

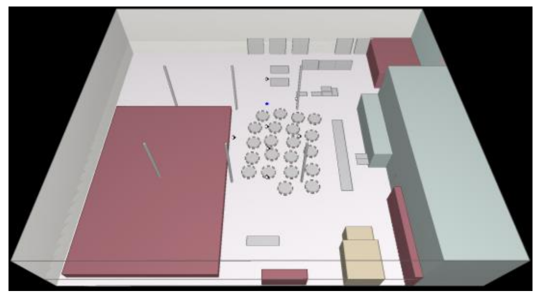

Figure 2: 3D View of the Warehouse Space

In the model, a point source representing the speaker was placed to the west side of the space. This was to simulate the sound pressure level testing. The receivers represent the microphone locations. The measured and predicted sound pressure levels are summarized in Table 1 below.

Table 1: Comparison of Measured and Predicted Levels

Testing Location	Measured Level, dBA	Predicted Level, dBA	Difference, dB
SPL-001	60.6	60.7	0.1
SPL-002	57.4	57.4	0.0
SPL-003	55.4	55.6	0.2
SPL-004	57.5	57.1	-0.4
SPL-005	60.1	59.6	-0.5
SPL-006	64.7	64.0	-0.7
SPL-007	67.9	67.2	-0.7
SPL-008	60.3	59.5	-0.8
SPL-009	57.1	57.6	0.5

Figure 3 shows the comparison between the measured and predicted reverberation time in the space. In general, both the measured and predicted results are in good agreement.

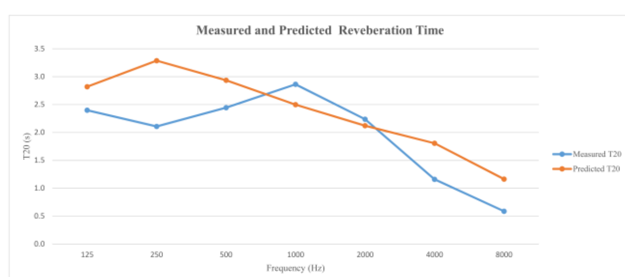

Figure 3: Comparison of Reverberation Time (T20)

4 Acoustical Improvements

A proposed treatment of absorptive materials placed around the space was incorporated into the acoustic model.

* wli@pinchin.com
† ldrisdelle@pinchin.com

Specifically, the absorptive materials considered included a carpet-type floor, 3.5 m high absorptive curtains around the space and absorptive baffles hanging at the top of the space. Figure 4 shows the 3D view of the proposed modifications to the space.

The model was used to assess the effectiveness of the proposed changes and showed that the predicted sound pressure level for receiver chains was reduced to 48 dBA, which meets the ISO 3382-3:2012 suggested value.

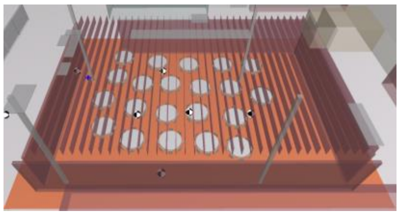

Figure 4: Warehouse Space with Proposed Treatments

The model also showed the average reverberation times were reduced to values of around 0.8 s, which are in line with the typical office uses. Most importantly, the predicted speech intelligibility for all receiver chains was increased to 0.8 - 1.0, which is a significant improvement.

5 Auralization Results

Auralization is a technique for creating audible files from numerical data. The auralization technique allows one to listen to the audio file created by the model, which would give an impression as to what it would sound like in the space. By playing and comparing the audio mixtures, it clearly shows that the treated space now has a much better speech intelligibility.

6 Conclusions and Discussions

In this study, selected acoustical parameters including sound pressure level, reverberation time, and speech transmission index were evaluated to understand the sound quality of an indoor space. The measured and predicted results showed that the untreated space had significant reflections, and rendered poor speech intelligibility.

To test the viability of acoustical improvements in the space, the calibrated acoustic model was revised and incorporated absorptive materials. The modelling results showed that the proposed acoustical treatments may be used in improving the acoustic quality of other spaces.

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The Case for Minimum Impact Noise Requirements in the National Building Code of Canada

Matthew Golden[*1] **and Roderick Mackenzie**[†2]
[1]Pliteq, Vaughn, Ontario, Canada
[2]Soft dB, Montréal, Québec, Canada

1 Introduction

Unlike the residential building codes of most developed and all other Organization for Economic Cooperation and Development (OECD) countries, the National Building Code of Canada (NBCC) does not contain any mandatory, minimum impact noise performance requirements for multi-family dwellings. This is despite a stated objective of the Code to provide "noise protection" to building occupants (Code Objective OH3). As of the latest 2015 Edition, the NBCC continues to only "recommend" that bare floors should achieve a minimum impact insulation class (IIC) of IIC 55. [1]

In Early 2016, the lead author submitted a Code Change Request (CCR) for the NBCC to make impact noise insulation a requirement in the 2020 code [1, 2]. The CCR, (CCR 1017), was modelled on the new airborne sound insulation requirements that were included in the 2015 code [2]. A new mandatory impact noise requirement was proposed in terms of the Apparent Impact Insulation Class (AIIC) metric, as well a suggested minimum criterion of $\leq$ AIIC 47 to both mirror current airborne requirements and reduce resistance to the CCR. It is noted that the submitted CCR was proposed to Codes Canada as a urgently required placeholder, in the knowledge that to truly address occupant health risks it is very likely that it may be necessary to modify the language, and possibly the metric and minimum level of protection required. The justification for the CCR is presented in Sections 2 through 4.

2 The Nature and Health Risks of Impact Noise in Dwellings

Typical sources of residential impact noise transmitted between dwellings include footfall (walking, running, jumping) noise; furniture being moved; objects being dropped; vacuum cleaners and appliances; doors or cupboards closing; and plugs going into or out of sockets. A study by Park et al. determined that approximately 80% of all measurable noise events (airborne and impact) recorded between dwellings separated by concrete floor partitions were either footsteps, movement of furniture, or dropping of small items [3]. Even with floating floors and suspended ceilings, many impact sources exceeded 45 dBA (LAmax), occurring throughout the day and night [3].

Canadian and International studies concerning occupant perceptions have shown that impact noise in multi-family dwellings is a major cause of complaint; accounting for up to 50% of all residential noise complaints [4], with footfall noise being the most annoying for occupants [5] and coming 2nd only to stereo music for noise generating moderate-to-high levels of concern [6]. The National Research Council of Canada (NRCC) observes that impact noise is the "Number one source of complaint by building occupants" [7].

Occupant exposure to impact noise in dwellings also carries significant short and long-term health risks, mostly indirect physiological and psychological consequences associated with persistent sleep disruption (including but not limited to stress, fatigue, behavioural problems in children, hypertension, cardiovascular disease, colorectal cancer, and increased mortality), and mental health consequences associated with annoyance (including but not limited to increased stress, depression, and decreased social well-being and intra-family relations) [8–12]. An expanded summary of the health risks associated with sleep disruption and annoyance, as well as the references for such studies, can be found in the authors previously published paper. [13]

3 2015 National Building Code of Canada Requirements

3.1 NBCC 2015 Code Objectives

The NBCC presents the technical provisions for the design and construction of new buildings, including multi-family dwellings [1]. Health Code Objective OH3, Noise Protection, has a stated objective to "limit the probability" that, as a result of the design or construction of the building, a person in the building will be exposed to an unacceptable risk of illness due to high levels of sound originating in adjacent spaces in the building [1]. A sub-clause, OH3.1, specifically restricts this exposure only to airborne sound transmitted through separating assemblies [1]. Based on the physical and mental health risks previously discussed concerning sleep disturbance and annoyance, to adequately meet the current code objective OH3 to protect occupants from noise-induced illness, it is necessary to require protection against all relevant sources of noise, including impact noise.

3.2 NBCC 2015 Current Provisions for Impact Sound Insulation

In Appendix A-9.11.1.4.- Sound Transmission, the NBCC 2015 confirms that there is no requirement for the appropriate control of impact noise, and yet confirms that: "Footstep and other impacts can cause severe annoyance in multi-family residences" [1]. The Section then continues that "Builders concerned about quality and reducing occupant complaints will ensure that floors are designed to minimize impact trans-

*mgolden@pliteq.com
†r.mackenzie@softdb.com

mission. A recommended criterion is that bare floors (tested without a carpet) should achieve an impact insulation class (IIC) of 55" [1].

Despite the NBCC containing no subsequent guidance for the user on how to achieve this criterion (as no floors in the example floor tables provided meet this level), IIC 55 is comparable to international peers (See next section). However references such as [14] cite studies showing that for the equivalent of AIIC 55, 47% of occupants found children on the floor above to be annoying, but for a lightweight floor also equivalent to AIIC 55, the annoyance rate was 80% [14]. They also cited another study showing a 60% occupant annoyance rate for walking noise when the separating floor was the equivalent of AIIC 52 [14]. Additionally, because there is no prescribed mandatory requirement for impact noise, dwellings are currently being built, tested, and lived in that have an AIIC in the low 30's, but because the ASTC is above 50, this is deemed acceptable (although clearly not to the occupants).

4 International Comparisons

All other OECD countries have already recognized the need for an impact noise insulation requirement as part of their respective building codes. In some European countries, impact noise requirements have been part of the national building regulations since the 1940s. In terms of minimum criterion; Austria, Finland and Germany are the most restrictive at approximately AIIC 62, whereas most Nordic and Baltic countries generally set the minimum around AIIC 55-60, and other European nations around AIIC 50-55 [11]. Closer to Canada, the International Building Code used in the U.S. specifies a mandatory minimum requirement for impact insulation (IIC $\leq$ 50, NISR $\leq$ 45), whilst the International Code Council G2 2010 Guideline for Acoustics recommends $\leq$ IIC 55 as an acceptable performance grade, whilst IIC 60 would be the preferred grade [15]. Due to the widespread use of impact insulation requirements, there is considerable knowledge and data regarding cost-effective solutions to achieve satisfactory impact insulation performance.

The Canadian Morgage and Housing Corporation propose a minimum AIIC 55 as acceptable for most multi-family dwelling rooms with hard floors, with AIIC 50 was suggested for ceramic floors [16]. The NRCC also promotes IIC 55 as a best practice criterion [17], with IIC/AIIC 50 deemed the minimum rating for occupant satisfaction [18]. Again, these are only recommendations, not requirements.

5 CCR proposed Additions

Based on the aforementioned issues, there is a clear need to address the omission of minimum impact noise criteria from the NBCC, hence the submitted CCR. The homebuilding industry should be assured that this would be implemented in a considerate and sensitive manner that takes account of existing practices with only incremental changes. The proposed minimum requirement in the CCR is deliberately set rather low compared to most other members of the OECD (AIIC $\leq$ 47), so as to allow Canadian industry to adjust to the new requirement and to align with building code regulations in the U.S. Note however that most people will not find this level satisfactory for subjective impact sound insulation. The IIC ratings for the most common floor construction designs have already been measured in the past and are readily available in datasheets or publications (for example by the NRCC). Overall, it is expected that the enforcement implications for this code change will be minimal; compliance can be demonstrated through field testing, prescriptive procedures using the Tables, or the Calculation procedure in the same manner and at the same time as is currently performed for ASTC/STC. The following changes to the NBCC were submitted as part of the CCR [2]:

1. Add new Code Sub-Objective OH3.2, "Exposure to impact sound transmitted through assemblies separating dwelling units from adjacent spaces in the building".
2. Add new Sections 5.8.2 and 9.11.2 "Protection from Impact Noise"; this essentially contains the same requirements as the airborne noise sections, but with all references to "airborne noise" and the associated standards replaced by "impact noise" and the associated standards.

6 Current Status of the CCR

Since the NBCC is an objective-based building code, the core objectives must be expanded. The process to change the objectives is detailed in Appendix L of the Policies and Procedures of the Canadian Commission on Building and Fire Codes (CCBFC) [19]. Through this process OH3 will be expanded or modified to include impact noise. The CCBFC sets the work plan for the Standing Committees (SC), and have assigned the CCR 1017 to the Standing Committee on Environmental Separation (SC-ES) as the lead committee joint with the Standing Committee on Housing and Small Buildings (SC-HSB) [20]. Once that process is complete the SC-ES and SC-HSB can begin the work of changing the NBCC through the code change request following the procedures is in Appendix F Policies and Procedures of the CCBFC.

In a recently published meeting agenda, the Joint Task Group on Impact Sound (JTG-IS) published an Agenda Item Summary Sheet [21]. Both the SC-ES and SC-HSB have given this task a high priority for completion in the next Code Cycle. To their credit, this joint task force has identified the issue well. They state that; Failure to address impact noise will (i) put the burden of addressing sound insulation performance on professionals/designers with no (or little) harmony in approaches and level of performances across the country; (ii) support design approaches that may result in investment in the wrong elements and most importantly may fail to address the system function that establishes a minimum acceptable performance necessary to satisfy the Objective OH3–Noise Protection; (iii) Leave Canada lagging most industrialized countries; and (iv) Leave Canadians at risk for adverse health effects (increased stress, compromised immune systems and depression) as recognized by the WHO.

This JTG-IS has also clearly identified their mandate to: Identify and review documents related to impact sound; Review codes and regulations from other jurisdictions; Review recent research findings; Review Provincial/Territorial and International requirements in terms of construction, design, operation, use, etc., where applicable; Report to the parent standing committees if no changes to the current requirements are required; Identify areas where additional information is needed and report to the parent standing committees if changes to the current requirements are warranted but there is insufficient information to support requests for changes; Recommend changes to the parent standing committees where changes to the current requirements are considered appropriate and there is sufficient information to support requests for changes; and coordinate between parent standing committees to maintain harmonization between the National Model Code provisions. Unfortunately, the JTG-IS estimated that the work to include impact noise into the NBCC will not be completed until September 2022. This means that impact noise will not be addressed in the 2020 NBCC, but would ”likely” be included in the 2025 NBCC, provided that they find that (i) changes are required, and (ii) sufficient information has been gathered. However this offers no guarantee that the changes will happen by 2025, only that there is a willingness to see the changes made. Even then, at best most residential buildings will not see any impact noise requirements enacted until 2032 at the earliest.

7 What you can do to help

There are a few things that the acoustical community can do to help. The first is to join the observers page for the NBCC. Then you can help collect the information that the JTG-IS will need to complete their work. Additional work that is encouraged to make the case includes quantifying the scale of the problem by asking acoustical consultants and large condo boards for anonymous failure rates found in new-build constructions. Repeating the studies of Park *et al.* to determine the apparent sound level (LAmax) that results from different objects being dropped, furniture moving, and footfall on different floor assemblies typically built in Canada would be of use in comparing against LAmax criteria for the protection of sleep. Finally, you can help raise public awareness to prevalence of impact noise in the general public; more articles and discussions in general publications about the pervasive issue of inadequate impact sound insulation protection will help document the need for these code changes. This may give encouragement to the Standing Committees to complete the work on time so as to include the code change in the 2025 NBCC.

It is also noted that historically the Canadian Acoustical Association (CAA-ACA) was active in passing motions at their AGMs in the 1980s calling for Code Changes to bring airborne sound insulation requirements to STC 50. We would encourage the CAA to again pass such a motion on the need for impact noise requirements, and proactively lobby for such as change on behalf of Canadians.

Acknowledgments

The authors wish to thank their employers for giving them time to work on this topic outside of there normal duties.

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COMPLEXITIES OF CURTAIN WALL FLANKING TRANSMISSION - A CASE STUDY

Kelly Kruger *[1], **Robert Ogle**†[1]
[1]Building Science Engineering, Edmonton, Alberta, Canada

1 Introduction

Aluminum and glass curtain wall construction is very common in modern commercial buildings. The continuous external façade can lead to significant limitations on the sound isolation between adjacent rooms, separated either laterally or vertically. Flanking transmission of curtain wall systems is very difficult to predict at the design stage of a project. Manufacturers do not routinely measure this parameter. The scarcity of data is partly due to a lack of laboratories that have the necessary specialized test environment. Generic flanking transmission loss data is of little value because of the large variation in curtain wall assemblies. Designers are often left with best guess approximations based on previous experience. These factors also make it difficult to significantly improve flanking transmission of an existing curtain wall installation.

The authors were recently involved in the acoustic commissioning of a university building where the sound isolation between floors was considered to be critical. A Noise Isolation Class of NIC 55 - 60 was required in order to allow music practice rooms to be located directly below conventional lecture rooms. Adequate acoustic isolation should have been achieved if the sound transmission was entirely determined by the 200 mm thick, post-tensioned concrete floor that separates the classrooms. The initial sound isolation, NIC 42, was found to be well below expectations, as shown on Figure 1.

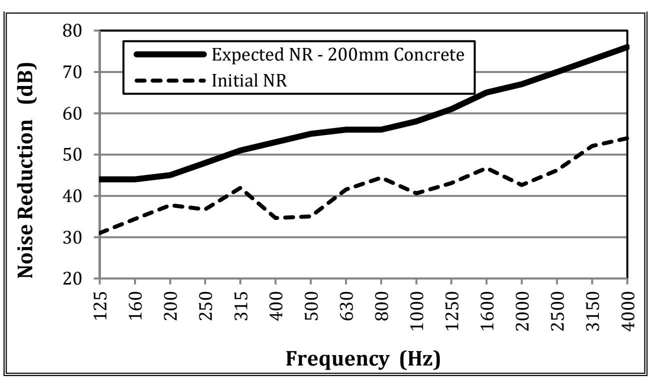

Figure 1: Expected vs measured noise reduction

It was evident that there was significant flanking occurring at or near the building façade, made up of a unitized curtain wall assembly. Due to the complexity of the junction between the curtain wall and the floor structure, several potential flanking paths were identified. It was possible to alter each flanking path individually so that the incremental improvement of each step could be quantified. Figure 2 shows a photo of the curtain wall and the construction detail at the intersection of the curtain wall with the floor.

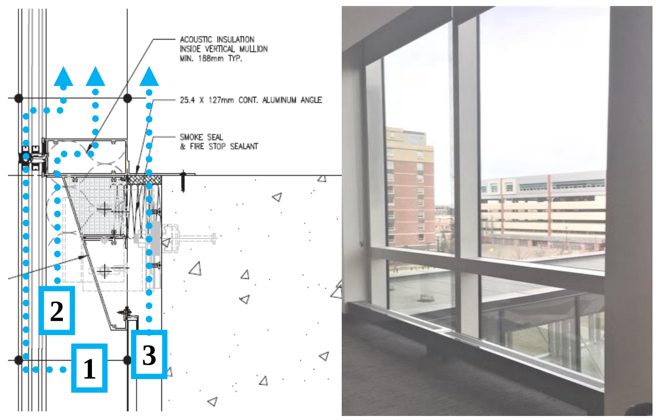

Figure 2: Curtain Wall Flanking Paths.

Flanking can occur along several paths. Path 1 shows the transmission from the glazing on the lower floor level through the spacer and into the glazing on the upper floor. Path 2 shows the transmission directly through the horizontal mullion. Path 3 shows the transmission through the gap between the curtain wall and the floor slab.

2 Method

A series of sound isolation tests were conducted between floors. The receiving room measurement positions were located approximately 2 m from the curtain wall due to the large influence of curtain wall flanking. Also, the receiving room had numerous desks located very near the curtain wall and so the receiver locations are indicative of the worst case sound isolation experienced by the room occupants.

3 Results and discussion

An inspection of the curtain wall/floor junction revealed that the fire stop had not been fully installed (Flanking Path 3). This left a significant opening at the edge of the floor slab that was only blocked by carpeting. It was assumed that this gap was the most significant flanking path. It was therefore dealt with first by inserting two layers of fire resistant, mass loaded joint filler, with mineral wool inserted between layers. Treating Flanking Path 3 provided a modest improvement to NIC 46.

Next, vibratory velocity measurements were taken on the various components of the curtain wall. Measurements were taken on the horizontal mullions, vertical mullions and glazing for both the lower and upper sections of the curtain wall. The sound power radiated by each component was calculated by accounting for the exposed surface area.

* kelly@bse.ab.ca
† Robert.ogle@bse.ab.ca.
‡

Overall the glazing was the most significant contributor, although the contribution from the lower horizontal mullion was comparable in a few high frequency bands, most notably above 500 Hz. The contribution from the vertical mullions was generally much less significant throughout the frequency range. The vibration measurements also revealed that there was substantially less sound power radiated from the upper section of the curtain wall compared to the lower section.

To verify that the sound power radiated from the horizontal mullion was significant, at least in some of the higher frequency bands, the bottom mullion was clad with a mass loaded vinyl (MLV) sheet with a surface density of 15 kg/m^2. This provided an improvement in noise reduction of up to 6 dB in the frequency range identified by the vibration analysis. The mass on the mullion was increased further by applying a layer of 16 mm drywall on top of the MLV. This did not provide an additional improvement in sound isolation.

The remaining and now most significant flanking path was along the glazing, Path 1. The vibration analysis indicated that the glass in the lower section of the curtain wall radiates approximately 10-15 dB more sound power than the larger upper section of glazing. This prompted the idea of adding an additional glass panel on the inside of the lower section of curtain wall. As proof of concept, a temporary drywall panel was installed at this location. The improvement in noise reduction that was provided by the drywall panel was promising. However, it was recognized that because the new inner glass panel would be manually glazed in the field some air leakage from the room interior into the cavity between the new glass panel and the existing insulating glass (IG) unit might occur. Since the presence of the new glass panel would reduce air and heat flow to the inside surface of the IG unit, the surface temperature would be reduced. Therefore there was a concern that condensation might occur under certain environmental conditions.

In order to confirm if condensation would be problematic, temperature probes were used to monitor the glass temperature of both a typical existing IG unit, and an adjacent unit covered using gypsum board of roughly the same insulation value as the proposed inner glass panel. Data loggers were also used to measure the interior and exterior relative humidity (RH) for a period from December 20^{th} 2018 to January 10^{th} 2019. During this time, a minimum glass temperature of 6.5° C and a maximum dew point temperature of 0.2° C were measured (not concurrently) and are shown on Figure 3. Maximum interior RH levels during both occupied and unoccupied classroom times were found to be less than 25%. Using the collected data, a "worst case scenario" of a -30° C exterior temperature with interior conditions of 25° C and 25% RH was evaluated. In this scenario the dew point of the interior air was 3.6° C and the minimum glass temperature was 5° C. Therefore under these design conditions, condensation would not occur.

Given these findings we concluded that the introduction of the interior glass panels would not result in condensation problems.

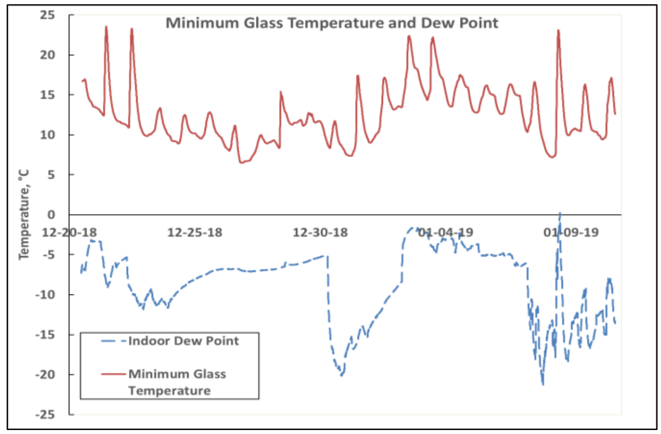

Figure 3: Minimum insulated glass temperature vs. dew point

Finally, a new glass panel was installed to cover the entire lower section of the curtain wall, thereby reducing the contribution of the last major flanking path. The lower mullions were clad with a layer of MLV that incorporated a sound absorptive foam layer exposed to the cavity between the curtain wall and the new glass panel. A significant improvement in sound isolation (NIC 57) was achieved.

4 Conclusion

Figure 4 summarizes the improvement that occurred after each step of mitigating the flanking through the various paths. Some flanking from the curtain wall is still evident but any further modifications to the existing construction are no longer thought to be practical.

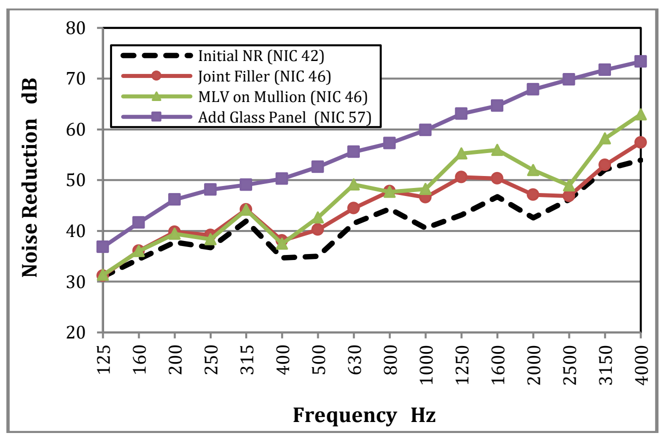

Figure 4: Improvement due to treatment of Flanking Paths

Acknowledgments

The authors would like to thank Mr. Frank Salopek of MacEwan University for his considerable assistance throughout the investigation and for his many key suggestions on implementing practical solutions.

RECENT EXPERIENCE WITH ACOUSTICAL PRIVACY CONSIDERATIONS IN ACADEMIC SETTINGS

Mandy Chan, and William J. Gastmeier.
HGC Engineering, Mississauga, Ontario, Canada

1 Introduction and Background

The provision of increased levels of acoustical privacy and confidentiality has become an ever more important consideration in the design and functionality office spaces and interview rooms. Occupants have always had a desire for freedom from distraction and an enhanced feeling of privacy, but with the advent of privacy of information legislation, confidential levels of privacy are an expectation.

In academic settings such spaces involve exam rooms for special needs students, mental health offices in student services facilities, cooperative education and other interview rooms and registration offices where sensitive information may be discussed.

This article discusses two investigations related to reportedly poor levels of acoustical privacy in academic settings with regard to applicable criteria, suitable acoustical measurements and analysis methods and contributing factors.

2 The Spaces

Both situations are typical private offices with carpet, suspended fibreglass acoustical tile or drywall ceilings, desks bookshelves and chairs. There were no other specific acoustical treatments. They both had similar wall partitions comprised of 5/8 inch Gypsum Wall Board (GWB) on each side of 3 5/8 inch metal studs with batt insulation in the inter-stud cavities. Such a construction typically has a Sound Transmission Class (STC) rating approaching STC 45 in the field and is generally acceptable in providing good privacy between enclosed offices for normal speech situations. In bot h cases our inspections indicated that the wall assemblies were built with industry standard construction techniques with no obvious deficiencies.

Supply, return and transfer ducts appeared to be provided with internal acoustical lining and were not observed to be a significant path of sound transfer in either case.

The common complaint was that conversations were audible from adjoining offices to varying degrees depending on voice level and with varying levels of intelligibility, all of which led to a feeling of poor privacy, and particularly where conversations were expected to be confidential.

2.1 Engineering Faculty Offices

The gypsum partitions separating the offices from each other extended 100 mm above the finished ceiling but did not extend to the underside of the ceiling stab. Batt insulation was placed between the metal studs above the acoustical tile ceiling.

* machan@hgcengineering.com
† bill@gastmeier.ca

The weak link in this case resulted from a Value Engineering exercise in which the drywall was omitted from the studs above the suspended acoustical ceiling which was comprised of a thin fibreglass tile not rated for sound transmission. This allowed sound to travel up and over the wall partition through the batt insulation ceiling tiles. A ceiling tile with a CAC rating of at least 35 is needed to provide even a basic level of privacy between enclosed offices if there is no effective plenum barrier in place.

Speech in a normal voice was audible and intelligible depending on voice level and orientation. Loud speech was clearly audible and intelligible. Staff felt like they had to speak in quiet voices to maintain some level of privacy, and that is generally not an acceptable condition for most individuals who occupy private offices.

Physical remedial actions would involve replacing the ceiling tiles with a high CAC tile and/or extending the drywall to the underside of the slab and sealing all penetrations. Drywall construction was not considered to be viable by facilities staff, and the installation of a sound masking, was also a consideration.

2.2 Business College Interview Rooms

In this case, the GWB partitions separating the rooms did extend from slab to slab demising them from adjacent spaces. A glazed wall partition and solid core wood entry door demised them from the corridor and an exterior waiting room.

Penetrations above the ceilings were found to be sufficiently well sealed. There were no obvious dominant paths of sound transmission between adjoining spaces indicating that the demising constructions comprise a functionally effective isolation system with no weak links. Door seals had not been installed. The gaps around the doors were the primary contributors to sound transfer into the corridor and an adjacent waiting area.

Since there are no specific weak links with the exception of installing missing door seals, specific remedial actions related to these demising constructions were not indicated. Upgrades to all components of the constructions would be required to effectively increase the sound isolation.

Conversations in a normal voice were audible, but generally unintelligible. Conversations in a raised voice were understandable to some degree, depending on the talker and the associated voice level, particularly into the corridors and

3 Measurements and Analysis

Physical inspections were conducted and measurements were undertaken to determine the degree of sound transmission between the spaces and also to record the levels of background HVAC noise within the spaces themselves.

3.1 Sound Transmission

Standardized field tests to determine the ratings of the demising constructions were performed in accordance with the protocols of ASTM standard E-336 [1]. This test is performed by placing a sound source (pink noise) in the "source room" and measuring spatially averaged sound levels in that room, and in the adjacent "receiver" rooms.

Because ASTM 336 was generally developed to be used in residential settings, the sizes and configurations of the rooms under test did not allow for the calculation of the STC or the more current ASTC rating. For this reason these measurements are used to calculate the Noise Isolation Class (NIC) rating as per ASTM Standard E-413[2]. The results are summarized in Tables 1 and 2 below.

Table 1: Faculty Office NIC Summary

Source Room	Rec Room	NIC Rating
204C	204B	33
204E	204D	29
204Q	204P	29
204Q	204R	28
335A	335B	29
335A	335F	29
435F	437B	37
435F	437C	30
435F	437G	29

Table 2: Business College Source Room 305

Receiver Room	NIC Rating
304	36
306	42
307	38
308	44
312	51
313	41
Hallway	28
326	34

The results indicate significantly lower NIC ratings in the faculty offices as would be expected from the constructions and the subjective audibility ratings. Nevertheless, both situations suffered from a perceived lack of privacy. For this reason, we question the use of the NIC rating system alone to provide suitable criteria to address privacy concerns.

3.2 Background HVAC Sound Levels

The other factor which affects speech intelligibility is the level of background sound, generally determined by using the Noise Criterion (NC) rating system.

In office settings where some level of background sound is helpful in reducing audibility and increasing the levels of perceived privacy, the simpler dBA rating system is also useful. A background sound level in the range of 40 to 45 dBA is appropriate in spaces for face to face communication where acoustical privacy is a consideration.

The background sound levels in all the rooms were found to be quite low, generally from 25 to 35 dBA, indicating that low background sound levels are a factor contributing to a perception of low interoffice speech privacy.

3.3 Speech Intelligibility Index

The Speech Intelligibility Index (SII) [3] is a measure of speech privacy which combines an analysis of sound transmission and background sound levels to achieve a more reliable and understandable measure of speech intelligibility.

Two levels of speech were used to calculate the SII for several of the interview rooms; the normal speech level and the raised speech level. The normal speech level is the normal vocal effort two people would use when talking if they were in an environment where they are not concerned about speech privacy. It is also a useful tool to demonstrate the relative improvements afforded by remedial actions, including the installation of a properly designed sound masking system.

An SII value of greater than 0.3 represents poor or no privacy. An SII value of less than 0.1 generally indicates confidential privacy. Representative SII results are shown in Table 3.

Table 3: SII Business College Source Room 305

Speech Level	Rec. Room	SII	
		No Sound Masking	With Sound Masking
Normal	304	0.03	<0.01
	307	0.02	<0.01
	Corridor	0.30	0.13
Raised	304	0.24	0.09
	307	0.20	0.05
	Corridor	0.53	0.38

4 Conclusion

The results of this study suggest that SII analysis methods provide a reliable and understandable way of investigating acoustical privacy and the relative benefits of remedial measures, including the installation of a properly designed sound masking system which can increase the perceived levels of privacy substantially.

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[3] ANSI S3.5-1997 "Methods for Calculation of the Speech Intelligibility Index".

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Sound Attenuation Of A Wood Noise Barrier Using A Meta-Material Approach

Umberto Berardi

The study of metamaterials represents a new line of research in the field of applied acoustics and noise control. Although the first studies in this sector date back to over half a century ago with V. Viselago followed by the results of J. Pendry. In this work a brief description of the state of the art of the metamaterial for acoustic applications is presented and a preliminary study on a noise barrier made of wood using metamaterial approaches is presented. In this first study, the barrier was built on a scale of 1:10, with the help of wooden rods with a cylindrical section 30 cm high and 1.5 cm in diameter. The overall length of the barrier is 100 cm. The final geometry of the barrier consists of four rows of cylindrical rods. The sticks are interspersed with an empty space equal to a distance of the diameter of the sticks themselves. This creates a regular geometry typical of a structure built with metamaterials.

Renovation Of The Northern And Southern Alberta Jubilee Auditoria: A Case Study On The Impact Of Orchestra Pit Renovation On Acoustics

Ellen Buchan

The orchestra pit renovation at the Northern and Southern Alberta Jubilee Auditoria was completed in 2019. Surfaces in the orchestra pit were changed as part of this renovation. Objective acoustical measurements were performed throughout the halls after the completion of the renovation. Measurements were acquired in audience, stage and pit areas. The results of these acoustical tests are presented and compared to the results measured in the halls after the last renovation in 2005.

Comparison Of High-Performance Floor And Ceiling Isolation Systems: Are Springs Always Better?

Matthew Golden, Wilson Byrick

There are several suppliers of high-performance floor and ceiling isolation systems. While some manufacturer have tested their products in independent third-party accredited laboratories others have not. Even the ones that do test their products rarely test them head to head with other construction methods. The authors have conducted the first known laboratory test of a spring jack-up slab as well as previously tested spring hung ceilings. Other ceiling construction methods include sound clips, resilient channels and resilient brackets. Other floor isolation systems include discrete isolation pads as well as continuous rebounded recycled crumb rubber mats. Tests were always conducted in the same lab and, when possible, within a short period of time to eliminate issues of reproducibility and rebuild-repeatability. Comparisons will be made among systems with respect to performance and constructability. Systems will also be compared to basic transmissibility theory.

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ENGINEERING AND PHYSICAL ACOUSTICS - GÉNIE ACOUSTIQUE ET PHYSIQUE

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A Comparative Evaluation of Hand-Arm Vibration Impacts

Weidong Li *, Linda Drisdelle †
Pinchin Ltd., 2470 Milltower Court, Mississauga ON L5N 7W5 Canada

1 Introduction

As per the applicable ISO standard ISO 5349-1 [1], hand-arm vibration is typically measured with a tri-axial sensor attached to the fingers, palm, or arm. Depending on the measurement locations, applicable adapters may be inserted between the sensor and hand/arm. For hand-arm vibration measurements, the frequency weighting filter W_h is specified in ISO 5349-1, European and American guidelines [2, 3]. The frequency weighting filter W_h is comprised of three filters: high-pass, low-pass and frequency-weighting [4].

$$W_h = H_h(s) \cdot H_l(s) \cdot H_w(s) \tag{1}$$

Where H_h, H_l, and H_w are high-pass, low-pass, and frequency-weighting filters, respectively. The individual filters are defined in ISO 2631-2 [5] and can be calculated using the following equations.

$$H_h(s) = \frac{s^2}{s^2 + \frac{\omega_1}{Q_1}s + {\omega_1}^2} \tag{2}$$

$$H_l(s) = \frac{{\omega_2}^2}{s^2 + \frac{\omega_2}{Q_2}s + {\omega_2}^2} \tag{3}$$

$$H_w(s) = \frac{(s + \omega_7) \cdot {\omega_8}^2}{(s^2 + \frac{\omega_8}{Q_7}s + {\omega_8}^2)\omega_7} \tag{4}$$

Figure 1 is a plot of the frequency weighting curve W_h over the frequency range which shows the weighting network having the lowest attenuation at about 10 Hz. At frequency ranges lower than 10 Hz, an attenuation up to 30 dB can be achieved. At higher frequency ranges, the attenuation can reach down to 40 dB (at 1000 Hz). This means that most of the energy outside the "favourable" frequency band (i.e. approx. 10 Hz) is significantly attenuated due to the filtering.

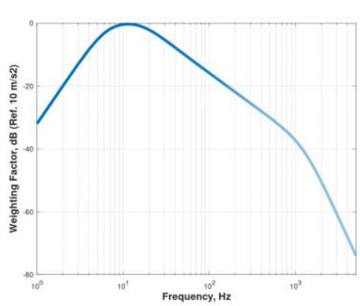

Figure 1: Frequency Weighting Curve Wh

Therefore, unless the vibration event has the dominant energy centred around 10 Hz, the vibration energy will be attenuated by the weighting curve. Consequently, the measured and calculated hand-arm vibration values, as per the ISO and other similar standards where the frequency weighting curve W_h is used, will likely be lower than the actual vibration levels as perceived by the person using the tool. In our opinion, this may be a concern because it could be leaving workers at higher vibration risks than is reported.

2 Method

In this comparative study, hand-arm vibration results between the filtered and un-filtered vibration impacts were compared in the time and frequency domains. The hand-arm vibration data was measured from the uses of two grinders, one air tool, and one hand tool. A tri-axial vibration sensor with appropriate adapters were used to measure the hand-arm vibration. The sensor was connected to a portable human vibration meter for data storage and post analysis. As per the ISO 5349-1 standard, the frequency weighting curve W_h was used for all directions. At the same time, raw (un-filtered) vibration data was sampled at about 7.2 kHz and streamed to the meter for further analysis.

Figure 2 shows the comparison of spectra from the use of one grinder. It is worth noting that a significant amount of high frequency energy was generated from this handheld grinder. Those high frequency components were attenuated by the filtering network when reporting vibration levels as per the ISO 5349-1 standard.

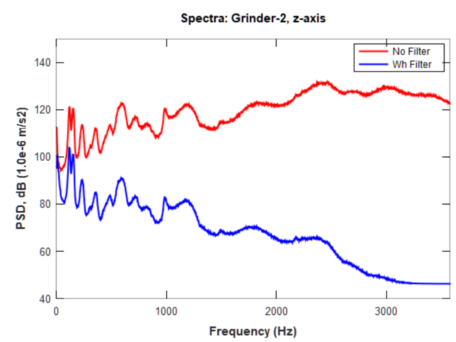

Figure 2: Spectra Comparison of Grinder #2

The frequency spectral comparison from the use of the air tool is shown in Figure 3. It can be seen that the meter "favours" the low frequency range up to approximately 50 Hz. The higher frequency components were again significantly attenuated by the filter.

* wli@pinchin.com
† ldrisdelle@pinchin.com

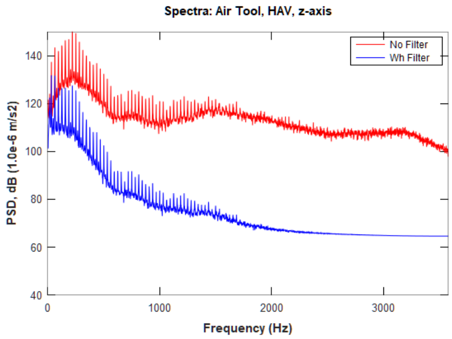

Figure 3: Spectra Comparison of an Air Tool

The comparisons of other spectra (grinder #1 and the hand tool) show similar favoritism by filter W_h.

3 Discussions

Table 1 shows the measured and calculated overall vibration energies.

Table 1: Comparison of Vibration Energies

Activity	Filtered Energy (m/s2)	Unfiltered Energy (m/s2)	Energy Ratio
Grinder – 1	0.4	3.6	12%
Grinder - 2	1.6	123.2	1%
Air Tool	11.5	96.7	12%
Hand Tool	12.8	105.4	12%

From Table 1 it can be seen that the meter measured/calculated vibration energy levels consisting of approximately 12% of the total vibration energy levels for 3 of the 4 test activities. For grinder #2, the reported energy is only 1% of the total energy.

The above analysis shows the meter measured/reported energy levels are typically lower than the total energy levels contained in the raw vibration data because of the frequency weighting curve W_h. In particular, the reported levels appear even more suppressed when the vibration energy is contained in the high frequency ranges.

To test if other parameters may be sensitive to the different types of vibration activities, statistical parameters such as Skewness, Kurtosis, Crest Factor, and Vibration Dose Value (VDV) were calculated for the above test cases. The Table below shows the calculated values for the selected parameters.

Table 2: Comparison of Additional Parameters

	Kurtosis	VDV
Grinder – 1 (Stationary)	256	196
Grinder -2 (Handheld)	3	4093
Air Tool	11	2507
Hand Tool	2308	10624

The purpose of using these values is to characterize the “peakness” of the vibration data. The VDVs show much higher values when compared against other types of vibration data that are more or less stationary and have central frequencies in the lower ranges, compatible with the frequency weighting curve W_h.

4 Conclusions

The attenuation characteristics of the frequency weighting curve W_h was investigated. The weighting curve has a favourable narrow frequency band around 10 Hz. As a result of the unique frequency response of the weighting curve, the majority of energies contained outside the favourable frequency band are attenuated. Therefore, the measured and reported vibration levels are typically lower than the vibration impacts without the filter. It is our opinion, that it may be worthwhile to look into some additional parameters in order to characterize the hand-arm vibration impacts.

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Development and Testing of an Aeroacoustic Wind Tunnel Test Section

Basim Al Tlua *[1], and Joana Rocha †[1]
[1]Department of Mechanical and Aerospace Engineering, Carleton University, Ottawa, Ontario

1 Introduction

The medium-speed wind tunnel at Carleton University was acoustically treated to create an environment for acoustical research. In order to develop aeroacoustic studies, many wind-tunnels with capacities to execute aeroacoustic tests have been designed around the world [1-3] or adapted from a pure aerodynamic wind tunnel [4, 5]. To develop the present investigation, the Carleton University aerodynamic wind tunnel has been upgraded in order to carry out this research.

2 Wind tunnel characterization

The experiments were conducted in the medium-speed, subsonic, closed-loop wind tunnel at Carleton University, as shown in Figure 1. The airflow is powered by a 37.3 kW (50 HP) variable-speed DC motor driving a 1.2 m axial propeller at speeds as high as 40 m/s. A variable frequency drive (VFD) modulates the rotational frequency of the fan at a resolution of 1Hz. A series of turbulence grids precede a 9:1 contraction, which reduces the turbulence intensity levels in the centre of the test section to less than 0.27%, as measured for speeds up to 15 m/s.

2.1 A new aeroacoustic test section

A new test section (shown in Figure 1) along with the surrounding anechoic chambers were designed and manufactured in order to be used for aeroacoustics testing. This test section is a 0.78 m x 0.51 m rectangular section, 1.83 m long. The upper and lower walls of the test section are each composed of two aluminum sheet panels and contain hardware (circle aluminum material) for the vertical mounting of two dimensional airfoil midway between the acoustic windows (i.e. test section side walls), and 0.45 m from the upstream end of the test section. The two sides of the walls of the test section are made of stretched, thin-weave cloth covering a streamwise length of 1.83 m, which provides a smooth flow surface and also a significant reduction in lift interference effects when compared to that of an open-jet test section. Cloth window allows sound to pass through the walls into the anechoic chambers with very little attenuation.

3 Anechoic system

3.1 Physical layout

The wind tunnel has an anechoic system that consists, primarily, of an aeroacoustic test section and two anechoic chambers (shown in Figure 1). The two anechoic chambers

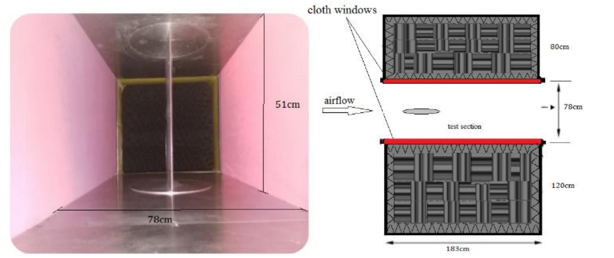

Figure 1. Left: photograph taken from downstream showing the test section interior. Right: cross section through the aeroacoustic test section and anechoic as seen from above.

are positioned on either side of the aeroacoustic test section to capture the sound emitted through the acoustic windows and reduce sound reflections inside the section.
The chambers are joined together with bolts and clamps to maintain a pressure seal. Both chambers have the same streamwise length of 1.83 m and different depths of 0.8 m right-side, and 1.2 m left-side. The chambers are lined with 0.015 m carpet bed and, 0.05 m acoustic wedged foam designed to reduce acoustic reflections.

The regions around each of the acoustic windows are covered with a carpet-bed and acoustic foam transitions to cover up all of the hard surfaces within the chamber. The chambers are each equipped with a door for access to the inside of the chamber, and for installation of data acquisition equipment. The entire system is removable so that the wind tunnel can be switched from a hard-walled configuration to an anechoic, and back again.

4 NACA0012 airfoil model

A NACA 0012 airfoil model was used as a benchmark test. The chord of the airfoil is 0.3 m, and the span is 0.51 m. The 2D airfoil is manufactured as two halves, each one composed of six pieces with eight screws. Eight holes were drilled on each side of the chord length of the airfoil, so this can be fixed on the circle rotating mechanism. The trailing edge is 0.08 m wide, and the leading edge is 0.08 m.

5 Results and discussion

5.1 Background noise measurements

The background noise of the wind tunnel was measured in the anechoic chamber with a single calibrated B&K microphone. Figure 2 shows empty test-section background sound pressure levels (SPL) in the starboard-side anechoic chamber as a function of flow speed, 0 m/s to 24 m/s. Noise levels in the port-side chamber are nearly identical. These measurements were made 1.4 m from the test-section centre. The highest spectral levels can be seen at low frequencies (100-500 Hz). Background noise levels for frequencies less than 500 Hz are mostly tones generated by the wind tunnel fan. Frequencies

* basim.altlua@carleton.ca
† Joana.Rocha@carleton.ca

above 500 Hz are believed to be due to a combination of noise sources including the fan, turning vanes, and scrubbing noise from flow surfaces in and around the test section. The peaks showed at 3 kHz, 4.5 kHz and 6 kHz are mostly associated with motor tones. To estimate the acoustic performance, the A-Weighted overall sound pressure level, OASPL, is compared to other acoustic facilities around the world.

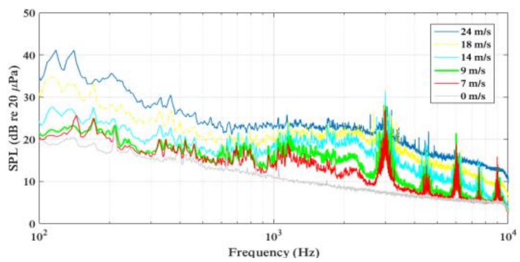

Figure 2. SPL in the starboard-side anechoic chamber as a function of flow speed in the empty test section.

Since the nozzle dimensions and microphone positions are different, the measured results must be transformed before comparing with each other. The equation is [3]:

$$OASPL_{corrected} = OASPL_{measured} - 10 \, Log_{10} \, (S / R^2) \qquad (1)$$

where R and S are the distance from the microphone to the wind tunnel centre-line and nozzle exit area, respectively.

The Carleton University wind tunnel background noise is scaled using Equation (1), and results are shown in Figure 3. Background noise of other acoustic facilities with data obtained from the literature [1, 3, 6] are also plotted in Figure 3 for comparison. Results indicate that background noise of the Carleton University wind tunnel is comparable.

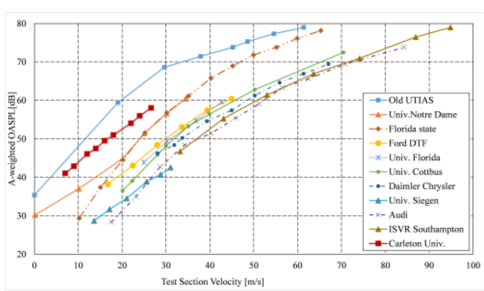

Figure 3. A-weight SPL for Carleton University wind tunnel compared with other aeroacoustic facilities around the world.

5.2 NACA0012 airfoil TE noise measurements

NACA0012 airfoil with straight and sawtooth TE was submerged within the potential core of the jet to assess TE self-noise in relation to wind tunnel background noise. The airfoil was held at zero angles of attack by side plates extended from the nozzle sidewalls. The radiated noise was measured at 1.4 m from the centre of the TE in the starboard-side, which corresponds to a 90^0 of polar angle. At first, the background noise of the wind tunnel was measured under free stream velocity of 14 m/s and 24 m/s. The airfoil with straight TE, as a reference, and the same airfoil with sawtooth TE were then attached to the sidewalls, and the same free stream velocities tests were conducted. Results for the TE self-noise spectra for these cases are plotted in Figure 4. It is shown that the serration geometry is effective in reducing TE noise. The TE self-noise measurement is seen to be more than 10 dB above the background wind tunnel noise, which guarantees the validity of the results.

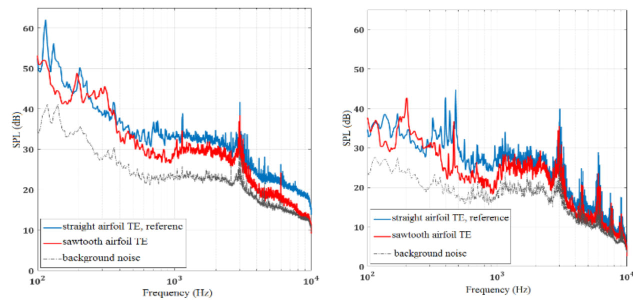

Figure 4. Measured SPL spectra for NACA0012 airfoil at free-stream velocities 14 m/s (left) and 24 m/s (right).

6 Conclusion

A small-scale aeroacoustic wind tunnel test section was built at Carleton University. The layout of the wind tunnel has been described in this paper. Aeroacoustic performance is measured and evaluated. Results show that the background noise can be comparable with other aeroacoustic wind tunnels. A simplified airfoil is tested as a benchmark test. Results show that the serration geometry is effective in reducing TE noise and that noise radiated from the TE is at least 10 dB higher than the background noise, satisfying the requirements for aeroacoustic measurements.

Acknowledgments

I would like to thank my Ph.D. supervisor Professor Rocha for all her guidance and supervision.

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Intentional Yaw Misalignment and the Effects on Wind Turbine Noise

Ian Bonsma *1, Nathan Gara †1, Brian Howe ‡1
1HGC Engineering, Mississauga, Ontario, Canada

1 Introduction

HGC Engineering completed a sound study at a wind facility in order to investigate the potential for the intentional yaw misalignment of selected wind turbines to create elevated sound levels or potentially objectionable sound characteristics, specifically amplitude modulation. The yaw misalignment was part of a study conducted by Stanford University to investigate a potential increase in the annual energy production of the site through wake control [1].

The wind facility is located in southern Alberta, and includes a number of 2 to 3 MW pitch regulated wind turbine generators. The terrain surrounding the facility consists of flat agricultural land. The prevailing wind direction in the vicinity of the facility is from the west.

The trial involved the intentional yaw misalignment by 20° of five wind turbines (Group1-T1 through T5) from October 11 to October 25, 2018.

2 Method

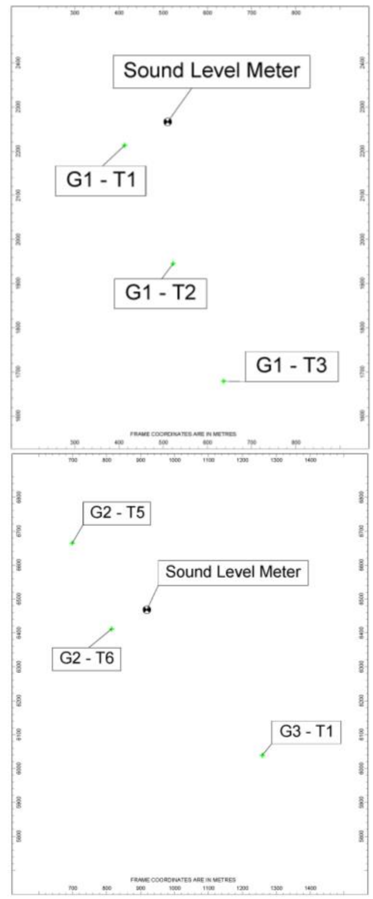
Two Svantek 977 sound level meters were installed at the wind facility between October 5 and October 29, 2018. One sound level meter was installed approximately 115 m from the base of turbine Group 1-T1 in the prevailing downwind direction. As a reference, a second sound level meter was installed approximately 115 m from the base of turbine Group 2-T6, also in the prevailing downwind direction. The microphones were set at a height of 1.5 m and equipped with 175 mm diameter windscreens to minimize the effect of wind-induced microphone self-noise. Figures 1(a) and 1(b) show the approximate location of the sound level meters in relation to the project wind turbines.

The sound level meters were configured to measure and record spectral (frequency-dependent) one-minute, A-weighted L_{EQ} sound levels. For identification of dominant sources, the sound level meters also recorded audio files.

Correct calibration of the acoustic instrumentation was verified using an acoustic calibrator manufactured by Brüel & Kjær (B&K) at the start and end of the measurement period. All equipment was within its annual or bi-annual calibration period.

The following data from the project wind turbines was utilized in the analysis:

- Wind speed at hub height,
- Yaw position,
- Electrical power generation,
- Rotor speed,



(a)

(b)

Figure 1: Sound Level Meter Location, Near Wind Turbine G1-T1 (a). Sound Level Meter Location, Near Wind Turbine G2-T6 (b).

- Blade pitch

The measured sound level data was filtered to exclude periods with inclement weather (rain and snow), gusty wind conditions and interference from birds, vehicles, and agricultural activity. Additionally, the data were filtered to only include downwind conditions (i.e. the turbine yaw position is within +/- 45 degrees from the line of sight between the measurement location and the closest turbine). The 20° yaw misalignment was taken into account for this filter.

Because of the correlation between the acoustic emission of a wind turbine and wind speed (and therefore electrical output), it is important to consider electrical output when completing a statistical analysis of wind turbine noise. Accordingly, the data were sorted into bins, 200 kW wide, based on the electrical power output of the closest turbine to the sound level meters.

The presence of amplitude modulation in the measured wind turbine noise was investigated using the methods

* ibonsma@hgcengineering.com
† ngara@hgcengineering.com
‡ bhowe@hgcengineering.com

described in the Institute of Acoustics: A Method for Rating Amplitude Modulation in Wind Turbine Noise [2]. The method uses sequential LA_{EQ} 100 ms data, calculated over three separate band-limited frequency ranges together spanning the range from 50 to 800 Hz. Each 10-second segment of data is de-trended using a third-order polynomial best fit curve. A Fourier transform is used to calculate a power spectrum, and the highest peak in a range of possible blade passing frequencies is found. The energy represented by this peak, and its possible harmonics, is used to calculate an inverse Fourier transform. Finally, the modulation depth is calculated by subtracting the L_{95} from the L_5 of the resulting time series. The method results in a series of 10-second data, as well as a series of 10-minute averaged results.

3 Results

Tables 1 through 3 present the A-weighted, energy-equivalent (L_{EQ}) sound level for each turbine power bin, as well as the number of valid data points and the standard deviation of the sound levels.

Table 1 presents the results from the sound level monitoring at wind turbine T1 in Group 1 during regular operation (i.e. not misaligned).

Table 1: Results - Turbine Group 1-T1, Regular Operation

Power [kW]	100 - 300	300 - 500	500 - 700	700 - 900	900 - 1100	1100 - 1300	1300 - 1500	1500 - 1700	1700 - 1900
Data Points	249	353	329	274	275	322	382	384	1147
L_{EQ} [dBA]	48.0	48.9	49.7	50.4	51.0	51.6	51.9	52.2	52.0
Std. Dev.	0.8	0.9	0.7	0.8	0.7	0.6	0.7	0.7	0.8

Table 2 presents the results from the sound level monitoring at wind turbine Group 1-T1 while the turbine was yaw-misaligned by 20°.

Table 2: Results - Turbine Group 1-T1, 20° Yaw Misalignment

Power [kW]	100 - 300	300 - 500	500 - 700	700 - 900	900 - 1100	1100 - 1300	1300 - 1500	1500 - 1700	1700 - 1900
Data Points	502	498	432	417	398	524	644	806	3355
L_{EQ} [dBA]	47.9	49.2	49.9	50.5	51.1	51.5	51.9	52.1	51.7
Std. Dev.	0.8	0.5	0.5	0.5	0.6	0.6	0.7	0.8	0.9

Table 3 presents the results from the sound level monitoring at wind turbine Group 2-T6 during regular operation (i.e. not misaligned).

Table 3: Results - Turbine Group 2-T6, Regular Operation

Power [kW]	100 - 300	300 - 500	500 - 700	700 - 900	900 - 1100	1100 - 1300	1300 - 1500	1500 - 1700	1700 - 1900
Data Points	1000	771	1103	936	641	682	722	679	1860
L_{EQ} [dBA]	48.2	49.3	50.2	50.8	51.4	51.8	52.2	52.7	52.6
Std. Dev.	1.3	1.2	1.3	0.9	1.0	0.8	0.8	0.7	0.7

Table 4 presents a comparison between sound level results from the regular operation of turbines Group 1-T1, and Group 2-T6, and the yaw-misaligned operation of turbine Group 1-T1.

Table 4: Sound Level Comparison

Power [kW]	100 - 300	300 - 500	500 - 700	700 - 900	900 - 1100	1100 - 1300	1300 - 1500	1500 - 1700	1700 - 1900
L_{EQ}, T6 Regular [dBA]	48.2	49.3	50.2	50.8	51.4	51.8	52.2	52.7	52.6
L_{EQ}, T1 Regular [dBA]	48.0	48.9	49.7	50.4	51.0	51.6	51.9	52.2	52.0
L_{EQ}, T1 20° yaw [dBA]	47.9	49.2	49.9	50.5	51.1	51.5	51.9	52.1	51.7

The change in sound pressure was analyzed in one-third octave bands. Where the turbine is expected to be operating near maximum sound level (1300 kW and above), the one-third octave sound level results indicate no change in any one-third octave band.

The results of the amplitude modulation investigation show no correlation between yaw misalignment and an increase in levels of amplitude modulation depth. Periods of elevated amplitude modulation depth were measured during yaw-misaligned operation and normal operation alike.

4 Discussion

The sound level results indicate that, on average, there is no statistically significant difference in the sound levels on an overall A-weighed or one-third octave basis between regular operation and yaw-misaligned operation.

A cursory review of the audio recordings collected during the yaw misalignment period did not indicate obvious changes in audible characteristics of the sound.

The results of the study indicate that the intentional yaw misalignment of wind turbines for the purpose of increased power production does not result in higher sound emission or an increase in amplitude modulation.

Acknowledgments

HGC Engineering would like to thank TransAlta Corporation and TransAlta Renewables for providing wind farm operational data and performing intentional yaw-misalignment on operational wind turbines.

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Comparison of Results from the STEAM Rail Noise Model to Potential Alternatives

Ian Matthew, *[1], Anthony Amarra, †[1]
[1]Valcoustics Canada Ltd., 30 Wertheim Court, Unit 25, Richmond Hill, Ontario, L4B 1B9

1 Introduction

This paper provides a high-level comparison of the STEAM ("Sound from Trains Environmental Analysis Method", FTA (Federal Transit Administration), and FRA (Federal Railway Administration) models with a specific focus on the emission levels/methods.

2 Background

2.1 "STEAM"

Prepared by the Ontario Ministry of the Environment (1990) [1], the STEAM model has been used for decades in Ontario and across Canada for prediction of rail activity sound levels.

2.2 Federal Transit Administration (FTA)

Originally released in 1995, the FTA "Transit Noise and Vibration Impact Assessment Manual" [2] provides an alternative model for predicting sound levels resulting from commuter/intercity rail activity (last updated 2018).

2.3 Federal Railway Administration (FRA)

The FRA released a similar manual in 1998 titled "High-Speed Ground Transportation Noise and Vibration Impact Assessment" ([3], last updated 2012) which includes Appendix E for the assessment of freight rail activities.

2.4 Comparison of Model Variables

Rather than presenting the specific sound level relationships, Table 1 presents a comparison of the variables that impact the sound level predictions for each of the models.

Table 1: Comparison of Variables in the Three Models.

	STEAM	FRA	FTA
Loco-motive	Speed Locos/period "Loading"	Speed Locos/period Type of Loco	Speed Locos/period Type of Loco Throttle
Wheel-Rail	Speed Veh/period	Speed Veh/period Type of Car	Speed Veh/period Type of Car

In the FTA and FRA models, the type of locomotive/car determines the SEL_{ref} and as such applies a speed independent adjustment to the emission level.

The STEAM model includes a term for "loading" (the number of cars per locomotive) that is unique to that model. As loading increases, locomotive sound level also increases.

3 Modelling Inputs and Scenarios

This study focusses on the relationship of speed and type of locomotive/car to the predicted sound level. Results are presented as the L_{eq} for a single train in a 24-hour period ($L_{eq,24hr}$). To eliminate any propagation characteristics, the reference distance of 50 ft (or 15 m) was used in this study.

The models assume a single rail segment, infinitely long in both directions with no variation in operation (speed, etc.)

3.1 Freight Rail Scenario

For freight rail, the STEAM model is compared against the FRA model using the "freight" parameters in Appendix E.

Both models have been analyzed using a total consist of 4 locomotives and 180 freight rail cars. Locomotives and cars are assumed to have a length of 90 and 68 feet, respectively.

3.2 Commuter Rail Scenario

For commuter rail, the comparison is based on a typical intercity train comprised of 1 locomotive and 12 coaches.

4 Model Results

4.1 Freight Rail Scenario

Figure 1 shows the locomotive and wheel-rail sound levels as a function of speed for the STEAM and FRA models.

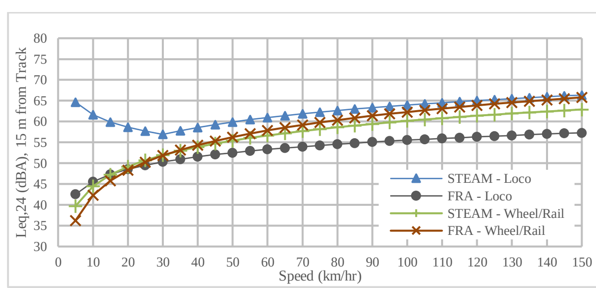

Figure 1: Freight Locomotive and Wheel/Rail Noise as a function of Speed

Above 30 km/hr, the shape of the locomotive sound level relationship is largely the same (emission level from the FRA model is approximately 7-9 dB lower than that for STEAM). Below 30 km/hr, the locomotive models exhibit opposite behaviour with regard to the speed relationship.

As shown in Figure 1, the wheel-rail trend is similar for both models (FRA model exceeding the STEAM model by

* ian@valcoustics.com
† anthony@valcoustics.com

as much as 3 dB at 150 km/hr). Note that the STEAM model uses a 15.7log(S) relationship for the wheel-rail noise where the FRA model uses a 20log(S) term.

Figure 2 shows the total source sound levels (loco plus wheel-rail) in the freight scenario as a function of speed.

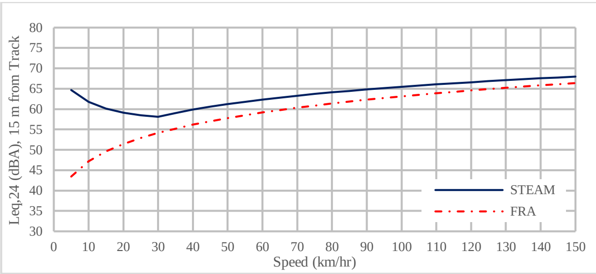

Figure 2: Total $L_{eq,24hr}$ as a Function of Speed, STEAM vs. FRA

The models agree reasonably well above 30 km/hr with STEAM exceeding FRA by up to 3.5 dB. As above, below 30 km/hr the models diverge rapidly.

4.2 Commuter Rail Scenario

Figure 3 shows the locomotive and wheel-rail sound levels as a function of speed for the STEAM and FTA models.

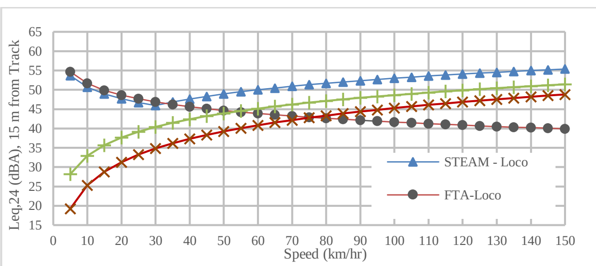

Figure 3: Commuter Locomotive and Wheel/Rail Noise as a function of Speed

In absolute terms, the locomotive sound models show excellent agreement below 30 km/hr where the STEAM model increases in sound level with decreasing speed (the comparison here assumes a throttle setting of 1). However, above 30 km/hr (where the STEAM relationship inverts), the two models diverge in their predicted sound levels.

The wheel-rail sound level trend is similar for both models in the commuter scenario. However, the absolute sound level is greater for the STEAM model by as much as 9 dB at very low speeds (with the difference decreasing with increasing speed).

Figure 4 shows the combined source sound levels in the commuter scenario as a function of speed.

Note that throttle position 1 (no correction) and throttle position 8 are both shown. Without the throttle correction (i.e. throttle 5 or lower), the models agree well below 30 km/hr.

Above 30 km/hr, the models show a significant divergence.

With the throttle correction (throttle 8) included, the predicted sound levels from FTA are up to 7 dB higher than from STEAM for speeds below 55 km/hr. Above 55 km/hr, the STEAM predictions are higher than the FTA predictions

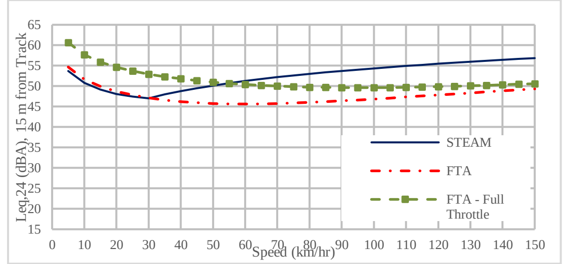

Figure 4: Total $L_{eq,24hr}$ as a function of Speed, STEAM vs. FTA

5 Discussion

The duration of a pass-by of a moving point source emitting a constant sound level will decrease as the speed increases, resulting in a lower time-averaged sound pressure level at a receptor (in simple terms, a faster train spends less time in the vicinity of the receptor than a slower train). This is reflected in the FTA reference emission levels for passenger locomotives and the STEAM model (below 30 km/hr), both of which have a 10log(1/S) relationship. However, the reference emission levels for locomotives in the FRA model and STEAM model (above 30 km/hr) increase with increasing speed. This implies that the sound energy emitted by the locomotive increases with increasing speed signify-cantly enough to compensate for the shorter pass-by duration.

This creates a significant divergence in the modelling between the STEAM model above 30 km/hr and the FTA model (although, it should be noted that below 30 km/hr, the models agree very well). In general, the models are thought not to agree well as most sound level predictions (for the purpose of assessing noise exposure) are done at speeds in excess of 30 km/hr.

Conversely, the FRA and STEAM models diverge below 30 km/hr but agree well in both shape and overall predicted sound level above 30 km/hr. As above, the models are generally in agreement in the important speed range.

6 Conclusion

With regard to the speed versus sound level relationship, the FRA model shows better agreement with the STEAM method above 30 km/hr (at least in terms of the shape of the relationship).

Conversely, the FTA model shows significant divergence from the STEAM model (above 30 km/hr).

Further study is recommended, with a focus on commuter operation, including a comparison of both model results with recent sound measurement data, in order to better understand which model provides superior agreement with real-world conditions.

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Minimizing Sonic Boom Noise to Meet Potential Regulatory Limits

John Halpenny *[1] and Joana Rocha †[1]
[1]Department of Mechanical and Aerospace Engineering, Carleton University, Ottawa,

1 Introduction

The boom produced by supersonic aircraft has prevented the introduction of new fast civilian air transports for the last 50 years. Proposed regulations would allow sonic booms of up to 75 dB of perceived noise level [1], which is very much less than the 110 dB sound signature of the Concorde, and which it is hoped will be acceptable to regulators and to the general public. Factors that affect the boom pressure and the sound levels perceived on the ground include aircraft weight and altitude, atmospheric conditions, and the aircraft shape and lift distribution. This study discusses the factors which can reduce the sound produced by a supersonic aircraft to an acceptable level.

2 Method

The code developed for this project predicts the sound pressure level which would be measured on the ground. The first step is to calculate the pressure signature in the vicinity of the aircraft, which is the near field sound level. Then, this signature is propagated through the atmosphere, taking into account the changes due to both the shaping of the shock wave and the attenuation of the atmosphere, to get the pressure on the ground. Finally, this pressure is converted to a perceived noise level (PLdB) to determine the effect on the observer. The maximum sound pressure is assumed to be directly underneath along the ground track, so this is what is calculated. The effect in the off track locations is assumed to be less.

2.1 Slender body in supersonic flight

A supersonic aircraft was first modelled as a slender cylinder in a uniform flow by Whitham,[2] which is a reasonable approximation to the long slender shapes actually used in high speed aircraft. The aeroacoustic pressure disturbance Δp at any position x and radius r from the aircraft is given by

$$\frac{\Delta p}{p_0} = \frac{\gamma M_0^2}{\sqrt{2\beta_0}} \frac{F(x)}{\sqrt{r}} \tag{1}$$

where M is the Mach number, $\beta = \sqrt{(M^2 - 1)}$, γ is the ratio of specific heats for air, and $F(x)$ is the Whitham "F Function

$$F(x) = \frac{1}{2\pi} \int_0^x \frac{S''(\xi) d\xi}{\sqrt{x-\xi}}, x > \xi \tag{2}$$

where S is the aircraft cross sectional area at station x, as determined by the aircraft geometry. Note that the F function depends only on variations of the body cross section area with length and represents the acoustic source signature. The pressure disturbance is a strong function of Mach number, but it decreases with distance only as $1/\sqrt{(r)}$, so it does not drop off very quickly.

The area function $S(x)$ represents the volume of air "pushed aside" and can be calculated for any reasonably slender shape. This produces a pressure function $P(x,r)$ at a distance r from the body, which propagates outward at the speed of sound. However, the local speed of sound will varies with pressure, so the x position of each point on the function will be changed to

$$x = \beta_0 r - kF(\xi)\sqrt{r} + \xi, k = \frac{(\gamma+1)M_0^4}{\sqrt{2\beta_0^3}} \tag{3}$$

The modified function may be multivalued - that is, different pressures may be moved to the same position. This is the characteristic of a shock wave. The sound calculation code collects these different values and assigns a unique pressure to each point. This pressure becomes discontinuous, and forms the shock waves which are characteristic of the aircraft noise signature.

2.2 Aircraft Weight

Air is pushed out of the way by the aircraft's volume as it passes, and is also pushed downward to create lift. This produces an additional shock in the downward direction which adds to the sound underneath the aircraft. Harris [3] modifies the shape function $S(x,\theta)$ to be the sum of the cross sectional area $A(x)$ and a lift term $l(x,\theta)$ such that

$$S(x,\theta) = A(x) - \frac{\beta}{2q} \int_0^x l(x,\theta) dx \tag{4}$$

$$q = \frac{1}{2}\rho U^2 \tag{5}$$

where q is the dynamic pressure at aircraft speed U and atmospheric pressure ρ . The lift term reduces the area above the aircraft for positive θ and increases it beneath the aircraft. The lift distribution is a function of the aircraft geometry but the total lift in level flight must equal the weight. The lift term varies as $1/\rho$ and so increases with altitude. The volume term does not, so the lift effect becomes dominant at high altitudes. The lift term also varies as $1/U^2$, which counters the increase in sound due higher Mach numbers and makes the total sound almost insensitive to speed. The area function is also modified by the presence of engine inlets, since the amount of air pushed aside is reduced by what is taken in. This volume is added back at the exhaust. The theoretical maximum inlet area is in the Busemann biplane configuration, where all of the volume is taken in, and the shock effect is theoretically zero.

For a given aircraft shape and weight, the S function and the total sound intensity can be calculated at a given Mach number and altitude. Using the example of Airplane B in

* john.halpenny3@carleton.ca
† Joana.Rocha@carleton.ca

Maglieri [4] where the weight is 15,500 kg and the maximum area is 4 m^2 the S function is shown with separate weight and volume contributions in figure [1].

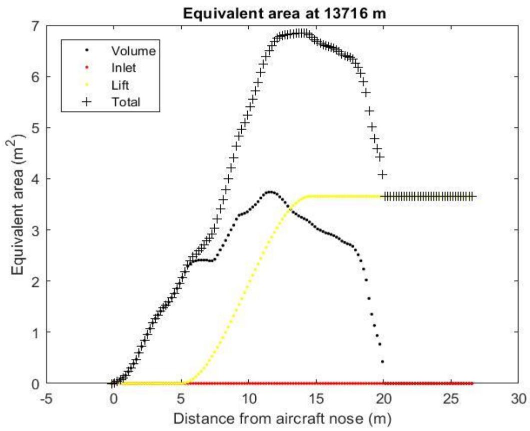

Figure 1: Contribution of volume and weight to total effective area for F-105 aircraft.

2.3 Sound Prediction

The above calculations give the sound signature in terms of a pressure profile, with sharp steps in pressure due to shock waves. The more useful result is the response of humans to these shocks, or the perceived sound volume, as determined by the frequency response of the human ear. The pressure was converted to a perceived sound level on the Stevens Mark VII scale, using the technique of Shepherd [5].

2.4 Aircraft Summaries

For conventional supersonic aircraft, the sound levels are calculated for a variety of weights and altitudes in figure 2 Weights over 100 tonnes are based on a Concorde shaped airframe, and lower weights are based on the F-105 shape as used in figure 2 Only the very lightest aircraft produce acceptable noise at 16 km altitude, and the heavier aircraft can only operate at 20 km, which is far above the maximum altitude normally used for civilian aircraft.

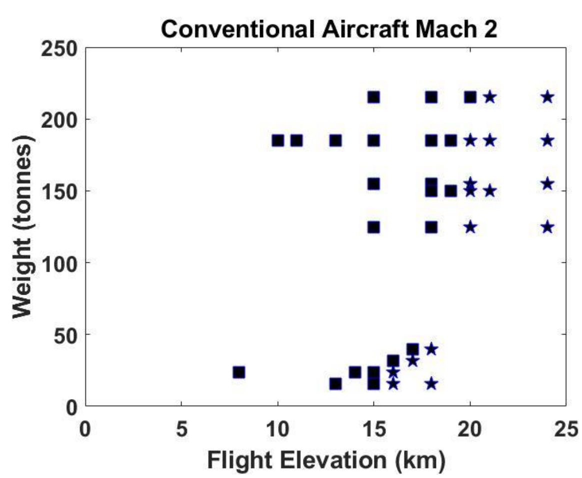

Figure 2: Conventional aircraft noise at Mach 2. Stars are values below the limit of 75 PLdB.

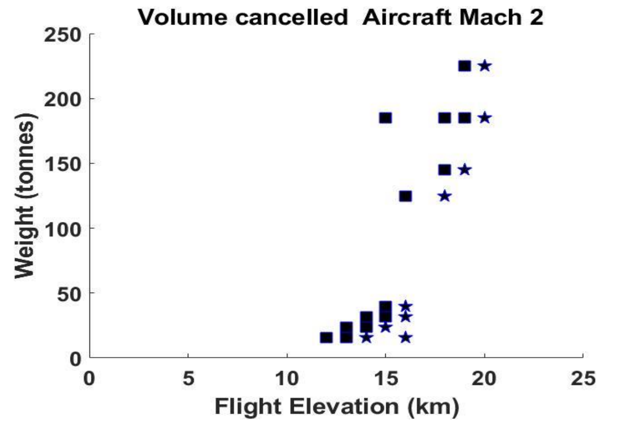

Figure 3: Theoretical volume cancelled aircraft noise at Mach 2.

To calculate the amount of sound reduction theoretically possible, the calculations were rerun without the aircraft volume term by using an engine inlet the size of the aircraft frontal area. The lift distribution has also been modified to be the entire length of the aircraft. Figure 3 shows that the acceptable weights and altitudes are slightly less restrictive, but are not dramatically better.

3 Conclusions

Using conventional technology, a supersonic aircraft flying at 2 at 16 km (50,000 feet) will have to weigh less than 40 tonnes, less than a small airliner like the A220 to meets the proposed noise standards, and one like the Concorde, with a maximum weight of 180 tonnes would have to climb to 20 km (66,000 feet) before going supersonic. A hypothetical aircraft with distributed lift and a volume-cancelling shape should be able to fly at Mach 2 at about 2 km lower**.**

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Investigation of Aircrew Noise Exposure Due to The Use of The Intercom system Onboard The RCAF CH-149 Helicopter

Victor Krupka*[1], Sebastian Ghinet†[1], Yong Chen‡[1], Andrew Price♦[1], Viresh Wickramasinghe♯[1] and Anant Grewal§[1].
[1]NRC Aerospace Research Centre, Flight Research Laboratory, Ottawa, Ontario, Canada

1 Introduction

The National Research Council (NRC) Flight Research Laboratory (FRL) conducted a series of intercom signal voltage measurements onboard several CH-149 Cormorant aircraft at Canadian Forces Base (CFB) Comox in March 2019.

The purpose of this test campaign and reporting was to support the Department of National Defence (DND) in investigating several issues related to the intercom system such as: multiple types of transitory noises reported by aircrew as well as to quantify the aircrew noise exposure levels due to the use of the CH-149 intercom system [1-5]. Transitory noise is defined as undesirable audio phenomena present in an intercom signal such as tones and squeals.

2 Method

2.1 Ground Calibration

The ground calibration was performed to characterize the CH-149 ICS (Intercom Communication System) and to develop a series of transfer functions required for the data analysis of the ICS voltage signal recordings during flight. This test procedure was defined such that the GRAS 45CB Acoustic Test Fixture (ATF) need not be flown onboard the aircraft. The co-pilot and stretcher positions onboard the aircraft were selected for ground calibration to support in-flight measurement and subsequent data analysis.

A schematic diagram indicating the equipment configuration for the intercom system characterization is shown in Figure 1. Note that for the ground calibration procedure, pre-recorded voice audio signals were prepared by NRC and used to ensure consistency in the input voice signals [6, 7].

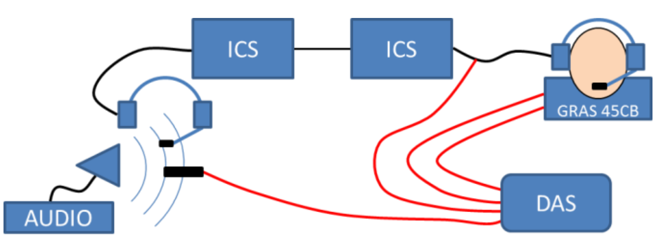

Figure 1: CH-149 Cormorant Intercom Ground Calibration Equipment Schematic [6, 7]

* Victor.Krupka@nrc-cnrc.gc.ca
† Sebastian.Ghinet@nrc-cnrc.gc.ca
‡ Eric.Chen@nrc-cnrc.gc.ca
♦ Andrew.Price@nrc-cnrc.gc.ca
♯ Viresh.Wickramasinghe@nrc-cnrc.gc.ca
§ Anant.Grewal@nrc-cnrc.gc.ca

The calibration measurements were used to generate transfer functions to characterize the dynamics of the CH-149 ICS within the audible frequency range. With the identified transfer functions using the input voice signal, transfer paths between the output voltage of the ICS and the left and right ear locations were used to accurately determine the overall sound pressure levels (OSPL); see Figure 2 for a visualization of the method. The SPLs at the ear drum locations of the GRAS 45CB ATF were estimated based on the voltage signals measured at the output of the CH-149 ICS. Therefore, by further considering the acoustic features of the ear canals of the GRAS 45CB ATF, the SPLs at the aircrew ear entrance locations were indirectly estimated so that the effect of the helicopter intercom system on aircrew hearing can be evaluated.

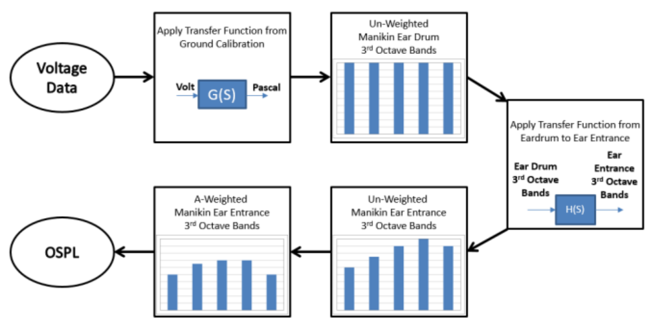

Figure 2: The method used to estimate noise exposure using intercom voltage data. [6, 7]

2.2 Flight Measurement Configuration

During the flight mission, an LMS SCADAS XS DAS was used to record the voltage signals of the CH-149 ICS at two locations: an unworn flight helmet at the stretcher location, and a flight helmet worn by the co-pilot in the cockpit location.

It is important to note that all intercom system settings for the stretcher location were consistent during the calibration procedure and the flight measurement. However, the intercom volume settings at the co-pilot location may have been adjusted by the co-pilot during the flight mission. Therefore, the data analysis presented in this paper focuses on the datasets recorded at the stretcher location to ensure consistent data comparison from both amplitude and frequency perspectives for the various flight missions.

3 Results and Discussion

3.1 Spectral Analysis of the Intercom Signals

Spectral analysis of the recorded signal voltages revealed that several types of transitory noise existed, and each type of

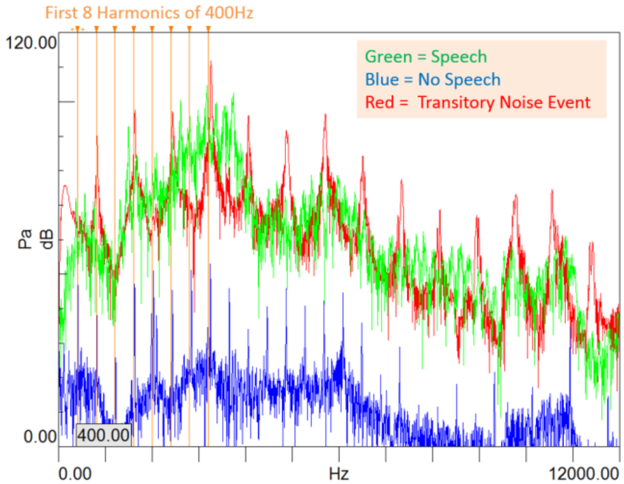

Figure 3: Comparing spectra from March 7th flight recording.

transitory noise contained acoustic energy at multiple discrete frequencies. Due to the short nature of transitory noise events, the spectral plots were generated with a frequency resolution of 5Hz. The ICS Sound Pressure Level measured in three different situations shown in Figure 3, highlights a harmonic behaviour of approximately 400Hz (signal without speech) and 800Hz during the transitory noise event. Figure 3 shows the SPL of a transitory noise event which occurred on the March 7th flight, and the 400Hz harmonics behaviour. It is important to note that 400Hz is the frequency of the aircraft AC power. This may indicate that the transitory noise is related to the AC power supply of the aircraft.

3.2 Assessment of Aircrew Noise Exposure

The ICS signal recorded during the flight mission on March 7th consisted of some instances of transitory noise occurring throughout the flight. Many of them were observed during normal flight operations and as such, there was no clear indication of a specific cause for them. It can be observed in Figure 3 that the transitory noise event chosen for this analysis comprised of a harmonic tone with a maximum peak amplitude at approximately 3250Hz and had a measured OSPL shown in Figure 4 of approximately 11.27 dB(A) higher than the normal speech sound level at the stretcher location (both recorded with a maximum ICS gain setting).

4 Conclusion

The data analysis of the recorded CH-149 ICS voltage signal showed that the aircrew are exposed to a variety of intense transitory noise events on a regular basis. Moreover, analysis of the recorded ICS signal showed that the CH-149 intercom system exposed the aircrew to relatively high sound pressure levels during a flight mission. With the maximum ICS gain setting, the CH-149 ICS generated an average OSPL of 96.45 dB(A) at the aircrew ear entrance during regular speech at the stretcher position in the measured flight missions. Several types of transitory noises were recorded and analyzed, including night-mode lighting transitory noise, slow-plug-in of helmet plugs transitory noise and other random transitory

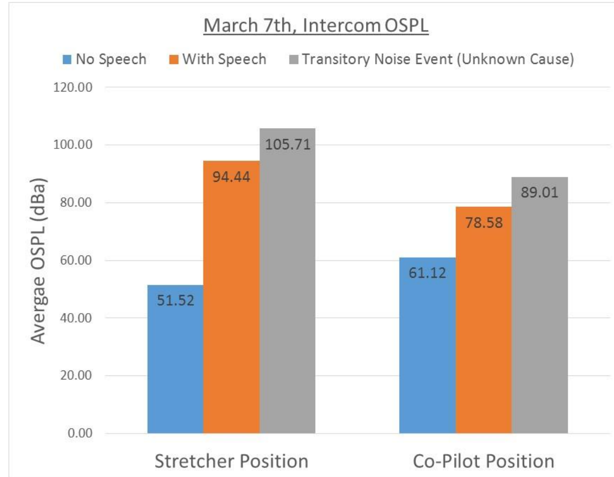

Figure 4: OSPL comparisons from March 7th flight recording.

noise events. The intercom transitory noise introduced high noise levels to the aircrew. For example, the slow plug-in transitory noise was the loudest transitory noise event, exposing the aircrew at the stretcher position to approximately 112.65 dB(A) OSPL for about 2 seconds.

Acknowledgements

The authors would like to acknowledge the support of the Department of National Defense, the IMP Group, the 442 Squadron of the 19th Wing Comox CFB, the NRC Flight Research Laboratory support staff and technical officers of the NRC Aeroacoustics and Structural Dynamics group.

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HEARING PROTECTION PERFORMANCE EVALUATION OF ACTIVE NOISE REDUCTION HEADSETS UNDER HIGH INTENSITY NOISE LEVELS

Victor Krupka*[1], Sebastian Ghinet†[1], Viresh Wickramasinghe‡[1] and Anant Grewal♦[1]
[1]NRC Aerospace Research Centre, Flight Research Laboratory, Ottawa, Ontario, Canada

1 Introduction

The present study focuses on assessing the performance of hearing protectors with Active Noise Reduction (ANR) systems at the limit of their capabilities. The concern addressed in this study was raised following the noise survey onboard the CH-147 Chinook helicopter when an unweighted OSPL of 132 dB was recorded by NRC during the "Open Doors Level 140 kts" flight segment at four different locations in the cabin and the cockpit. Previous performance evaluation by NRC of David Clark headsets with ANR systems at an OSPL (Overall Sound Pressure Level) of 110dB in accordance with Standard Testing Procedure ANSI S12.42 resulted in positive recommendations based on their performance.

In the present study, the four David Clark headset models e.g. 40600G-15, 40600G-20, 40750G-01 and H10-76XL were tested at various sound pressure levels such as 111 dB, 115 dB, 120 dB, 125 dB and 131 dB in order to assess their stability and performance consistency. As a result of the NRC evaluation, it was observed that the performance of the four headset systems with ANR ON was consistent for noise excitation levels below 120dB while a performance degradation was observed for excitation levels above 120dB.

2 Method

The noise attenuation performance of four David Clark headset models e.g. 40600G-15, 40600G-20, 40750G-01 and H10-76XL, equipped with Active Noise Reduction (ANR) electronic modules was evaluated in the NRC Aerospace Hearing Protection Evaluation Facility at the Flight Research Laboratory using the procedure outlined in ANSI Standard S12.42 under various sound pressure levels such as 111 dB, 115 dB, 120 dB, 125 dB and 131 dB.

The Acoustic Test Fixture (ATF): G.R.A.S. 45 CB manikin was used. This device was specifically designed for impulsive and continuous noise measurements [1].

The testing was completed in a reverberant chamber with sufficient control and volume to produce a homogeneous acoustic field with consistent sound pressure levels in the desired frequency range of 0.1 to 10 kHz.

Special care was taken to ensure the optimal fitting of the headsets. Each headset was tested following the standard series of occluded and unoccluded ear conditions. Moreover background noise was measured at the beginning and at the end of each evaluation testing series. The Insertion Loss (IL) was calculated for each unoccluded / occluded ear pairing following the ANSI Standard S12.42 procedure [2].

Throughout the measurements, the acoustic levels and temperature within the test chamber were monitored to ensure consistent and repeatable testing conditions.

The two occluded and unoccluded ear configurations are depicted in Figure 1.

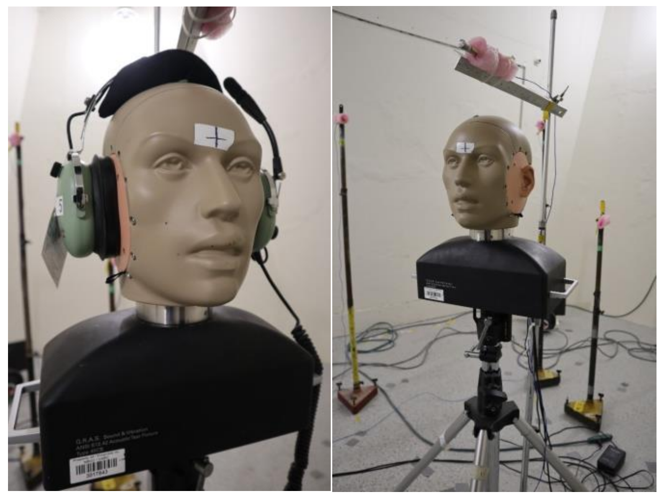

Figure 1: Left: Occluded Ear (with David Clark headset) and Right: Unoccluded Ear configurations.

3 Results and Discussion

3.1 Insertion Loss (IL) Data

During this test campaign, the following four headsets, equipped with ANR, were considered:

1) David Clark headset model 40600G-20
2) David Clark headset model 40600G-15
3) David Clark headset model 40750G-01
4) David Clark headset model H10-76XL

The IL measured for the David Clark headset model 40600G-20 in passive (ANR OFF) configuration is presented in Figure 2. The tests were performed at five different overall sound pressure levels e.g. 111 dB, 115 dB, 120 dB, 125 dB and 131 dB. It can be observed that the passive IL (with ANR OFF) of the headset is consistent at all frequencies below 3000Hz and all sound pressure levels. For frequencies above 3000Hz the headset performance is so great that the GRAS 45CB ATF manikin microphones measured only background noise at low excitation SPLs. It has to be mentioned that only the IL measured for acoustic excitation superior to 125dB are accurate at frequencies above 3000Hz for which higher sensitivity microphones have to be used.

* Victor.Krupka@nrc-cnrc.gc.ca
† Sebastian.Ghinet@nrc-cnrc.gc.ca
‡ Viresh.Wickramasinghe@nrc-cnrc.gc.ca
♦ Anant.Grewal@nrc-cnrc.gc.ca

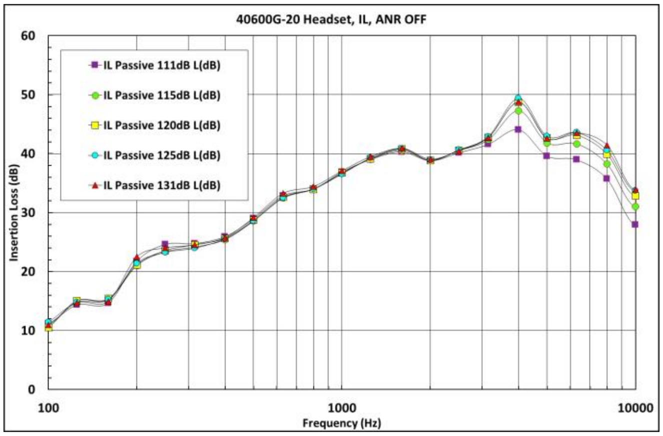

Figure 2: Insertion Loss, Left Ear, David Clark headset model 40600G-20, ANR OFF

The IL measured for the David Clark headset model 40600G-20 in active (ANR ON) configuration is presented in Figure 3. The tests were performed at five different sound pressure levels e.g. 111 dB, 115 dB, 120 dB, 125 dB and 131 dB.

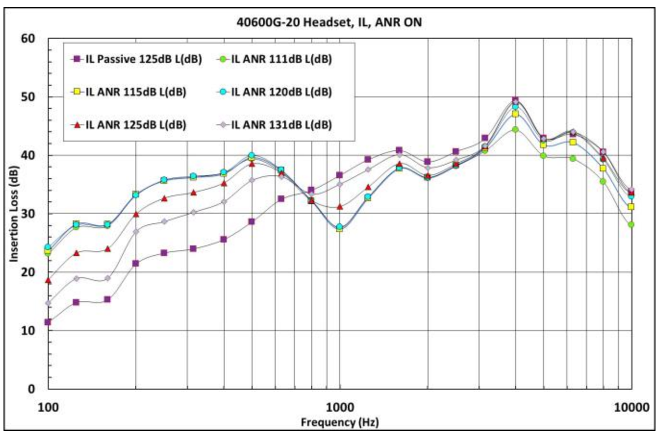

Figure 3: Insertion Loss, Left Ear, David Clark headset model 40600G-20, ANR ON

The IL results obtained for the ANR ON configuration at each sound pressure level of excitation were compared with the results of the ANR OFF configuration at 125dB of excitation – the minimum OSPL required to accurately evaluate the performance of this helmet at high frequencies. It can be observed that at a low OSPL of excitation e.g. 111dB, 115dB and 120dB the performance of the headset is consistent for frequencies below 3000Hz. Moreover the performance of the headset with ANR ON shows a gradual degradation in performance for OSPLs of excitation above 120dB. Moreover, it can be observed that above 800Hz the IL of the headset decreases. The ANR system, instead of cancelling the noise at those frequencies does the opposite and increases the noise level at the eardrum (as compared to the passive – ANR OFF configuration), by as much as approximately 10dB at 1000Hz for low SPLs of excitation. This degradation in performance regresses as the SPL of excitation increases (for OSPL of 125dB and 131dB) and the headset IL performance tends to approach the passive (ANR OFF) configuration results.

Similar behaviour was observed and similar conclusions can be drawn for all the headsets tested.

3.2 ANR Performance for Long Time Exposure

It was very important to test the headsets at high sound pressure levels and observe if their performance remains stable over long periods of exposure. The first testing case was to expose the David Clark 40752G-01 headset with ANR ON at OSPL of 131dB for 15 min. Furthermore, David Clark 10600G-20 headset with ANR ON was tested at OSPL of 125dB for 45 min. For both cases, it was observed that no degradation of the headset performance occurred.

4 Conclusion

In the present study, the four David Clark headset models were tested at various sound pressure levels As a result of the NRC evaluation, it was observed that the performance of the four headset systems with ANR ON was consistent for noise excitation levels below 120dB.

However, as is demonstrated by the present preliminary evaluation, the insertion loss performance of the four headsets with ANR systems ON, when exposed to unweighted overall sound pressure levels (OSPL) superior to 120dB, significantly degraded with each increasing noise level increment. It is also very important to mention that the passive hearing protection performance of the four headsets (ANR OFF) remained, consistent (no degradation observed) at all high intensity noise levels considered in this study.
Additionally, it was observed that longtime high OSPL noise exposure did not degrade the headset performance.

Acknowledgements

The authors would like to acknowledge the technical support from Mr. Christophe Legare and Mr. Brent Lawrie.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Design Of A Test Apparatus For Measuring The Low-Amplitude Vibration Sensitivity Of Large Test Articles

Todd Busch

The author provides design details for a test apparatus that was designed to accommodate test articles with dimensions of several metres and masses of several thousand kilograms. The apparatus included a spring supported inertia base as a rectangular platform for mounting the test article and a 1500 lbf electrodynamic shaker that could drive the platform in both horizontal directions as well as vertically. A case study is provided that discusses the utilization of the apparatus for determining the low amplitude vibration sensitivity of a process tool used in the microelectronics manufacturing industry, a so-called probe tester. The probe tester is used to determine the electrical integrity of chips on a semiconductor wafer by applying two electrodes to each uncut chip within a 30-micron diameter target. The vendors of this probe tester had developed their own vibration sensitivity curves, but their veracity was subject to challenges from a purchaser and user of large quantities of these test articles. As such, the purchaser wished to determine of the low-amplitude vibration sensitivities for themselves and achieving potential savings on vibration isolation for a manufacturing facility in proximity to a railway track. As such, the test apparatus was developed that could accommodate the dimensions and mass of a probe tester while inducing vibrations with displacements on the order of microns at both external and internal points on the probe tester. With the successful utilization of the test apparatus, low-amplitude vibration sensitivities were determined and the vendors warranty provisions regarding acceptable vibration were voided. As a result, the purchaser was able to avoid many millions of dollars of expense on vibration isolation within their renovated manufacturing facility.Keywords: MICROELECTRONICS, VIBRATION, SENSITIVITY, MICRON, SEMICONDUCTOR

Évaluation Des Nuisances Sonores Provenant Des Systèmes À Traitement D'air En Milieu Urbain

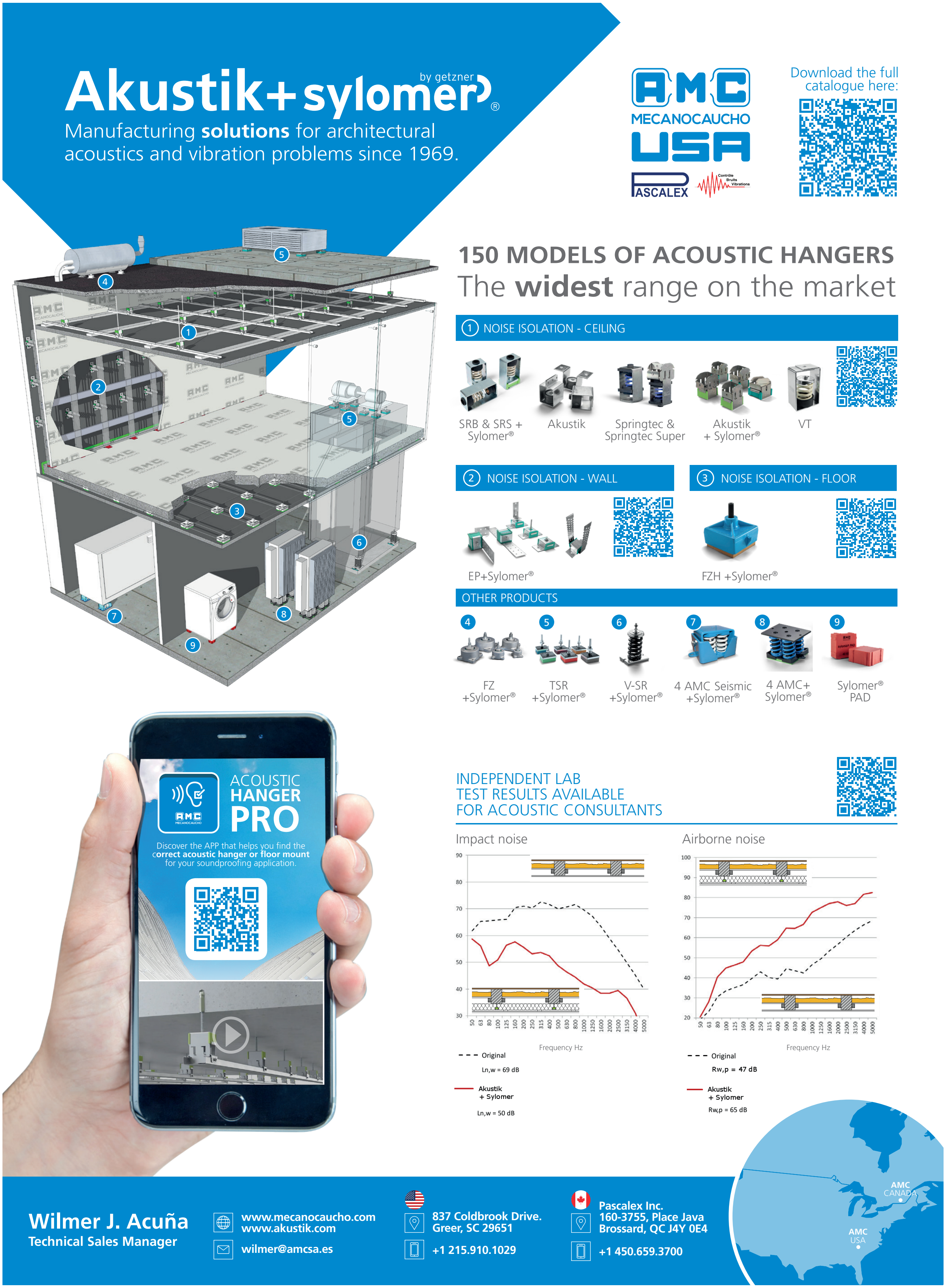
Maryse Lavoie

Dans le cadre réglementaire actuel de l'arrondissement du Plateau-Mont-Royal, les plaintes de bruit concernant les systèmes à traitement d'air (climatisation, chauffage, ventilation, réfrigération) sont souvent difficiles à traiter. En effet, elles requièrent la prise d'une mesure acoustique pour démontrer le dépassement des normes de bruit. Avec la multiplication des sources sonores (traitement d'air, travaux de construction, festivals, circulation, etc.), il est souvent difficile d'évaluer la contribution sonore d'un équipement en particulier, à moins de pouvoir le contrôler directement (mise en fonction, hors fonction, paramètre de vitesse, etc.) ou de pouvoir s'en rapprocher suffisamment. Toutefois, malgré la difficulté technique à isoler une source sonore, l'oreille humaine discerne amplement plusieurs nuisances liées aux bruits des équipements mécaniques. L'analyse des requêtes de bruit enregistrées à l'arrondissement sur une période d'un an vise à recenser les emplacements problématiques, ainsi que les différentes nuisances afin de prévenir les nuisances sonores liées aux systèmes à traitement d'air et d'émettre des recommandations notamment en urbanisme.

Sound Insulation And Absorption In Rail Applications

Mark Salsberg

Transit and Rail systems can be a noisy environment. In a suprisingly increasing number of instances it can cause havoc in urban and rural areas either by blaring horns or general operations between the wheel and rail interface.As density around rail increases, more applications and solutions are needed to address this growing and ongoing concern.Using recycled materials for automotive textile waste and recycled tires, Brens Europe has been able to successfully decrease noise and vibration by up to 15 dB using its GreenTrack system in an urban area.Depending on the application, and location, rural or urban, Brens Europe was able demonstrate significant improvements in noise and vibration reduction in the area without using large and imposing noise walls.Brens Europe has been successful installing discrete solutions for rail environments such as low barrier noise walls, vegetation walls, Green Track and recycled rubber noise blocks on the rail.Brens North America, the North American affiliate to Brens Europe, will showcase projects around Europe that have leveraged recycled waste, and its wide solution range. These solutions can be leveraged across Canada with its vast network of rail through communities both big and small. These solutions will not only help communities, but also improve the environment in terms of ecology and noise pollution.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DEVELOPING THE PORT OF VANCOUVER'S PORT NOISE RATING METHODOLOGY

G. Mak [*1] and M. Bliss [†1]
[1]BKL Consultants Ltd., North Vancouver

1 Introduction

The Vancouver Fraser Port Authority manages Canada's largest port, the Port of Vancouver, encompassing federal port waters, lands, and shorelines in and around Vancouver, BC. In total, port-managed areas border 16 municipalities. As urban densification has increased near port operations, so too has the potential for port-related noise to disturb nearby communities.

Between 2013 and 2015, the port installed 11 permanent noise monitoring terminals (NMTs) along the north and south shores of the Burrard Inlet and at Roberts Bank to better understand port-related noise in nearby communities and address community noise concerns. Figure 1 below shows the location of each NMT. The NMTs continually log sound data in or near communities potentially affected by port noise.

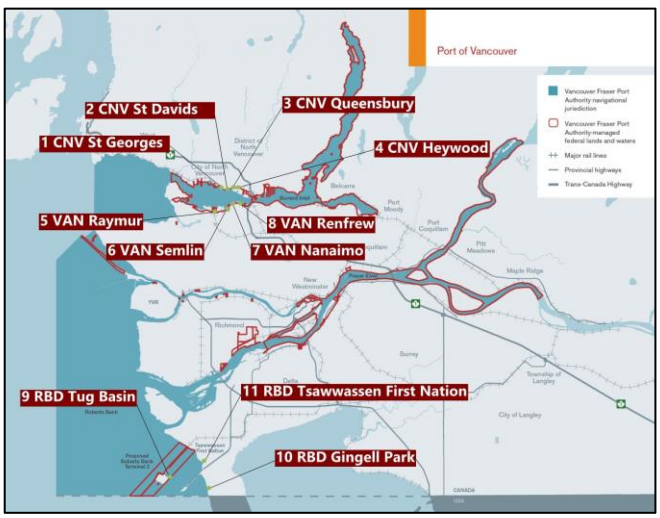

Figure 1: Port of Vancouver NMT Locations

In order to provide value from this data to the port and its stakeholders, BKL developed a Port Noise Rating (PNR) metric which relates the measured sound pressure levels to the potential annoyance in the surrounding population using noise modelling and census data. The PNR metric provides a useful way for the port to interpret the significance of measured noise levels, changes over time, and differences between NMTs.

2 Rationale

Although the NMT system continuously logs sound levels, the recorded data alone do not give much insight on how port noise is impacting the nearby residential communities. There are three shortfalls with only reviewing NMT sound levels: they do not necessarily represent community noise levels, they do not take into account the number of people in the vicinity, and they do not directly correlate with annoyance.

Due to practical considerations, the Port's NMTs are installed at varying distances from nearby residential communities. Some are installed in residential neighbourhoods while others are installed on or very close to Port properties including roads, railways and terminals. Hence, measured noise levels do not represent the overall noise exposure of the entire community represented by each NMT. Furthermore, the population density near each NMT varies, with some areas with single family housing and some with mid-rise multi-family residential.

The sound data recorded by the NMTs alone does not represent the annoyance caused by port-generated noise. The sound pressure level is not the only factor in how annoying noise is. The time of day and type of noise, or the quality of it, can also greatly affect how annoying the sound is perceived. Night noise is more disturbing than day noise. Furthermore, sound with tonal, impulsive, or excessive low-frequency content can all increase the level of annoyance. Noise levels adjusted for time of day and intrusive characteristics are called rating levels.

In light of these shortfalls, the PNR was developed to relate the measured sound pressure levels to potential annoyance in the surrounding residential communities through a methodology involving noise modelling, percent highly annoyed calculations, and census data.

3 Methodology

The PNR is the estimated number of people highly annoyed (*#HA*) using the day-evening-night equivalent sound level (L_{den}) rating metric, per ISO 1996-1:2016 [1], combined with census data, for each community area represented by each NMT. The ISO standard references outdoor noise levels at receiver locations, but the L_{den} at each residence in the surrounding neighbourhoods varies with distance from the dominant port noise sources and is not equivalent to the L_{den} logged at the NMTs. Therefore, those values need to be estimated before the PNR can be computed.

3.1 North and South Shore NMTs

At the north and south shores of Burrard Inlet, community noise levels from road traffic, rail traffic, and industrial activity within port lands are estimated for residences in the area surrounding each NMT using NMPB-Routes-96 [2], SRM II [3], and ISO 9613-2:1996 [4] standards implemented in Cadna/A noise modelling software. The models are used to estimate the sound level difference between each NMT location and nearby residences based on modelling

[*]mak@bkl.ca
[†] bliss@bkl.ca

assumptions developed for each of the major noise sources. In general, the models included residences with estimated L_{den} levels of approximately 50 dBA or greater. The residences represented by each NMT are divided into subgroups with similar noise exposures.

For each subgroup, a review of residential buildings and census average household size in the area is conducted to estimate the population. Only half of the residences are considered for multi-family buildings since only approximately half of the building facades would have direct exposure to port-related noise sources. The *#HA* is then calculated by multiplying the population and the corresponding *%HA* for each subgroup.

The sum of all the subgroups surrounding each NMT represents the total *#HA* or *PNR* for each NMT. The *PNR* for the entire north or south shore can also be calculated by summing the *PNR* from each NMT.

3.2 Roberts Bank NMTs

The same approach used for the north and south shores cannot be used for Roberts Bank due to the large distance between the port and the nearest residences, and because most port-related noise is quieter than local ambient noise levels in the nearest communities. One NMT is sited in Roberts Bank and two are sited in the community. Since noise complaints in this area have historically been related to low frequency noise, the *PNR* has been calculated considering the low frequency rating level per ANSI S12.9-2005 Part 4 [5].

A Cadna/A noise model was developed to predict monthly community noise levels. Each month, the noise sources in the model are calibrated to the noise levels measured at the RBD Tug Basin NMT and the C_{met} is calculated at 30-degree direction increments using the reported weather conditions and LfU Bayern method implemented in Cadna/A. After calibration, the L_{den} and L_{LF} are predicted at the residences and the *#HA* is calculated at each residence using census data. In general, the model includes the first two rows of residences facing Roberts Bank.

4 Discussion

The *PNRs* for each NMT has been calculated from monthly NMT data for more than 5 years now. Through comparing *PNRs* between sites and reviewing long-term trends, a number of observations were made during this time.

The *PNR* predicts where port-related noise is affecting the most people. The NMTs with the highest measured noise levels did not always yield the highest *PNRs* due to differences in population density and proximity to port noise sources. For example, a 5 dBA increase at the CNV St Georges NMT increased the *PNR* by about 15 whereas a similar increase at the VAN Semlin NMT increased by about 50. Thus, the PNR is able to show that similar noise increases could potentially cause much more impact at one site compared to another. Furthermore, at the CNV St Davids and Queensbury NMTs, the PNR has increased dramatically over the past few years due to large scale rezoning in the area which converted most single-family homes to multi-family dwellings.

At Roberts Bank, changes in *PNR* depended more on wind speed and direction and low frequency noise over measured L_{den} levels. The *PNR* could change even when L_{den} levels did not change noticeably. Some of the shipboard generators have much more low frequency noise than others, and due to the large distance between port noise sources and the residences, the sound propagation is significantly influenced by upwind versus downwind conditions.

5 Limitations

The NMTs are located near many noise sources that are related and unrelated to the port. As the *PNR* metric hinges on the modelled noise level differences between the NMT locations and residences, the accuracy of this approach depends on many noise source modelling assumptions necessitated by the complex noise environments that exist throughout the port's jurisdiction.

Currently, not all port noise sources are modelled in detail which affects the accuracy of *PNR* values. The accuracy of this approach could be improved by further studying the noise sources in the area, whether port-related or not. Additional NMTs could also be added so that modelled predictions are relied on less heavily to estimate noise levels at residences farther away from the current NMTs.

To predict annoyance more accurately, more adjustments can be incorporated to account for sound source character. While the evening and nighttime penalties are currently incorporated in the L_{den} rating, weekend, tonal and impulsive penalties are not included at this time. Further analysis would be required in order to incorporate these specific penalties.

Acknowledgments

Special thanks are given to Mr. Gary Olszewski at the Vancouver Fraser Port Authority for his kind support and permission to discuss the Port's noise monitoring program.

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INNOVATIVE AND FEASIBLE NOISE MITIGATION PLANNING

Amir A. Iravani * and Lucas Arnold †
Dillon Consulting, Toronto, Ontario, Canada

1 Introduction

For For industrial sites located near residential areas, noise pollution can be a limiting factor in expanding production. When KFP Inc. wanted to upgrade machinery and increase their annual throughput, they faced challenges dealing with increasing noise levels above acceptable limits at the neighbouring residential community. They needed a solution that would achieve continuous noise shielding, however, a permanent noise barrier wall would cost millions of dollars and extend the project timeline. We envisioned a way to arrange for stacks of logs to be used as noise barriers and developed a plan to maintain continuous shielding even as the log inventory is removed for processing. The log wall changeover plan was developed through implementation of parallel walls system. By prototyping this barrier and completing the changeover process with sensors in place, we confirmed that our noise mitigation strategy was successful. This solution demonstrates the value and viability of using available onsite materials to provide noise mitigation solutions and suggests the approach can be applied more widely. With good design, thorough investigation, and creative planning, regulatory compliance and improved noise environment for sensitive receptors in proximity of industrial facilities can be achieved.

2 Methods

2.1 Facility overview

The facility consists of a sawmill building, planer building, two existing kilns, one proposed kiln, a baghouse and an outdoor log conveyor. The facility is located on land that was zoned Heavy Industrial and is surrounded by water on the north, west and south sides. The lands immediately to the east of the site are zoned Heavy Industrial. However, across the water to the north, west and south are residentially zoned areas, with a large existing community less than 500 m to the west, as shown in Figure1.

The background noise, exclusive of that generated by the site, was characterized as a Class 2 (urban) area, as described in NPC-300 [1]. The primary contributors to the background sound during the daytime and night-time periods include road traffic on nearby roads/Highways, including Trans-Canada Highway and Darlington Drive, train noise associated with CP Rail operations along its main line and several road and rail bridge crossings

* airavani@dillon.ca
† larnold@dillon.ca

Figure 1: Log processing facility in the centre, with receptor locations shown in red indicating residential communities impacted by noise

2.2 Noise measurements and mitigation

The facility operates seven days per week with operations occurring during both daytime and nighttime hours. The dominant noise sources at the facility were identified and individually measured as part of the study. Noise source measurement was conducted in accordance with MECP publication NPC-103 Noise Measurement Procedures [2].

Noise control measures were implemented at the facility, including enclosures over equipment, rubber strips at plant openings, rubber pads at log impact contact points, broad-band mobile equipment back-up alarms, as well as procedures for keeping doors closed and time-of-day restrictions for specific equipment. However, even with these mitigation strategies, field measurements showed that the nearest community was subject to noise levels above the acceptable limits.

2.3 Creative planning and modelling

The standard practice would be to construct a long vertical noise barrier, in this case at the cost of several million dollars, between the facility and the community. However, during site reconnaissance, we observed that large stockpiles of wood were placed all over the yard, waiting to be processed. By collaborating with the facility, we devised a plan to mitigate noise by using the existing on-site stockpiles of logs. We envisioned a two-part solution: 1) 4-5 m high earth berm to be constructed at the northwest end of the stockyard where the available land allows for the wide footprint required for an earth berm; and, 2) 7 m high, 125 m long log pile to be constructed along the western edge of the property to protect the residents closest to the facility. This scheme is displayed in Figure 2. We also developed a changeover plan to maintain continuous shielding even as the log inventory is rotated through for processing. This was achieved by having several parallel log piles with an access corridor in between for the log moving equipment to manage the log piles.

Effective dimensions and locations of the log piles were determined through noise propagation modelling. The model incorporated site-specific features such as elevation, berms, ground absorption, and barriers to predict noise levels at specific receptors.

Figure 2: Proposed noise barrier consisting of a log wall (red) and an earth berm (green).

3 Results

3.1 Modelling mitigation measures

The ISO 9613 based model accounts for a reduction in sound level due to increased distance and geometrical spreading, air absorption, ground attenuation, and acoustical shielding by intervening structures and topography. The model is considered conservative since it represents atmospheric conditions that promote the propagation of sound from source to receiver, Graphic output from the model, illustrating post-mitigation sound level contours and predicted receptor noise levels for noise sources are presented in Figure 3.

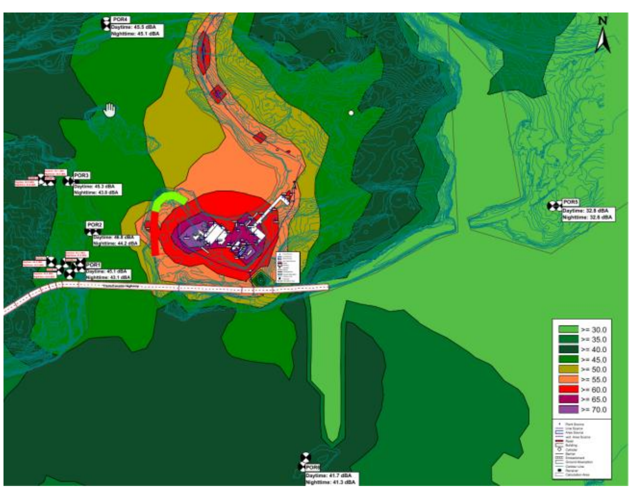

Figure 3: Model indicating non-impulsive noise levels after implementation of log barrier (red) and earth berm (green).

3.2 Permitting and approvals

The earth berm and log wall were constructed and receptor noise verification testing was conducted to verify modelling predictions. This study confirmed that even with the addition of the new machinery and production capacity, the facility was in compliance with the applicable daytime and night-time noise exclusionary limits defined in the MECP's NPC-300, for all sources assessed.

The key to our success in gaining regulatory approval was taking a data-driven approach to design, modelling and implementation, then validating our results in the field through measurement. We also designed a clear, easily executed set of protocols for the facility to follow in order to mitigate noise impact through an effective and cost-neutral solution. Proactive consultation with MECP, supported by thorough investigation and documentation, allowed us to acquire the necessary permits and approvals for the facility to proceed with their expansion.

4 Discussion

One of the most important aspects of the log wall design is that it must be replaced quarterly to rotate stock, ensuring that inventory is not wasted through rot and that the yard remains fully functional. This replacement process is done by constructing several parallel log piles (walls) parallel to the first one so at no time is there a direct line of sight between onsite noise sources and the residential community to the west.

5 Conclusion

This solution demonstrates the value and viability of using available onsite materials to provide effective noise mitigation solutions and suggests the approach can be applied more widely. Through divergent thinking, thorough investigation, and creative planning, innovative solutions to noise pollution are attainable. Assessment and utilization of unique opportunities and/or limitations that may exist for each site/facility is key in developing innovative solutions for noise mitigation.

Acknowledgments

The author would like to thank KFP Inc. for working diligently with us to improve the noise environment in the community and Thomson Hawks at Dillon Consulting for support in the preparation of this paper.

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Études Des Procédés D'Évaluation, De Représentation et de Législation Du Bruit Environnemental Applicables Au Québec

Jean-Philippe Migneron *1, Jean-François Hardy †1, André Potvin ‡1, Jean-Gabriel Migneron ♦1 et Frédéric Hubert ♯2

[1] Groupe de Recherche en Ambiances Physiques, École d'Architecture, Université Laval, Québec

[2] Centre de recherche en intelligence et données géospatiales, Université Laval, Québec,

1 Introduction

Une étude exhaustive et multifactorielle publiée en 2015 par l'Institut national de santé publique du Québec (INSPQ) dressait un portrait du bruit environnemental et de son influence sur la santé populationnelle [1]. Le rapport a permis d'inscrire le sujet dans le plan d'action interministériel 2017-2021 de la Politique gouvernementale de prévention en santé visant la protection de la qualité de vie des citoyens.

En vue de bonifier la réflexion gouvernementale sur les différentes manières de contrôler le climat sonore au Québec, les autorités provinciales ont mandaté des équipes de recherche provenant du milieu académique afin de produire des études complémentaires sur divers aspects. Le présent article résume l'implication de trois groupes de recherche de l'Université Laval pour l'étude de la cartographie sonore et pour la préparation d'un guide pratique axé sur l'insonorisation des bâtiments.

2 Cartographie sonore

2.1 Description du projet

Le Département de géographie, l'École supérieure d'aménagement du territoire et de développement régional, ainsi que l'École d'architecture travaillent conjointement sur le projet dirigé par le Centre de recherche en intelligence et données géospatiales (CRDIG), lequel vise à réaliser un inventaire des techniques de cartographie sonore utilisées dans le monde. L'objectif étant d'abord de dégager comment les données disponibles sont transposées en cartes de bruit à l'aide des outils modernes, en s'attardant sur les divers contextes d'application et sur la fiabilité des méthodes. Cette connaissance permettra de soumettre les solutions de cartographie sonore les plus adéquates et réalistes dans le contexte québécois, encourageant du même fait une plus grande cohérence dans les méthodes utilisées.

2.2 Recherche systématique

La recension systématique sur la cartographie cerne les articles publiés dans les quinze dernières années, afin d'identifier l'ensemble des méthodologies applicables à partir des mesures acoustiques, des modèles prédictifs ou d'une combinaison de ces approches. Cette recherche préliminaire a permis d'extraire un important compendium de 11835 références.

Inspirée du protocole PRISMA [2], elle se sert de quelques-unes des plus importantes bases de données scientifiques et d'un certain nombre de mots clés anglais séparés dans les trois thématiques résumées au tableau 1.

Tableau 1: Mots clés de la recherche systématique.

Géomatique	Acoustique	Environnemental	Mixte
gIS	noise*	urban*	noise map*
geograph*	acoustic*	environment*	
carto*	sound*	pollution*	
map*			
measurement*			
assessment*			

Le premier tri réalisé exclusivement sur la lecture des résumés par une paire d'évaluateurs a exclu une grande quantité d'articles sur la base de paramètres primaires. Une seconde sélection, en cours d'achèvement, devrait permettre de ne conserver que les articles les plus pertinents dont les informations seront extraites et analysées.

Tableau 2: Paramètres d'inclusion utilisés pour le second tri.

Caractéristiques	Description des paramètres d'inclusion
sujet	concernent la cartographie du bruit urbain
langue	rédigés en anglais ou en français
date	publiés entre 2004 et 2019
type d'articles	publiés dans des revues scientifiques avec comité de lecture ou actes de conférence révisés
échelle géo.	liés au milieu urbain à l'échelle de la ville impliquant des populations
sources sonores	étudiant la propagation des sources si un milieu urbain est affecté par le bruit
méthode	décrivant suffisamment les méthodes ou approches de cartographie du bruit

2.3 Prochaines étapes

La classification de ces données servira à comparer toutes les méthodes de cartographie sonore et à identifier les plus favorables à l'analyse du climat sonore dans le contexte québécois. Pour ce faire, les méthodes seront inventoriées en fonction des données acoustiques disponibles, du traitement des impacts du bruit et de l'évaluation des nuisances sur la population affectée.

Une étude de cas est également prévue afin de valider la représentativité d'une ou plusieurs approches en tenant compte des infrastructures géomatiques, de même qu'une

* jean-philippe.migneron.1@ulaval.ca
† jean-francois.hardy.1@ulaval.ca
‡ andre.potvin@arc.ulaval.ca
♦ jean-gabriel.migneron@arc.ulaval.ca
♯ frederic.hubert@scg.ulaval.ca

série de données acoustiques recueillies sur le terrain. Guide de bonnes pratiques en matière d'insonorisation contre le bruit extérieur

2.4 Description du projet

Le Groupe de recherche en ambiances physiques (GRAP) est responsable d'un autre projet de recherche consistant à la rédaction d'un guide de bonnes pratiques en matière d'insonorisation contre le bruit extérieur s'adressant aux ministères, aux services municipaux et aux développeurs immobiliers. L'objectif est de leur offrir une ressource bien illustrée, basée sur un contenu trié à travers la littérature et la pratique existante pour qu'ils arrivent à optimiser la conception des nouvelles constructions résidentielles ou encore à amenuiser certains problèmes d'exposition sonore dans le cadre de rénovations.

S'appuyant sur les travaux antérieurs initiés par le Conseil National de Recherche du Canada [3] et en consolidant un corpus de ressources documentaires rassemblées par une revue narrative, le guide doit répertorier les exemples et les interventions applicables dans la réalité climatique, économique ou constructible de la province.

2.5 Contenu du guide

Le guide sera subdivisé en quatre sections. La première, plus générale, expliquera les normes de construction et de mesures acoustiques. Ensuite, les sources de nuisances et la disposition du bâtiment feront l'objet d'une seconde section axée sur l'environnement physique et sonore des habitations. La partie suivante s'avérera la plus importante du guide. Elle renfermera de multiples exemples concrets de design des diverses parties de l'enveloppe des bâtiments avec des traitements acoustiques adaptés selon les besoins préétablis. Finalement, un résumé de l'isolation phonique recommandée et des caractéristiques vibroacoustiques des matériaux présents dans la construction des façades ou des toitures procureront les ressources nécessaires aux lecteurs pour comparer et ensuite intégrer les solutions envisageables.

Une section importante comprenant des exemples d'assemblage ou des compositions typiques inclura aussi des détails de construction au niveau des éléments sensibles des habitations, tels que les fenêtres, les balcons, les murs, les plafonds et les ouvertures. Le tableau 3 fait d'ailleurs état des aspects incontournables devant être analysés dans le futur guide.

Tableau 3: Éléments à considérer pour l'insonorisation extérieure.

Éléments
surface et isolation du vitrage
types de verres et leur disposition
modes d'ouverture des fenêtres et influence sur le bruit entrant
vitrages fixes avec possibilité de ventilation naturelle ou mécanique
différents aspects du doublage des fenêtres
disposition et traitement acoustique des balcons
oriels et diverses formes du doublage partiel d'une façade
double enveloppe pour les bâtiments résidentiels
combinaison avec l'isolation thermique
portes exposées aux bruits
importance de la masse pour l'isolation contre les basses fréquences
murs lourds avec maçonnerie ou recouvrements massifs
murs légers avec finition extérieure de bois, de métal ou enduits
toitures avec charpentes
toitures plates ou à faibles pentes

3 Conclusion

La recherche systématique au sujet de la cartographie sonore permet déjà de constater la grande diversité méthodologique utilisée dans le monde. Les conclusions qui en seront tirées pourront servir de référence afin de choisir les représentations du climat sonore qui sont les plus adaptées aux besoins des différentes instances publiques. Il est supposé qu'une uniformisation des procédures pourrait faciliter le travail des autorités compétentes, de même que celui des praticiens en acoustique environnementale. De plus, ce récent effort de recherche pourra bénéficier à plus long terme aux recherche pourra bénéficier à plus long terme aux développements techniques pour que la pollution acoustique soit traitée dans les nouvelles villes intelligentes.

Quant au guide de bonnes pratiques en matière d'isolation contre le bruit extérieur, il constituera une ressource importante et contemporaine pour la construction et la rénovation au Québec, comme en font foi les divers sujets qu'il traitera. Le ministère de la Santé et des Services sociaux devrait contribuer à rendre le futur document accessible à tous en finalisant le graphisme, tandis que l'équipe de l'École d'architecture de l'Université Laval participera à sa diffusion dans la province, voire à l'échelle canadienne.

Les différents projets en cours cherchent à synthétiser les multiples aspects entourant le contrôle du bruit communautaire au Québec. Pour réussir à rassembler les différents points de vue et ainsi répondre aux attentes de la majorité des parties prenantes, les intervenants intéressés sont cordialement invités à signifier leurs préoccupations ou à partager leurs expériences avec les chercheurs impliqués.

Remerciements

Ces projets de recherches sont rendus possibles grâce au financement du ministère de la Santé et des Services sociaux et du ministère de l'Environnement et de la Lutte contre les changements climatiques du Québec.

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DIFFERENCES IN SOUND EXPOSURE RESULTS FROM FIREARM DISCHARGE DUE TO MEASUREMENT EQUIPMENT SELECTION

Peter VanDelden *[1], Philip Tsui †[2] and Alec Medemblik.
[1]RWDI, 600 Southgate Drive, Guelph, Ontario N1G 4P6
[2]RWDI, 301-2250 Oak Bay Avenue, Victoria, British Columbia V8R 1G5

1 Introduction

The high sound levels associated with firearms discharge are of interest to recreational users and police forces. When engaged in training, target practice or competition it is normal to use personal hearing protection to reduce sound exposure from the large numbers of impulses. Hearing protection is also used by high volume shotgun hunters (e.g., for waterfowl) and by those hunting moose or deer with a high-power rifle. However, law enforcement personnel discharging a firearm in the line of duty generally do not have the benefit of hearing protection. Whether there are high sound levels or many impulses, there is interest in knowing the significance of sound exposures for the firearms users.

The sound from a firearm is influenced by several factors. These include the projectile and propellant characteristics, projectile speed, firearm barrel length and gas discharge influences such as a muzzle brake and noise suppression systems. The combination of these factors results in sound that is not evenly distributed around a firearm.

The position of the hearer is also significant to the sound level experienced. People not operating the firearm, such as instructors, safety officers, competitors and observers each have a different and lesser sound exposure based on where they are positioned. The highest sound level generally occurs at the user position.

The relationship of the user's right and left ears to the firearm varies. When a handgun is held in the Isosceles Stance (i.e., with two hands), it is positioned equally in relation to the right and left ears. However, in a Weaver Stance or in single-handed use one ear is closer than the other. Similarly, with a longer firearm, sighting down the barrel brings one ear closer than the other. This paper considers the ear furthest from the gun barrel.

Accurately measuring sound level at the ear position requires consideration of many factors. The US Department of Defense standard MIL-STD-1474E [1] includes guidance for conducting such measurements. Of interest to this paper is the specification of a 192 kHz sampling rate. A 192 kHz sampling rating is intended to capture the very-fast-rising, short duration pulse. The capability of measuring sound at this sampling frequency is not available in a format readily accessible to lay users. Standard sound level meters sample at approximately 50 kHz. Systems at 192 kHz are significantly more expensive, often require knowledge of signal processing and data analysis, lack portability, or operate on batteries for comparatively only periods of time.

Where users are aware of the MIL-STD-1474E standard, equipment selection is still often determined by challenges surrounding 192 kHz measurement systems and convenience of a sound level meter. Without a comparison of measurement results between a sound level meter and a 192 kHz system, the exposure significance is unknown. This paper seeks to inform the equipment selection with a summary of the differences in measured sound levels in proximity to a user's ear over a diversity of firearms.

2 Method

Measurements were conducted using seven different firearms. The following were selected to represent a diversity of what is in use:

Kimber 1911: 45ACP, 5" barrel
FNH FNS40L: 40S&W, 5" barrel
Smith and Wesson M&P Pro: 9x19 mm, 5" barrel
Browning BPS: 12 ga, 26" barrel
Browning X-Bolt: 300 Winchester Magnum, 26" barrel
AR-15 (M4/C8): 5.56x45 mm, 16" barrel
AR15 (M16/C7A2): 5.56x45 mm, 20" barrel

These offered a range of projectile sizes and speeds. Standard issue or common bullets were used for most of the firearms. However, variation due to charge weight and propellant burn rate were also nominally considered.

The measurement program was conducted at an outdoor range. A range was selected that was free of reverberance or reflective surfaces, other than the benches designated for firing. The non-reverberant space was selected to provide the most demanding measurement conditions, where the high sound level is present for the smallest amount of time. A reverberant indoor firing range is understood to be less challenging to measure at slower sampling rates because an elevated sound level is present for a longer duration.

The pistols were fired from a seated position in Isosceles Stance. A standing position was used for the Browning BPS, with the remaining firearms being supported on the bench and fired from a seated position.

Sound levels were measured simultaneously by three measurement systems. The basis for comparison was an LMS system with 204.8 kHz sampling rate. A more portable system, at 102.4 kHz was one alternative. The third system was a sound level meter targeted specifically to firearms noise applications: the Larson Davis LxT1-QPR. The LxT1-QPR has a 51.2 kHz sampling rate. Prior to the measurements each system was field-calibrated.

* Peter.VanDelden@rwdi.com
† Philip.Tsui@rwdi.com

Each of the measurement systems was equipped with a blunt cylinder microphone as indicated in the standard. The ¼” microphone size was selected to prevent the overrange that occurs with larger diameter microphones at high sound levels. The microphones were arranged at 5/8” from centre to centre but separated from each other by vibration isolation material. They were positioned in line with a virtual axis through the ears and at approximately 6” to the left side of the head. In all cases where the firearms operator was seated, the microphones were supported on a tripod. They were mounted to the shoulder in the case of the Browning BPS: 12 ga.

The sound from each firearm was measured multiple times, with a total of 92 files being recorded on each measurement system.

3 Results

For each of the 92 files on a system the Z-weighted peak level was recorded and average Z, A and C-weighted sound levels were calculated. The noise floor of the systems and background sound levels in the firing range were determined to be sufficiently low as to not influence the results. The sound level differences between measurement systems were calculated for each file. The 51.2 kHz and 102.4 kHz systems were each compared with the 204.8 kHz system. Results are presented in Table 1 below.

Table 1: Recorded Sound Level Difference When Compared with System Having 204.8 kHz Sample Rate

	102.4 kHz System	**51.2 kHz System**
Peak (dB)	≤ 2.1	≤ 3.4
L_{EQ} (dB)	≤ 1.0	≤ 0.6
L_{EQ} (dBA)	≤ 2.7	≤ 0.5
L_{EQ} (dBC)	≤ 0.7	≤ 0.5

The measurement sets had average projectile speeds between 800 feet per second and 3100 feet per second.

4 Discussion

A clear trend in the peak sound levels has been identified from the measurement data. The faster the sampling rate, the higher the peak level measured. However, the difference in measured peak level between the 51.2 kHz and 102.4 kHz sampling systems is not constant from one firearm measurement set to another. Overall, the 102.4 kHz system provided results up to 2.1 dB quieter than the reference system. The system sampling at 51.2 kHz was up to 3.4 dB quieter than the one sampling at 204.8 kHz.

This trend was expected to continue with the calculated average sound levels. This was generally true for the C-weighted levels. Data sets on the 102.4 kHz system were normally within 0.4 dB, and not more than 0.7 dB of the reference system. The 51.2 kHz sampling rate system was generally close behind, with a difference of 0.5 decibels separating C-weighted levels on the 204.8 kHz and 51.2 kHz systems.

The A and Z-weighted results from the alternate measurement systems are also quieter than the reference system. However, they had an unexpected trend. The calculations show that the 51.2 kHz system produced results closer to the reference system than the 102.4 kHz system in all cases. The difference between 51.2 kHz and 102.4 kHz was generally about 0.2 dB for Z-weighted data sets and 1.7 dB for A-weighted data sets. Maximum A and Z-weighted differences shown in Table 1 were reasonable in the context of the data sets. The trend prompted additional analysis, and consultation with technical staff at the manufacturer of the 102.4 kHz system. The unexpected results suggest that there is more involved than simply sampling rate.

The projectile speed data offered the opportunity to look for trends. The relationships between projectile speed and a system’s ability to capture peak, Z, A and C-weighted average sound levels was considered. No clear correlation was found overall or when the sub-sonic and super-sonic groups were considered separately. Charge weight and propellant burn rate had small influence besides speed.

In general, the differences between measurement systems cannot be ignored. A difference of 3 dB in peak level can make the difference between meeting or exceeding a 140 dB peak-level threshold. For L_{EQ} averaged sound levels, a difference of 3 dB is equivalent to a halving or doubling the allowed time (with a 3-dB exchange rate). If a user is unaware that a system shows A-weighted L_{EQ} results that are 3 dB quieter than actual, the sound exposure may be allowed to continue for twice as long as it should.

Understanding how a measurement system compares with the standard can allow users to compensate. The results of measurements presented here indicate that when measuring a user’s hearing exposure using the Larson Davis LxT1-QPR (i.e., the 51.2 kHz sampling system here), the meter’s peak level (Z-weighted) output would be increased by 3.4 dB. An increase of 0.5 above the meter’s output would apply to A-weighted decibel levels. Measurement using another system, even with a higher sampling rate, does not necessarily produce comparable results. To determine suitable adjustment factors for other sound level meters or sampling systems a similar side-by-side testing program with the diversity of firearms would be needed.

5 Conclusion

Use of a sound level meter to measure user sound exposure when discharging firearms produces quieter results in comparison with systems capable of a 192 kHz sampling specification. The results suggest that data for a Larson Davis LxT1-QPR presented here may not be representative of other systems. Adjustment based on the sampling system should be made for sampling rates less than 192 kHz.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Overview Of Recent Project Work At Meanu

Corjan Buma

This presentation describes some of the more-interesting projects conducted at the MEANU in the recent past. The MEANU conducts (almost exclusively) commercial testing to determine STC-ratings and NRC ratings. While many projects can be considered routine – where a Lab Client needs to quantify the performance of their product or system for purposes of approval or marketing – there are some where an unconventional product or assembly is submitted, or where the ultimate purpose for the results is unusual. Despite the "novel" aspects, the fundamentals do not change : results are not always predictable beforehand, but long-standing principles are always re-affirmed.

Alternative Measurement Techniques For Quantifying Motorcycle Noise

Andy Metelka

There are many worldwide standards for measuring professional motorcycle noise. In general, most refer to near field techniques while complaints still exist, and uncertainty may exist with far field measurements. Both 2-stoke and 4 stroke commercial and professional competitive racing bikes can be quite loud without silencers especially in rural remote areas close to dwellings. The methods proposed utilize new technology including beamforming, tonality, and long-term monitoring for characterizing noise disturbance of motorcycles in urban and rural areas. Past methods may leave reasonable doubt from multiple sound sources, background noise and other conditions; therefore, a method of validation is shown minimizing reasonable doubt. Simultaneous recording, spectrum, octave, tonality, video recording, SLM parameters and weather parameters are explored with examples indicating the relationships and why annoyance is still reported. Remote operation over the Internet offers real-time listening and viewing for 24-hour surveillance and analysis off-line. Methodologies may be deployed with existing standards to enhance reliability and accuracy.

Wind Turbine Noise Relationships Of Tonality, Infrasound, Amplitude Modulation And Audible Noise

Andy Metelka

The measurement of Wind Turbine noise creating annoyance and disturbance is ongoing in the province of Ontario and throughout the world. With over 5000 Wind Turbines situated in rural Ontario measurement uncertainty still exists. Advancement with processing and sensor technology simplify measurement methods allowing on-site real-time accurate assessment of intermittent noise components that quantify the nature of the disturbance beyond NPC-350 guidelines. Continuous monitoring with the same technology show relationships between tonality, infrasound, amplitude modulation and audible noise. Accurate assessments can be made on-site. Also 24-7 monitoring with the same technology with modified setups reduce the requirement of witnessed measurements and the lengthy requirements of post processing, editing and filtering. Nearfield vs. Far field examples indicate how these noise sources, under certain conditions, propagate into the home, far field as well as infrasound into the extreme far field.

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Production Study of Spanish Spirantization in Naturalistic Speech

Gracellia Purnomo *[1], Gloria Mellesmoen †[1], Arian Shamei ‡[1] and Bryan Gick #[1,2]
[1]Department of Linguistics, University of British Columbia
[2]Haskins Laboratory, New Haven

1 Introduction

Spirantization refers to an alternation between plosives and fricatives in speech production. This process is often labelled as a case of phonetic reduction, wherein oral closure movements may contribute to the failure to reach the target (a failed or reduced form) or muscles implicated in production of a full occlusion fail to achieve full activation. In Spanish, spirantization commonly affects the voiced plosive series (at velar, dental, and bilabial places of articulation). Thus, a spirantized /d/ would often be transcribed as an interdental fricative [ð]. This observation is cited as a case of phonetic undershoot [1] or articulatory ease [2,3]. Literature describes spirantization as occurring post- and intervocalically in Spanish [3,4], often following non-stressed vowels [5], within prosodic constituents [6] and during higher speaking rates [7].

However, there is an important aspect of Spanish spirantization that the literature does not address. The spirantized form [ð], an interdental fricative, differs not only in manner, but also in place of articulation from [d]. While it may be possible to account for this difference as undershoot in a 2-dimensional midsagittal space (e.g., [1]), an undershoot account becomes harder to support in 3 dimensions.

In a reduction account, spirantization is posited to be the result of failure to either achieve activation. If we were to measure a series of productions, we would expect to observe optimal target achieved most frequently, and increasing levels of "failure", (i.e. lenitions), would occur less frequently (in a /d/ target, more visible tongue). We might model this approach in a histogram as a unimodal distribution (Hypothesis 1 in Figure 1).

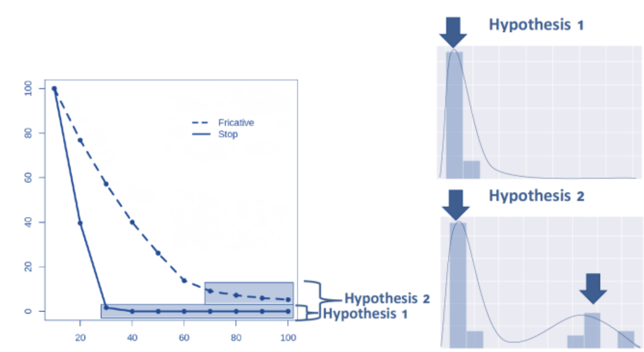

Figure 1: Schematic model distributions for each hypothesis, image adapted from [8]. Graphs on right represent the size of opening (y axis) corresponding with muscle activation (x axis)

* gracellia@alumni.ubc.ca
† gloria.mellesmoen@ubc.ca
‡ arian@alumni.ubc.ca
# gick@mail.ubc.ca

We can also consider a categorical approach to describe the spirantization process. This account would suggest that these spirantized plosives represent distinct targets (i.e., a motor abundance or modularization account [8,9]). If we were to consider a broad sampling of these productions where there are two potential targets, the most frequent productions would correspond to the targets, with production failures associated with each target. Thus, we can expect a bimodal distribution in a plotted histogram (Hypothesis 2 in Figure 1).

We propose that it is this second description that accounts for the spirantization of dental plosives in Spanish.

2 Method

2.1 Participants

Twelve videos of native Spanish speakers were extracted from the video platform YouTube. These videos were in vlog ("video blog") format in which speakers are recorded in frontal headshot format and normally produce extended running speech. This format was chosen in order to obtain the most naturalistic speech without sacrificing facial feature visibility. The speakers were located in Latin American countries (6) and in Spain (6). Each speaker was examined for productions of voiced (50 tokens), voiceless (20 tokens), and nasal (20 tokens) dental stops.

2.2 Annotations and Measurements

Each token was annotated for produced sound, stress patterns, and phonological environment. The frame was extracted at the peak of utterance (Figure 2). Frames were uploaded on ImageJ and measured for area of protruding tongue. Measurements were recorded in pixels, then normalized using the distance between the outer corners of the eyes, and converted to cm^2

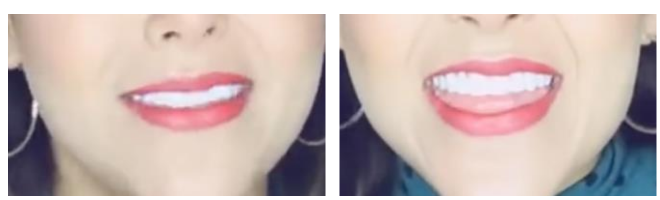

Figure 2: Non-spirantized (left) vs. spirantized (right) tokens of /d/

3 Results

3.1 Descriptive analysis

Results of the annotations of stress patterns and phonological environment showed no distinct pattern, occurring in free variation. Voiced tokens were spirantized far more frequently than the voiceless and nasal tokens. A chi-squared analysis on counts of spirantized vs. non-spirantized tokens indicated that differences observed across target sounds were significant ($p < 0.001$).

Table 1: Proportion of spirantized tokens within each target sound

	/d/	/t/	/n/
Spirantized	58.6%	3.3%	0.8%
Plosive	41.4%	96.7%	99.2%

3.2 Visible tongue analysis

The average visible tongue area for [d] tokens measured 0.142 cm^2, while average tongue area for [ð] tokens was 2.251 cm^2. ANOVAs were performed on area measurements, within and across speakers. Results were significant across the three target sounds ($p < 0.001$) as well as between spirantized and non-spirantized /d/ tokens ($p < 0.001$). Each discrete measurement was plotted on a histogram (Figure 3).

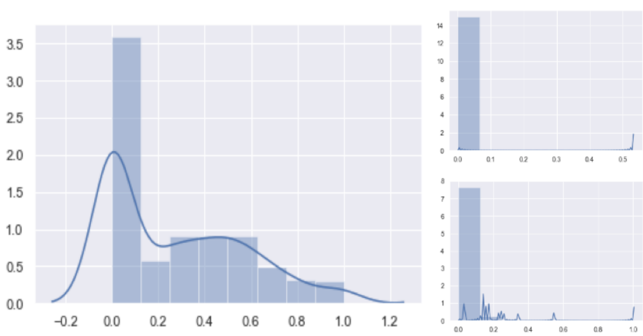

Figure 3: Distribution of each target (/d/, left; /n/, upper right; /t/, lower right) for all speakers combined, with proportion of visible tongue along the x axis and kernel density estimation on the y axis

In Figure 3, the plot of measured tongue area by token indicates a bimodal distribution in the voiced dental tokens, consistent with hypothesis 2. This contrasts with the strongly unimodal distributions of both the /n/ and /t/ tokens, which show little to no failure to reach target. These results are consistent with a view in which the spirantized variants of these /d/ tokens are not the result of weakening but rather the output of a separate module, consisting of a different set of muscles.

3.3 Biomechanical simulation

To further test this, we modeled productions of [d] and [ð] with a forward activation model using the BadinJawTongueHyoid model in the three-dimensional biomechanical simulation software Artisynth (http://www.artisynth.org/). Formation of the dental stop closure required activation of the superior longitudinal, transversus, and posterior and medial genioglossus muscles. In contrast, modelling of the interdental fricative required substantially increased activation of the transversus, verticalis and all portions of the genioglossus (anterior, medial, posterior). The exact parameters utilized in the simulation of both productions are described in Table 2 below. Crucially, we were unable to produce a [ð] configuration by reducing the activations used in the production of [d].

Table 2: Muscle activations used for simulation of [d] and [ð]

	SL	TRANS	VERT	GGm	GGp	GGa
[d]	30%	25%	0%	15%	20%	0%
[ð]	30%	45%	30%	45%	50%	10%

4 Discussion

The above simulation results are unsurprising considering that the tongue posture visible in the spirantized variant in Figure 2 does not appear to be an intermediate point along a trajectory towards the non-spirantized [d] configuration in Figure 2.

The current set of studies supports the hypothesis that spirantized and non-spirantized variants of the Spanish dental plosive result from activating distinct sets of muscles, each with its own end target.

Acknowledgments

The authors would like to thank Katherine Rivera for providing video transcriptions. Work funded by NSERC RGPIN-2015-05099 to the last author.

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SPECTRAL MOMENTS TO DESCRIBE FRICATIVE EMERGENCE OF FRENCH-QUEBEC CHILDREN

Carolina Salinas-Marchant *[1], Geneviève Meloni †[2] and Andrea A.N. MacLeod ‡[2]
[1]École d'orthophonie et d'audiologie, Université de Montréal, Montréal.
[2]Laboratoire de Psychologie et de NeuroCognition,, Université Grenoble Alpes, Grenoble.
[3]CHU Sainte-Justine Centre de Recherche, Montréal.

1 Introduction

In the study of fricatives, various acoustic measurements have been described to objectively classify these consonants. One of the measures most used by researchers is the analysis of spectral moments [1 - 4]. Spectral moments describe the distribution of frequencies in an area of the spectrum, with the aim of identifying the energy concentration patterns [2, 5]. There are four spectral moments used to describe these patterns: center of gravity, standard deviation, skewness and kurtosis [2]. The acquisition of fricative consonants is well documented in English-speaking children [6] However, studies conducted in languages other than English, such as Japanese, have revealed differences in the order of acquisition of these consonants, independent of the status of the speech sounds in the phonology of the language and despite little articulatory variance [7, 8]. These differences would indicate that some phonological development trends are language-specific [9]. Studies carried out in the area of production show that the realization of fricative consonants is different between children and adults [2, 5]. In addition, there is evidence of great articulatory variability among subjects in children [10]. In this study, we aim to obtain acoustic measurements that describe the phonetic-phonological development of fricative consonants in French from Quebec.

2 Method

2.1 Participants

This cross-sectional study is developed on the data of 47 monolingual children, divided into 6 age groups: 2;6, 3;0, 3;6, 4;0 and 4;6, from Quebec City, Canada. According to the parent's report, the children recruited presented typical cognitive, motor and linguistic development. The Peabody Picture Vocabulary Test (PPVT), in its French version [11], was used to screen for typical language development.

2.2 Methods

The evaluation was conducted in a room with sound treatment. Audio recording was performed with a lapel microphone inserted in a child-friendly vest (Audio Technica 600) connected to an audio preamplifier, connected to a computer (44.1 kHz sample rate, 16 bits per sample). The children were evaluated by a trained research assistant. The children's productions of the fricative consonants were obtained through a picture naming task. Subsequently, the recorded observations were labelled and transcribed phonetically, with a double entry of 10% of the corpus.

* carolina.salinas.marchant@umontreal.ca
† genevieve.meloni@gmail.com
‡ andrea.macleod@umontreal.ca

2.3 Analysis

An acoustic analysis of the target words was performed using PRAAT [12]. A window of 200 ms was used to see each targeted word and measurements were taken in that window size. To determine the beginning and the end of each fricative, the segmentation decisions considered the simultaneous consultation of the waveform and the wideband spectrogram, the occurrence of high frequency aperiodic noise and the rapid increase of cross by zero [3,4]. The statistical analysis was developed with R [13] for two reasons: first, due to the use of variables to fixed effects and to random effects; second, to better manage the possible missing values due to the complexity of recording data with children.

3 Results

The analysis by mixed models show an effect of the point of articulation on the center of gravity and on the duration. Moreover, the analyses showed an effect of the point of articulation on the standard deviation. An effect of the voiceless on the center of gravity and on the duration is also observed.

4 Discussion

Preliminary results indicate that the centre of gravity is the most robust spectral moment to classify the fricative consonants [4] of these French-speaking preschool-aged children. This result coincides with previous literature, which suggests a common key acoustic cue for fricatives. On the other hand, duration, a measure often described as secondary [4], appears to be an important indicator for the classification of fricative consonants produced by these French-speaking children, which suggests a language-specific cue is also in play.

Acknowledgments

This study was conducted with the ethical approval of the Center for Interdisciplinary Research in Rehabilitation and Social Integration, of the Province of Quebec.

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L2 Production of American English Vowels in Function Words by Spanish L1 Speakers

Scott J. Perry *[1] **and Benjamin V. Tucker** †[1]
[1]Department of Linguistics, University of Alberta

1 Introduction

Studies investigating the pronunciation of second language (L2) speakers have focused on the production of individual segments in stable environments (e.g., vowels in the stressed syllables of lexical words). As such, there is a need for further investigation into the L2 pronunciation of segments which are more variable and subject to phonetic and phonological processes. The present study addresses this gap by investigating the production of vowels in function words in American English (hereafter English) by native (L1) speakers of Spanish.

Vowels contained in English function words are often reduced [1] (they move toward the center of the F1 × F2 space), while those in Spanish are less so [2]. Thus, the task faced by the Spanish speaker learning English is not only to acquire a new vowel inventory, but also to learn the process of reducing and/or centralizing these vowels in function words contrary to the orthographic input they receive.

2 Method

The data from this study was obtained from the Speech Accent Archive [3]. The methodology concerning the recording process is available online on the corpus website.

2.1 Participants

Participants in this study included L1 speakers of American English (n=30) and L1 Spanish speakers who are L2 speakers of English (n=62). All L2 speakers had resided in the United States for a minimum of 1 month. Dialect was not controlled for as the location of residence was not reported for L2 speakers. The L2 learners were split into two groups based on when they started learning English to examine the effect of age of onset of acquisition (AOA). The Early AOA group (n=30) was comprised of those who started learning before the age of 13 and the Late AOA group (n=32), contained participants who started learning English after the age of 13.

2.2 Analysis

The recordings were imported into Praat [4] where the vowels were segmented by hand using the spectrogram and waveform to determine vowel onset and offset. The formant values were then extracted from the vowel midpoint using the Burg algorithm. The maximum number of formants was always set to 5 and the formant ceiling was set to the value at which the formant tracker best aligned with visible formants, between 4800-5100Hz for men and 5400-5800Hz for women. The formants were then normalized using the "Nearey Intrinsic" normalization method in PhonTools [5].

In order to measure the degree of neutralization of the vowels in function words and lexical items, pairwise comparisons were done between vowels within each group by use of MANOVA, which output a Pillai score that can be used as a measure of vowel overlap [6]. A Pillai score of 0 signifies that the two vowels are indistinguishable and a Pillai score of 1 indicates that the two vowels are completely distinct, showing no overlap.

3 Results

3.1 Vowel Spaces

The vowel plots for the three experimental groups are in Figure 1 below. Vowels produced in lexical items are on the left, vowels from function words are on the right. The English native speakers are on top, Low AOA L2 learners in the middle, and High AOA L2 learners on the bottom.

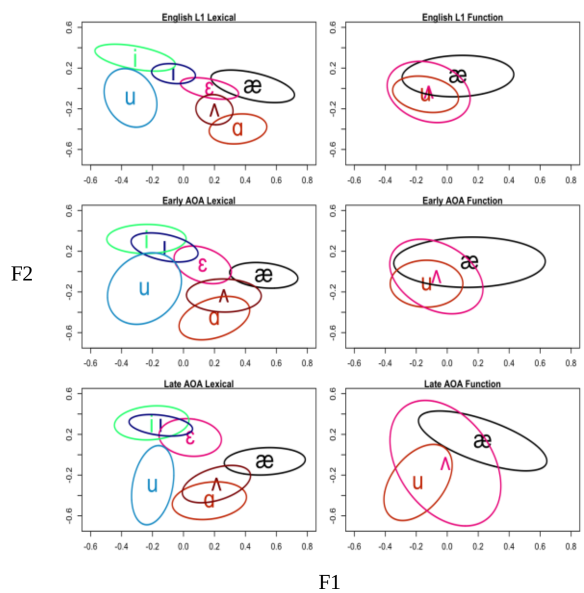

Figure 1: Vowel plots with normalized formant values for different word types and experimental groups. Mean values are plotted along with ellipses of 2 standard deviations.

3.2 Pillai Scores

The MANOVAs run for each pair of vowels present in both lexical and function words each output a Pillai score, used here as a measure of vowel overlap. The Pillai scores of the vowels produced in function words are listed by group in

* sperry1@ualberta.ca
† bvtucker@ualberta.ca

Table 1. The same vowels produced in lexical items appear in Table 2.

Table 1: Pillai scores: comparisons of vowels in function words.

Vowel Pair	Eng L1	Early AOA	Late AOA
ʌ - æ	0.716	0.624	0.796
ʌ - u	0.045	0.220	0.483
æ - u	0.694	0.661	0.823

Table 2: Pillai scores: comparisons of vowels in lexical words.

Vowel Pair	Eng L1	Early AOA	Late AOA
ʌ - æ	0.857	0.782	0.743
ʌ - u	0.933	0.834	0.850
æ - u	0.940	0.921	0.926

The Pillai scores confirm the general trends that can be seen in the vowel plots. We see that for L1 English speakers, /ʌ/ and /u/ produced in function words were practically indistinguishable, while /ʌ/ and /æ/ along with /æ/ and /u/ show less overlap. These same vowels do not overlap to the same extent when produced in lexical items by L1 English speakers.

The Early AOA group patterned closely with the L1 English group, with /ʌ/ and /u/ showing a great deal of overlap, though less than the L1 group. The other function word vowel pairs show less overlap, like the L1 group. All pairs of vowels show less overlap in lexical items.

The Late AOA group produced more centralized vowels in function words than they did in lexical items, but not to the extent that the L1 English or Early AOA L2 learners did. For the Late AOA group, /ʌ/ and /u/ also show the most overlap, like the other two groups. The other vowel pairs show only small differences across lexical items and function words.

4 Discussion

The goal of this study was to examine the L2 production of vowels in English function words by Spanish L1 speakers. Spanish vowels in unstressed syllables are only slightly centralized [2], and therefore Spanish speakers learning English must learn a centralization pattern of vowel reduction in their L2. This could be considered a difficult task considering these vowels show ample variability [7].

The L1 Spanish speakers who began learning English at a younger age patterned closely with the native English speakers. However, they showed more variability with their production of vowels in function words which led to less overlap between /ʌ/ and /u/ than the L1 English speakers and more overlap between those two vowels and /æ/, again mostly likely due to increased variation in their productions.

There are two possible explanations for why the High AOA L2 speakers were able to centralize vowels in function words. The first option is that this change in vowel quality is the same change that occurs in Spanish, and they are directly transferring the pattern from their L1 to their L2. The second option is that the learners have learned to reduce vowels in function words more than they do in their L1.

The first option cannot be evaluated properly, as there is a dearth of acoustic studies that have investigated Spanish vowel quality across stressed and unstressed syllables. The few studies that have been done have shown evidence of centralization, but not to the same degree that occurs in English or that was produced by the High AOA group.

The second option, that L1 Spanish speakers have learned a new pattern of reduction in their L2, is also possible. It would be possible for increased centralization to occur without Spanish L1 speakers accurately perceiving spectral differences. Based solely on the concept of undershoot [8], the L1 Spanish speakers would only have to perceive that the durational difference between stressed and unstressed syllables in English is larger than it is for Spanish, and produce relatively shorter syllables in unstressed positions. An advantage has been shown for learning durational differences in L2 Speech [9], making this the most plausible explanation for increased overlap by the late L2 learners.

5 Conclusion

The results of this study align with the large body of literature that reports an advantage for early L2 learners. It also suggests that spectral properties are also affected by the interaction between temporal and articulatory constraints in L2 speech production.

Acknowledgments

Thanks to Matt Kelley for code to compute Pillai scores and to all other members of the Alberta Phonetics Lab.

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Speech Perception and The Role of Semantic Richness in Processing

Filip Nenadić[*1], **Matthew C. Kelley**[†1], **Ryan Podlubny**[‡1], **and Benjamin V. Tucker**[§1]
[1]Department of Linguistics University of Alberta

1 Introduction

Semantic richness is ordinarily defined as a construct encompassing indicators of variability in a given word's meaning. Higher semantic richness has consistently been connected to facilitation in processing, at least when recognition tasks are used [1]. These findings, however, have come largely from studies based on visual word recognition, and related studies focusing on the acoustic signal and speech perception are less common. Effects observed may vary across modalities, as is exemplified in the opposite effect of *phonological neighborhood density* when the visual and the auditory lexical decision tasks are compared [2]. Furthermore, models of spoken word recognition tend to focus on the activation-competition process concerning some representation of the acoustic input [3], and disregard how the activation of networks of meaning based on that very acoustic input shapes competition and facilitates or inhibits speech perception. In other words, very little is known about how semantic richness interacts with perception of the acoustic, speech signal.

One study that investigated the effects related to a multitude of semantic variables in the auditory modality was conducted by Goh et al. [4]. They found linear effects of *concreteness* and the *number of semantic features* and a quadratic effect of *valence* — all of which facilitated spoken word recognition in the auditory lexical decision task. *Arousal*, *semantic neighborhood density*, and *semantic diversity* had no effect. However, stimuli used by Goh et al. were a limited set of under 500 words, most of which were concrete nouns. The goal of the present study is to expand the scope of the analysis to a larger number of words by using the data collected in the Massive Auditory Lexical Decision (MALD) project [2] and to further explore semantic richness effects as they may pertain to the perception and processing of spoken language.

2 Method

2.1 Massive Auditory Lexial Decision

Responses were collected from 231 native monolingual listeners of western Canadian English (78% female, 22% male; age ranged from 17 to 29, M = 20.11, SD = 2.39). Participants were recruited at the University of Alberta and received partial course credit for participation.

The stimuli were 26,800 English words and 9,600 phonotactically licit pseudowords recorded by one male speaker of western Canadian English. Stimuli were further organized into 134 experimental lists, with each list containing 400 word and 400 pseudoword items.

*nenadic@ualberta.ca
†mckelley@ualberta.ca
‡podlubny@ualberta.ca
§bvtucker@ualberta.ca

Participants completed the experiment in E-Prime [5], in a sound-attenuated booth. Their task was to listen to a single MALD experimental list and decide whether each of the stimuli presented in random order is a word of English or not, and indicate so by pressing the appropriate button on the button-box. Participants could attend as many as three sessions and would provide responses for a single MALD list every time. The total number of completed sessions was 284.

2.2 Data preparation and analysis

For the analysis, we selected responses to those MALD words that had available estimates for the variables *concreteness*, *valence*, *arousal*, *semantic neighborhood density*, and *semantic diversity*. The variable *number of semantic features* was not included as the values were available only for the limited set of words used in Goh et al. [4]. The number of retained words was 9,086, which is an 18-fold increase in comparison to the Goh et al. study. Furthermore, the words retained in this sample had a much wider range of values on the semantic richness variables (whereas Goh et al. study mostly included, e.g., high *concreteness* and low *arousal* words).

We then removed all incorrect responses and all responses shorter than 500 ms or longer than 2,000 ms (5.75% of the data excluded). Log-transformed response latency was used as the dependent variable in a generalized additive mixed-effects model [6]. The model included smoothed effects of participant *age*, *trial* number, stimulus *duration* in milliseconds, *phonological neighborhood density*, *word run length* (i.e., the number of consecutive word stimuli prior to the trial), *concreteness*, *arousal*, *valence*, and *semantic diversity* as predictors. Additionally, we included *frequency weighted semantic neighborhood density* as a predictor. Weighting was performed by multiplying log-transformed *frequency* from the Corpus of Contemporary American English [7] and *semantic neighborhood density* to avoid issues of multicollinearity because the two variables correlated highly ($r = .78$). The new variable correlated highly with both logged *frequency* ($r = .95$) and *semantic neighborhood density* ($r = .91$). All predictor values were standardized. Finally, the model included participant *sex* as a factor, as well as random effects of word stimulus and random slopes of *trial* by participant.

3 Results

The results of the generalized additive mixed-effects model showed that there was no significant effect of participant *sex*, while there was a small effect of participant *age*, as older participants were slightly faster in their responses. The effects of non-semantic predictors matched the ordinary findings — response latency was shorter in later *trials*, shorter word *duration*, words with smaller *phonological neighborhood density*, and longer *word run length*.

Where semantic richness variables are concerned, only the effect of *semantic diversity* was not significant. *Frequency weighted semantic neighborhood density* had a negative linear effect (Figure 1a). We also find that high *concreteness* is facilitatory, while low *concreteness* is inhibitory (Figure 1b). When *valence* is considered, extreme values (i.e., both negative and positive *valence*) are connected to shorter response latency (Figure 1c). High *arousal* predicted longer response latency, while there was no difference between words with average versus low *arousal* (Figure 1d).

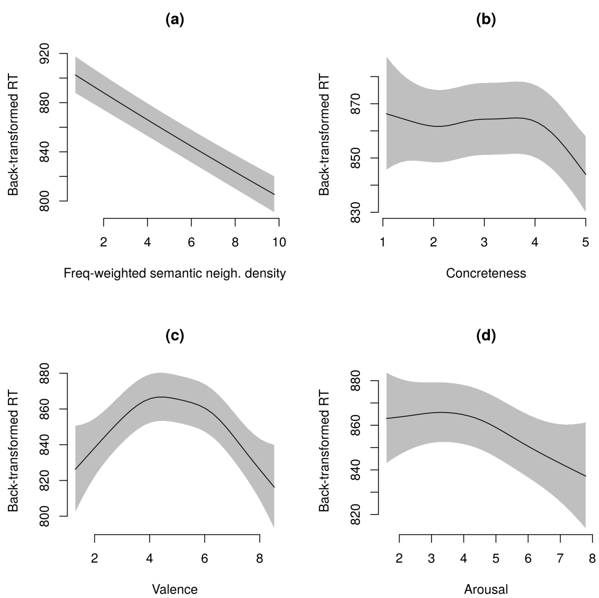

Figure 1: Effects of the four significant semantic predictors of response latency.

4 Discussion

Our results differ in some important ways from those of Goh et al. [4]. The effect of *valence* in our study is not exactly quadratic, but that is the smallest change we note. We find a non-linear effect of *concreteness* — besides in high values covered in Goh et al., we find the same effect in low values, but not in words with average *concreteness*. Furthermore, we record significant effects of *arousal* and *frequency weighted semantic neighborhood density*. The effect of *arousal* likewise seems to be explained by the distribution of this variable — we find an effect in higher *arousal* values, whereas the limited dataset previously used by Goh et al. mostly included words with low *arousal*. For *semantic neighborhood density*, since the distributions in the two datasets are comparable, perhaps the effect is significant simply due to larger sample size.

Although the relationship is not always linear or simple, we find that higher values in variables that operationalize semantic richness tend to be connected with faster response latency. Given these findings, we claim that at least some semantic characteristics of a word play a role in speech perception, facilitating processing. As the acoustic signal unfolds and competitor words are activated, word meanings are accessed, not just their form. This process would arguably be highly adaptive, as it may allow the listener to quickly access items that carry more (and more important) meaning. In comparison, most models of spoken word recognition treat the mental lexicon as a simple list of words [3], focusing solely on the process of matching (acoustic) input to abstract representation when describing speech perception.

5 Conclusions

Our results indicate that semantics can play a role in speech perception even when isolated words are presented. We argue that models of spoken word recognition should treat words in the mental lexicon as more than simple lists of items by taking into account their semantic richness and semantic connections. We also show that additional studies using large word samples with an encompassing variability of word characteristics are needed [8], especially given the discrepancies in the results between word sets of different sizes.

Acknowledgments

This project was funded by the Social Sciences and Humanities Research Council: Grant #435-2014-0678.

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Forced-alignment of the Sung Acoustic Signal Using Deep Neural Nets

Dallin Backstrom[*1], **Matthew C. Kelley**[†1], **and Benjamin V. Tucker**[‡1]
[1]Department of Linguistics, University of Alberta

1 Introduction

Sung speech shows significant acoustic differences from spoken speech. One challenge in analyzing both spoken and sung speech is identifying the individual speech sounds. Forced-alignment systems such as P2FA [1] and the Montreal Forced Aligner [2] have been designed to accomplish this task for spoken speech, however, there is no such tool for sung speech. Previous work used a combination of hidden Markov models and convolutional neural networks on log-Mel filterbanks to segment phones in sung Mandarin opera [3]. We, in turn, trained a deep neural network to extract phone-level information from a sung acoustic signal. The primary objective was to create a model that can take a WAV file containing a target song as the input, and produce time-aligned phonemic labels automatically as output. To measure the performance of our model on these tasks, we primarily measured accuracy on identifying the correct phone label at a given time-step. We also compared the accuracy of our model to other state of the art systems, trained on spoken speech, performing the same task with sung speech.

2 Method

We used a selection of traditional Canadian folk songs from the Moses and Frances Asch Collection [4] to train our model. We selected songs from the collection that had either no or minimal accompaniment. There were two unique singers in the data set, one male [5] and one female [6].

2.1 Building and Training the Model

Time-aligned labels for the data were created by manually transcribing the songs using a Praat TextGrid [7]. Approximately 30 minutes worth of songs were transcribed in total. Twenty-five (25) millisecond windows of audio spaced 10 milliseconds apart were then extracted from the recordings. The label for which phone class the window belonged to was determined based on the time-aligned transcription. To increase the amount of data a second copy of each window was created by adding Gaussian noise to the audio and then extracting the windows and labels again. We also used the TIMIT [8] spoken speech data set.

Using the Keras deep learning library with the Tensorflow backend, we built several models to train on our data. Our most successful model had an architecture comprised of two convolutional layers, four bidirectional LSTM layers, and a time-distributed result layer. The first convolutional layer has a filter size of 4 and 512 filters with a stride length of 1. The second convolutional layer has 256 filters and a filter size of 4 with a stride length of 1 and a dilation rate of 2. Both convolutional layers used Keras's "causal" padding option. The convolutional layers are both immediately followed by a spatialized dropout layer at a rate of 30% and a batch normalization layer with default parameters. The first, second and fourth Bidirectional LSTM layers had a size of 256, while the third had a size of 512. Default batch normalization was again used and the LSTM layers all had Gaussian dropout at a rate of 60%. All other parameters were left at their default values.

The model was trained with categorical cross-entropy loss. Its framewise phone recognition accuracy was monitored during training to identify the best performing model. The Adam optimizer was used, with the default configuration. This model was trained first for 10 epochs on our singing dataset, then for 10 epochs on the TIMIT corpus of spoken speech, and then was trained again on our singing dataset for another 10 epochs. The batch size for each of these training routines was 1. By training on TIMIT, the model was exposed to a far greater amount of data than it would otherwise have seen, and because the TIMIT data is formatted in the same way as the singing data, the model is more robust as a result.

3 Results

Table 1: Accuracy of Different Model Architectures

Model Architecture	Highest Accuracy
Convoluional	18 %
Convolutional + LSTM	53 %
Convolutional + Bidirectional LSTM	81 %

Our best performing model achieved training accuracies of approximately 80 percent (see Table 1). However, ultimately the output that we want to achieve is not a list of the most probable phone every 10ms; rather, we want to know where the boundaries fall between each phone. To do so, we use the decoding algorithm specified by Kelley & Tucker [9], but using standard backtracking instead of the argmax routine, to produce time stamps for the boundaries of each phone. Of course, if the model had correctly labeled every phone with 100% confidence at each time-step, this task would be trivial, which is why we used accuracy as a training and evaluation metric.

Figure 1 presents a sample alignment of a short segment of a song. Analysis of the quality of the automatic transcriptions overall is more difficult, but as demonstrated in Figure 1, vowels are segmented acceptably well, and certain other segments are also consistent, such as the burst releases of /t/

*dbackstr@ualberta.ca
†mckelley@ualberta.ca
‡benjamin.tucker@ualberta.ca

and /d/ and sonorants like /ɹ/. In fact, the model tends to overestimate its confidence in these segments and compress all of the phones that it is not as confident about into a tiny margin in between the ones that it is more confident in. This is an undesirable behaviour, but we encountered this same tendency, to minimize certain phones in models used on this task, for regular speech. Beyond this problem, the model does place certain boundaries in nearly the same position as the hand-alignment, which is encouraging, as placing boundaries with high accuracy is necessary for the model to be useful in automating the alignment task.

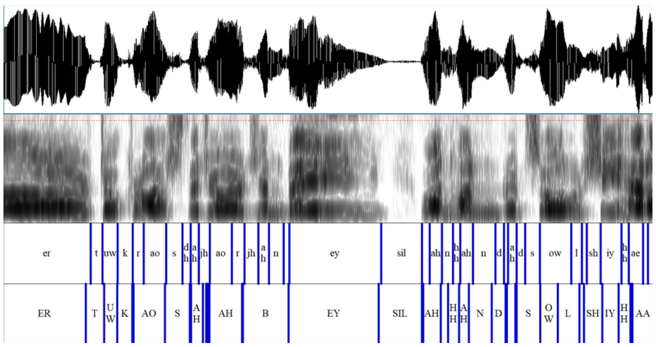

Figure 1: Sample alignment of a section of a song. From top to bottom: Waveform, spectrogram, target transcription, model transcription.

4 Discussion

One important question that arises from our results is why, despite achieving good frame-wise accuracy rates in the training set, the alignments produced by the model are so bad in certain places. Manual inspection of the alignment suggests that the model seems to almost ignore certain phones entirely, which is most likely due to it not having as much confidence in being able to identify those phones correctly when compared to sonorants and bust releases. If its confidence is spread between many classes, it will have a hard time deciding which one it should select, and may ignore those lower probability phones in favour of items on which the model reaches a much higher level of confidence. For this reason it may be very useful for us to look at the way that we translate the labels outputted by the model into a Praat TextGrid, as there might be a way to do this that takes into consideration the fact that some classes of phone will have overall lower levels of probability than others due to them appearing more similar to other options. Minimum and maximum duration constraints, for example, on the alignment algorithm should help ameliorate this behavior.

Future work should focus on determining why the model is making classification mistakes. A confusion matrix or inverse layer maximization may help indicate what kinds of mistakes the network is making.

5 Conclusion

In this project, we achieved our main initial goals of creating a model that is able to classify phones when presented with singing as well as produce a time-aligned Praat TextGrid that can be compared to the original audio track. Some of our other goals, such as using various audio pre-processing methods to compare their effectiveness, we did not achieve. Despite the difficulty in creating from scratch a dataset that would be sufficiently large to adequately train a model for this challenging task, we were able to create a model that is able to classify phones with some degree of accuracy. Excitingly, this suggests to us that this task, although more difficult than the same task performed on regular speech, can be handled in a similar way and with similar levels of success.

Ultimately, we hope to improve the model and the automatic phone alignment further, until it can be packaged as a bespoke application for use in research. We also plan to explore the many questions about how our model can classify sung phonemes, and what differences and similarities it might have with humans, as we continue to test and improve upon it.

Acknowledgments

This project was made possible through an Undergraduate Research Grant from Sound Studies Institute at the University of Alberta, and their stewardship of the Moses and Francis Asch collection of folk music. We also acknowledge the support of the NVIDIA Corporation with the donation of the Titan X Pascal GPU used for this research. The authors would also like to thank the members of the Alberta Phonetics Laboratory for their support and suggestions.

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Effects of Modality and Linguistic Materials on Memory In younger and older adults

April Pereira *[1], Jenna Pattison †[1], M. Kathleen Pichora-Fuller ‡[1] and Sherri L. Smith ♦[2]
[1]Department of Psychology, University of Toronto, Mississauga, Ontario
[2]Division of Head and Neck Surgery and Communication Sciences, Duke University School of Medicine, North Carolina

1 Introduction

Sensory factors can affect recall [1]. Visual tests are immune to hearing loss, but auditory tests are more ecologically relevant for evaluating speech understanding [2]. Previously, an auditory test was preferred to a visual test because it yielded a greater range of working memory scores [2]. The purpose of this study was to evaluate the relative effects on recall of auditory versus visual materials with simple versus complex linguistic properties.

2 Method

2.1 Participants

The 64 participants were native English speakers with good health and normal hearing (audiometric thresholds ≤ 25 dB HL at and below 3 kHz and no asymmetry ≥ 20 dB. The younger 32 participants (mean age = 19.9 years, *SD* = 1.8, range 18-26; 5 male, 27 female) received course credit for participating. The older 32 participants were recruited from a pool of research volunteers and received $12/hour (mean age = 71.9 years, *SD* = 5.6, range 60-83; 5 male, 27 female).

2.2 Procedures

Audiograms were obtained using standard clinical methods. Participants completed a survey to report years of education and levels of language fluency. All participants completed four test conditions which were counter-balanced (2 modalities x 2 linguistic levels): simple auditory, complex auditory, simple reading, complex reading. Participants repeated sentence-final target words. In simple conditions, each target word was preceded by the same carrier phrase; in complex conditions, each was preceded by a unique sentence. In each condition, 100 items were presented, with five trials in each of five setsizes (2, 3, 4, 5, 6 words to be recalled per set). The number of words correctly recognized, judged, and recalled was measured.

2.3 Materials and apparatus

The complex auditory and visual stimuli were sentences from the Revised Speech Perception in Noise (SPIN) Test [3]. The simple auditory and visual stimuli were from the Word Auditory Recognition and Recall Measure (WARRM) [4]. Auditory conditions were completed in quiet in a sound-booth. Visual conditions were presented on a computer screen and completed in a quiet office environment.

3 Results

3.1 Audiometry

Younger adults had clinically normal (≤ 25 dB HL) audiometric thresholds at all frequencies. Older adults had normal thresholds at frequencies ≤ 4 kHz and had typical age-related high-frequency hearing loss.

3.2 Younger adults

There were significant main effects of 1) linguistic complexity ($F(1,31) = 81.0$, $p < .001$, $\eta p^2 = .72$), with higher recall for simple than complex stimuli, 2) modality ($F(1,31) = 57.1$, $p < .001$, $\eta p^2 = .65$), with higher recall for auditory than visual stimuli; and 3) setsize ($F(4,124) = 336.5$, $p < .001$, $\eta p^2 = .92$), with recall decreasing with increasing setsize. There was a significant interaction between linguistic complexity and setsize ($F(4,124) = 9.3$, $p < .001$, $\eta p^2 = .23$) and modality and setsize ($F(4, 124) = 14.4$, $p < .001$, $\eta p^2 = .32$). As setsize increased, the differences in recall due to complexity and modality increased (Figure 1). There was no three-way interaction. Bonferroni corrected post-hoc tests of multiple comparisons revealed that recall scores in all setsizes were significantly different from each other. Collapsed across modality, recall was better for word-level than sentence-level materials by 8 percentage points. Collapsed across linguistic complexity, recall was better for auditory than for visual materials by 9 percentage points. For total scores, recall was better for word- than sentence-level materials by 9 percentage points and recall was better for auditory than visual materials by 11 percentage points.

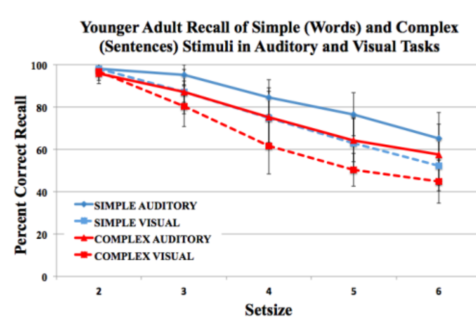

Figure 1: Percent correct recall by setsize for target words in word-level and sentence-level materials during listening and reading tests for younger adults. Error bars represent SDs.

* april.pereira@mail.utoronto.ca
† jenna.pattison@mail.utoronto.ca
‡ k.pichora.fuller@utoronto.ca
♦ sherri.smith@duke.edu

3.3 Older adults

There were significant effects of linguistic complexity ($F(1,31) = 89.72$, $p < .001$, $\eta p^2 = .74$), modality ($F(1,31) = 6.16$, $p = .02$, $\eta p^2 = .17$) and set size ($F(4,124) = 275.27$, $p < .001$, $\eta p^2 = .89$), with recall decreasing with increasing set size. There was an interaction of setsize and complexity, $F(4,124) = 6.56$, $p < .001$, $np^2 = .18$. As setsize increased, the differences in recall due to complexity (blue vs. red lines) increased (Figure 2). There was a marginally significant two-way interaction between setsize and modality ($F(4,124) = 2.42$, $p = .052$, $\eta p^2 = .072$). As setsize increased, the differences in recall due to modality (solid vs. dashed lines) increased (Figure 2). There was no significant two-way interaction between modality and setsize or a three-way interaction. Bonferroni corrected post-hoc tests of multiple comparisons revealed that recall scores in all set sizes were significantly different from each other. Using the total scores, recall was better for word- than for sentence-level materials by 7 percentage points and recall was better for auditory than for visual materials by 4 percentage points.

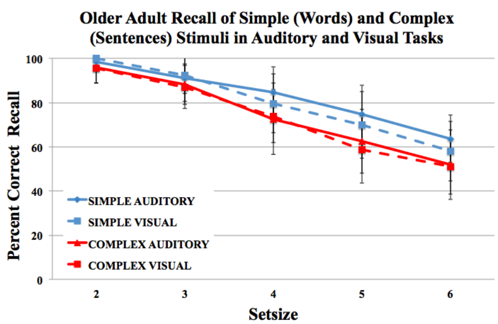

Figure 2: Percent correct recall by setsize for target words in word-level and sentence-level materials during listening and reading tests for older adults. Error bars represent SDs.

3.4 Age Comparisons

There was no effect of group on recall, $F(1,62)=1.343$, $p=.251$, $\eta p^2<0.021$, with an overall score of 72.9% for older and 70.1% for younger adults. There were main effects of setsize, modality, and linguistic complexity. Importantly, there was a significant interaction between age and modality $F(1,62) = 27.433$, $p < 0.001$, $np^2 = .307$), such that modality only mattered for the younger (Figure 3; striped bars), but not older group (Figure 3; solid bars). Younger and older adults scored similarly on the auditory tests but younger adults scored worse on visual compared to auditory tests.

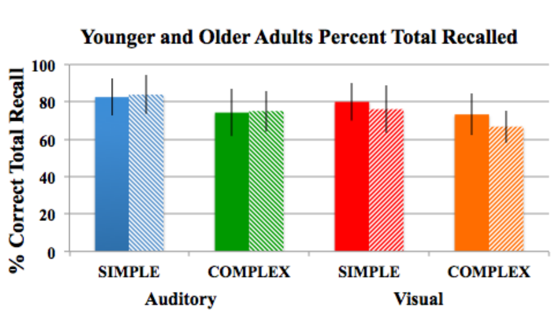

Figure 3: Percent correct total recall for target words in word-level and sentence-level materials during listening and reading tests. The solid coloured bars represent older adults and the striped bars represent younger adults. Error bars represent SDs.

4 Discussion

The current results for older adults differed from previous findings for younger adults. Notably, there was an effect of linguistic complexity for both age groups, but the older group did not demonstrate the modality effect that had been found for younger adults. The older adults performed similarly regardless of modality, with overall scores of 78.4% and 76.6% for auditory and visual tests, respectively. In contrast, the younger adults scored better on auditory (80.0%) than on visual (70.9%) tests. Notably, younger adults performed just as well as older adults on both auditory tests, whereas they performed worse on both visual tests (Figure 3). For younger adults, it is suggested that auditory recall is easier than visual recall because spoken stimuli have direct access to the phonological loop and therefore may require fewer cognitive resources to process than visually presented read stimuli, thus facilitating recall [5]. The younger adults in our study may be demonstrating this auditory advantage (or visual disadvantage), but the older adults did not. Perhaps younger adults found reading to be more challenging than listening, whereas reading and listening were similarly challenging for older adults. Older adults may be more expert readers than younger adults and have an advantage on the reading task; however, they may have sub-clinical declines in supra-threshold auditory processing which would put them at a disadvantage on listening. These findings may or may not apply to people with hearing or vision loss or to non-native speakers or those with low literacy. Further research will need to be conducted in order to generalize these findings to clinical or special populations and to explore if the listening memory test is sensitive to intra-individual differences in performance in various listening conditions (e.g., listening is aided vs. unaided or in quiet vs. in noise).

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Abstracts for Presentations without Proceedings Paper
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Do Mothers Prefer Monkeys Over Nembees?

Stephanie L Archer

Do mothers enhance specific words to children based on the familiarity or novelty of the word? Parents' speech directed to infants (known as IDS) is typically marked by the use of exaggerated pitch, louder intensity, slower speech, and shorter utterances when compared to speech directed to adults (ADS). As early as 2 days old, infants demonstrate a preference for IDS over ADS (Cooper & Aslin, 1990) and this attention to IDS is likely due to salient acoustic information (Vihman & Keren-Portnoy, 2013). However, we do not know whether IDS is beneficial to infants' word learning abilities. In this study, we recruited 16 English-speaking mothers with infants between 18 and 24 months. Of a possible 6 'creature stories', mothers chose 3 creatures familiar to her child to use as stimuli. Novel 'creature stories' were matched with the familiar creatures and were also used as stimuli (e.g., monkey/nembee). In total, mothers were given 6 stories (3 familiar, 3 novel), each composed of 6 sentences, and were asked to read all stories aloud to their child (IDS condition) and to an adult (ADS condition) while being recorded in a sound attenuated booth. Data collected were acoustically analysed for pitch (mean Hz, range Hz). Currently, analyzed data includes only 4 'creature stories' (2 familiar ('monkey', 'chicken') and 2 novel ('nembee', 'jidam')). Each creature name was included in each sentence in each story (n = 479). Preliminary results of a 2x2x3 ANOVA revealed a marginal interaction between familiar/novel, IDS/ADS, and sentence position (initial, medial, final), p = .069. Mothers' pitch was higher when reading familiar targets than novel targets when in initial (p <.0001) and medial (p =.03) position, but not final (p =.263). This might suggest that mothers' use pitch as a way to focus infants' attention, presumably aiding in lexicon building.

Prosodic And Semantic Effects On The Perception Of Mixed Emotions In Spoken Speech

Ayslin Bubar, Tara Vongpaisal

Mixed emotions, those comprising of more than one basic emotion, are frequent occurrences in everyday conversational experiences, and the perception of these complex emotions have important consequences in understanding the non-literal meaning of a linguistic message and the actual emotional state of the speaker. In two experiments, we examined the perception of mixed happy-sad emotions elicited by a combination of prosodic voice cues (pitch and tempo), and sentence content (semantics) in spoken speech. In Experiment 1, using a 6-point Likert scale measuring happiness and sadness, participants rated sentences with happy, sad, and neutral semantic content spoken by a female talker. Prosodic cues that characterize happy (high pitch, fast tempo) and sad (low pitch, slow tempo) emotions elicited the highest ratings when these cues were congruent with the semantic content of sentences. When these prosodic cues were incongruent with sentence meaning, there was less differentiation in emotion ratings indicating that sentence meaning interacted with the perception of emotions elicited by prosodic cues alone. Mixed emotions (high pitch, slow tempo; and low pitch, fast tempo) received intermediate ratings and were less affected by the semantic content of sentences. In Experiment 2, eye-tracking was used to examine the pattern of attention directed toward dynamic facial information in the talker's expressions of happy, sad, and mixed emotions. Overall, participants attended more to the eye regions and less to the mouth regions of the face. Furthermore, attention to the eye region was not influenced by semantic content of speech but was affected by prosodic pitch and tempo cues with a higher number of fixations occurring for high pitch and slow speech. In contrast, attention towards the mouth region was influenced by semantic content with a decreasing number of fixations occurring across sad, happy, and neutral content, respectively. In addition, a greater number of mouth fixations occurred in slow speech than in fast speech. The findings of the current study are expected to extend our knowledge on the mechanisms affecting the perception of basic and mixed emotions in spoken messages, which have important social-emotional implications in normal and special populations.

Lexical Stress In Plains Cree: An Acoustic Account

Quinn Goddard

A member of the Algonquin language family, Plains Cree is one of the most widely spoken Cree dialects, with 4,300 speakers across Canada (Statistics Canada, 2019). As of yet, accounts of word-level stress in Plains Cree have come primarily from impressionistic descriptions of the language (Wolfart, 2006). Specifically, the antepenult (of words with three or more syllables) and the final (of disyllabic words) are considered to be primarily stressed. In work that addresses the prosody of the language more formally, mention of the exact dimensions which characterize the purportedly stressed syllables is still neglected (Cook, 1991, and Rosen, 2006). Muehlbauer (2006) describes pitch as defining lexical stress, however he argues that primary stress is instead realized through a pitch fall on the penultimate. In order to evaluate the above models, we analyzed publicly available recordings of Plains Cree to determine the position and acoustic properties of word-level stress. We used the recordings produced for the Canadian Bible Society (2010) featuring Mrs. Dolores Sand (Muskeg Lake Cree Nation). We measured the duration, F0, intensity, and quality of vowels in words with no vowel deletion. Vowels adjacent to glides or with aspiration were also omitted. Both short and long vowels were analysed. The findings indicate that F0 is the strongest cue for stress. In words of three or more syllables, the antepenult has higher F0 than the other syllables, with the final syllable having the lowest F0. In contrast to the level F0 of short vowels, long vowels have a rising F0 on the antepenult. The results corroborate Muehlbauer's finding of a distinct F0 fall on the penult. Long vowels undergo a more dramatic pitch fall, facilitated by a lengthening of long vowels on the penultimate. In contrast, short vowels do not differ in duration across syllable positions.

Deep Learning Based Continuous Vowel Space Mapping From Hand Gestures

Yadong Liu, Pramit Saha, Bryan Gick, Sidney Fels

Converting hand gestures to speech sounds has been proved to be successful in Glove Talk II [Fels and Hinton, IEEE Transactions on Neural Networks, 8(5), (1997), 977]. This work used an adaptive interface including gloves, space trackers and a foot-pedal to map gestures to English speech sounds, with each gesture corresponding to one English segment; with this text-to-speech interface, Glove Talk II had difficulty producing diphthongs with natural transitions. The present study aims to develop a more intuitive and compact, single-handed user interface for vowel space control. This system converts hand movements directly to a continuous formant space to generate English vowels through a formant-based speech synthesizer. We have collected kinematic glove data of two participants using Cyberglove corresponding to wrist movements (up-down) and finger abduction (sideways) for 8 different English vowels as well as diphthongs. We employed a variety of deep neural networks, with varying hyperparameters, mapping the finger and wrist movements to the continuous vowel quadrilateral formant space (F1 and F2) and analysed the performance of these networks. Results demonstrated that our system achieved successful continuous mapping of one hand movements to the formant space, thereby generating English vowels accurately from a variety of hand gestures, and also showed the prospect of producing vowels of other languages.

The Effects Of Transcranial Direct Current Stimulation (Tdcs) On Intermuscular Coherence During Maximum Performance Tasks In Healthy Younger And Older Adults

Kirsten Mulder, Alesha Reed, Benjamin V. Tucker, Carol A. Boliek

The typical ageing process is associated with motor, cognitive, and neuroanatomical declines beginning as early as 50 - 60 years of age. Specifically, declines in speech motor control have been observed in healthy ageing. Transcranial direct current stimulation (tDCS) uses electrodes placed on the scalp to modulate cortical activity by modifying neuronal excitability. This method may effectively modulate speech motor control in healthy ageing adults. We investigated the effects of anodal and cathodal tDCS over Broca's area during high lung-volume non-speech and speech tasks on: speech rate, utterance durations, respiratory kinematics, and intermuscular coherence in chest wall and orofacial muscles. Thirty healthy younger and thirty healthy older adults completed the tasks before and after 13 minutes of randomly assigned anodal, cathodal, or sham (control) 1-mA tDCS. EMG signals from the orbicularis oris, intercostal and oblique muscles were recorded, along with the acoustic signal. Chest wall kinematics were tracked using variable inductance plethysmography. We found that neither anodal or cathodal tDCS had a significant effect on intermuscular coherence, respiratory kinematics, speech rate, or duration. Practice effects were observed in older and younger adults for speech rate and duration. Significant practice effects related to coherence were observed for older participants including changes in firing time of the left and right orbicularis oris muscles and a decrease in intercostal-oblique intermuscular coherence, suggesting that older adults are more likely to utilize adaptive neuromuscular control strategies. The lack of stimulation effect observed is in line with recent literature and may indicate that online stimulation protocols are more successful at modulating behavior than the offline tDCS protocol used here. The coherence practice effect observed in older adults may be linked to increased fatigue and the use of different neuromuscular control strategies as a result.

Dutch Learners Of English Can Identify Reduced Pronunciation Variants In English Running Speech, While Spanish Learners Cannot

Annika Nijveld, Louis Ten Bosch, Mirjam Ernestus

Spanish (N = 69) and Dutch (N = 35) learners of English listened to six stretches of everyday British English speech and transcribed on a computer keyboard what they thought was being said. The stretches were recorded from (unscripted) BBC radio podcasts. While the stretches have a normal articulation rate (5.6 syllables/second on average), they contain numerous reduced pronunciation variants (words produced with fewer or altered segments compared to their citation forms), which are typical of spontaneous speech. The two groups of learners were both at CEFR B2 ('upper intermediate') proficiency level in English. Unlike the Spanish learners, the Dutch learners are familiar with English reduction patterns because these resemble the systematic reduction patterns of Dutch.The Dutch outperformed the Spanish on this task. For instance, they successfully reconstructed [B AH SH G AH V AH M] as British government, while the Spanish provided answers like the discoverational and press coffee here. These answers show that the Spanish learners were unable to identify the reduced pronunciation variants like the Dutch learners did, and that they experienced interference from Spanish L1 phonology (e.g., incorrect perception of English voicing in /b/ and /g/) as well as word segmentation issues.Importantly, how well the learners could identify critical fragments of the stretches could be predicted by their performance in an auditory lexical decision experiment that contained reduced stimulus words presented in isolation. This finding suggests that the Spanish learners' issues understanding reduced pronunciation variants are not primarily due to word segmentation problems. We conclude that Spanish learners face difficulties with reduced pronunciation variants in English running speech for various reasons, including L1 interference, a lack of familiarity with English reduction processes and - to a lesser extent - word segmentation problems.

Reduced Flaps In English Processed By Non-Native Speakers

McKae Pawlenchuk, Yoichi Mukai, Benjamin V Tucker

Reduction is a regular occurrence in casual speech, such as the common pronunciation of "I don't know". This utterance is easily understood by native English speakers, but the perception of reduced variants is typically more difficult for non-native learners (Ernestus, Dikmans & Giezenaar, 2017). The present research examines the time-course of native and non-native (Mandarin) learners' perception of reduced speech. A study was designed to investigate listeners' perception of reduced and unreduced word-medial flaps in English (e.g., "butter" or "ladder"). Experimental items were split between flapped words spelled with a 't' or 'd' (30 for each phoneme) with additional 60 filler items. We use a delayed naming task paired with pupilometry to measure listeners' cognitive load and the time-course of perception while hearing reduced and unreduced stimuli. We compare the results from English native speakers with non-native speakers (Mandarin native speakers). Previous research has shown that reduced forms are harder to process by native speakers (Tucker, 2011) and we expect that this difficulty will be amplified for the non-native listeners' responses. We predict an interaction between pupil dilation and length of residence on the Mandarin speakers' processing of reduced flaps, as well as an interaction between pupil dilation and amount and type of language input on the Mandarin speakers' processing of reduced flaps. In addition, we will investigate the effect of orthography 't' vs. 'd' flaps on the time-course of the native and non-native listeners processing.

Modeling Categorization Of A Vccv Continuum In The Presence Of Response Lapses And Dropouts.

Terrance Nearey, Benjamin V Tucker

In synthetic English /VCCV/ disyllables /arga, alga, arda, alda/, listeners reliably give relatively more /-da/ responses after /ar-/ syllables [e.g., Lotto and Kluender, doi: 0.3758/BF03206049]. Explanations in the literature range from simple auditory contrast to complex phonetic mechanisms. In three experiments in our lab, we synthesized factorially crossed F3 continua for both the liquid /r-l/ and the stop /g-d/, while also recording responses to all disyllables (4 alternative forced-choice). Experiments 1 and 2 gave apparently conflicting results. Both showed a phonological bias: categorizing the liquid as /r/ is associated with an abrupt boost in /d/ response. Experiment 1 also showed effects consistent with a continuous auditory contrast of the F3 values across the stop. Experiment 2 sampled the ambiguous region of the /l-r/ continuum more finely. While the /arda/ phonological bias was confirmed, there was little reliable evidence of any continuous contrast effects in F3. Experiment 3 was larger (n > 60) and had more stimuli (70 compared to fewer than 50 each in the earlier experiments) that subsumed the ranges of the other two. Multinomial logistic analysis suggested very complex behavior that does not fit neatly with any published accounts of the phenomenon. However, the task was difficult: more than a third of the listeners showed 20% or more non-responses. The pooled data also showed evidence of response lapses (or button press errors). If a substantial portion of responses occur when listeners are effectively ignoring the presented stimulus, an adequate model should account for this directly. We present results of a hierarchical Bayesian analysis of Experiment 3 data incorporating recently elaborated methods that can deal with both dropouts and lapses [Bak and Pillow, doi:10.1167/18.12.4]. It is hoped that explicitly modeling such effects will lead to a better understanding of the full set of experimental results.


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An Acoustic Analysis of Cannabis-Intoxicated Speech

Arian Shamei[*1] **and Sonya Bird**[†2]
[1]Department of Linguistics, University of British Columbia
[2]Department of Linguistics, University of Victoria

1 Introduction

Acoustic analysis has proven to be an effective method for discriminating the speech features of countless mental and physiological states such as alcohol intoxication [1], Schizophrenia [2], and heart disease [3]. All of these states result in alterations to the muscle-control underlying speech, which can be detected in the acoustic signal. The goal of this study was to determine if it is similarly possible to detect cannabis intoxication (CI) through acoustic analysis of speech.

2 Method

2.1 Participants

Eight medicinal users of cannabis (4 male, 4 female) aged 21-25 provided voice recordings before and after the consumption of medicinal cannabis. All participants were undergraduate students at the University of Victoria who used Cannabis to treat a variety of ailments. As all participants were medicinal users, all had previous experience consuming cannabis, although the duration and frequency of use varied greatly between participants.

2.2 Procedure and materials

The experiment was conducted utilizing a *pretest (PT) and retest (RT)* format, which involved completing the same task before (PT) and after (RT) cannabis consumption. Participants abstained from cannabis use for twenty-four hours prior to completing the PT. Following consumption, participants waited 30 minutes before completing the RT. In all cases, cannabis was consumed via inhalation, however the exact quantity and potency of cannabis consumed by each participant was not controlled; each participant provided their own cannabis and consumed the amount they normally would in accordance with their prescription and daily routine.

The task itself included three components used to measure the acoustic parameters of interest: a reading passage for measuring pitch trajectory, a list of words contrasting voiced~voiceless stops (e.g. *pit ~ bit*) to measure voice onset time (VOT), and a second list of words contrasting seven sustained English vowels to measure voice quality.

Pitch range and trajectory

Prosodic trajectories - pitch in particular - have been useful in distinguishing alterations to executive functions such as attention and planning in speaker states [2]. It stands to reason pitch trajectory may also distinguish CI. To test this, participants were asked to read the entire passage of *The North Wind and Sun* – a short, emotionally neutral text often used in phonetic research. In each condition (PT and RT), participants provided two readings of the story after familiarizing themselves with the text. Approximately twenty seconds of uninterrupted speech from the second repetition of the passage was selected for analysis, for each participant in each condition, by excluding the first and last stanza of the story. Two participant's data had to be excluded from the pitch trajectory analysis on the grounds that no fluent readings could be obtained in the RT, although PT readings were unaffected.

Intersyllabic and intrasyllabic pitch within the reading passage was measured using a similar methodology to that employed by [2] for their assessment of schizophrenic speech, namely their application of the Prosogram [4] for forensic assessments of speaker state: automatic syllable segmentation was implemented with a sampling rate of 0.005/s and an adaptive glissando threshold of [0.16-0.32/T2, DG = 30, dmin = 0.05], allowing rapid collection of F0 (pitch) data and cross-conditional comparisons of pitch trajectory.

Voice Onset Time

VOT provides an indirect measure of motor coordination, since it is a function of precise articulatory timing between the larynx and articulators in the vocal tract. To track the effect of CI on VOT, participants were asked to produce two repetitions of 20 minimal pairs (80 tokens), contrasting only in voicing of the first stop, eg : *pit ~ bit, dire ~ tire*. VOT was calculated by measuring the time in milliseconds (ms) between the release burst of the plosive and the onset of regular voicing in the following vowel. If the onset of voicing commenced after the release burst, the VOT was recorded as positive, but if voicing commenced before the release burst, the value was recorded as negative. For each speaker, the mean VOT of individual target phonemes - /b, p, t, d/ - was calculated to determine the standard deviation of VOT (σVOT). Phoneme-specific σVOTs were combined to calculate the mean σVOT of voiced phonemes (/b, d/) separately from voiceless phonemes (/p, t/), producing measurements of σVOT [+voice] and [-voice].

Vocal Quality

Measures of vocal quality have been used to generate insight about laryngeal physiological states such as muscle tension [3]. The goal of the *sustained vowel production task* (SVPT) was to allow measurements of laryngeal functioning in a non-

*arian@alumni.ubc.ca
†sbird@uvic.ca

speech setting where voluntary control of the larynx can be observed unimpeded. Participants were asked to produce the following words three times while sustaining the vowel for roughly five seconds: *see, saw, sue, sit, so, sat, say*. For each participant, the final two iterations (14 tokens) were selected for analysis in each condition. Jitter and Shimmer (local) were measured automatically with a Praat script, omitting the initial and final 25% of each token.

3 Results

3.1 Pitch range and trajectories

Results of the F0 analysis as described in 2.2.1 are provided in Table 1 below. All measurements are provided in semi-tones. Total pitch range and intrasyllabic trajectory were significantly reduced in RT compared to PT; intersyllabic trajectory and phonation trajectory were not significantly different in PT vs. RT.

Table 1: Pitch range and trajectories in semi-tones

Range		**Intra**		**Inter**		**Phon**	
P=.04		P=.01		P=.21		P=.06	
PT	RT	PT	RT	PT	RT	PT	RT
11.9	9.5	15.2	12.1	18.3	18.9	16.4	14.9

Figure 1 below provides comparative prosograms of the PT (top) and RT (bottom). Note that intersyllabic trajectory is reduced in RT, but intersyllabic fluctuations remained consistent between conditions.

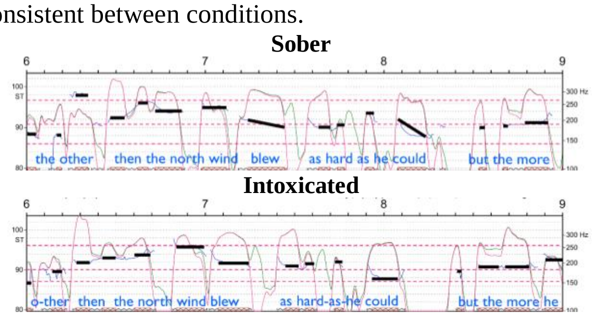

Figure 1: Comparative prosograms of PT (top) & RT (bottom)

3.2 Voice Onset Time

Table 2 below provides the mean σVOT measurements for both voiced (left) and voiceless (right) phonemes. The σVOT for voiced phonemes was significantly in the RT compared to the PT, meaning VOT was more variable in RT. Voiceless phonemes were comparable in the PT and RT.

Table 2: Standard deviation of voice onset time

σVOT [+voice]		**σVOT [-voice]**	
P<.005		P=.12	
PT	RT	PT	RT
20ms	36.2ms	18.1ms	15.9ms

Figure 2 below illustrates prevoicing of onsets observed in the RT (right) but not in the PT (left). English stops are not normally prevoiced, but sometimes were in the RT, resulting higher σVOT [+voice] measurements in the RT condition.

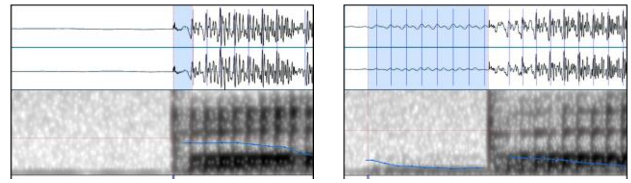

Figure 2: prevoicing in RT token (right) absent in PT(left)

3.3 Vocal Quality

Table 2 below provides the mean measurements of vocal quality for all participants as described in 2.2.2. Shimmer, but not jitter, was significantly lower in the RT to than PT.

Table 3: Jitter and Shimmer

Jitter		**Shimmer**	
P=.18		P=.04	
PT	RT	PT	RT
0.38%	0.34%	3.7%	3.1%

4 Discussion

Analysis of pitch trajectories, VOT, and vocal quality indicate that CI affects speech stream acoustics in significant ways. The range of F0 following intoxication is significantly reduced, and these reductions match previous observations of schizophrenic speech [2], which reflect impairments to executive functions and processing load. Measurements of VOT demonstrate significant impairments to motor timing between the larynx and lips (/b, p/) and tongue (/d, t/). Measurements of vocal quality indicate significant reductions to shimmer which suggest greater flaccidity of the vocal folds and impairments to muscle tone [3]. Future work should investigate the utility of automated acoustic detection of CI.

Acknowledgments

This research was conducted on the unceded and ancestral territory of the WSANEC, Lkwungen, and Wyomilth people. We also wish to thank all the participants, whose volunteerism made this research possible.

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The Effects of Outer Space on Vowel Space

Arian Shamei *[1] **and Bryan Gick** †[1,2]
[1] Department of Linguistics, University of British Columbia.
[2]Haskins Laboratories, New Haven, CT

1 Introduction

After returning from the *Expedition 35* space mission, astronaut Chris Hadfield reported "I could feel the weight of my lips and tongue and had to change how I was talking... I didn't realize I had learned to talk with a weightless tongue." [1]. Such comments indicate that adaptation to changing gravity conditions can have a substantial effect on speech articulation. Previous work has investigated the effects of altered gravity during space travel on speech production [2], and found some effects on vowel space. However, the audio assessed in [2] was from the 1969 Apollo Moon landings, and thus of poor quality due to technological limitations. These limitations resulted in a lack of appropriate data, eg; F1 and F2 were only assessed for one vowel, with only F1 investigated for the remaining vowels. Further, the previous work did not evaluate speech from astronauts immediately before leaving earth, and instead limited itself to speech produced throughout space travel, the surface of the moon, and a news interview after landing. The present analysis makes use of higher quality audio from the more recent STS-134 mission in 2011, and investigates F1 and F2 measurements for five vowels produced by astronaut Mark Kelly immediately before, during, and immediately after the STS-134 mission.

Plotted vowel spaces from each condition suggest substantial reduction of vowel spaces in both the in-space and post-landing conditions. Single factor ANOVAs independently comparing F1 and F2 across all conditions indicate significant differences for the F1 of high vowels, which corresponds to vowel height. These observations corroborate Chris Hadfield's descriptions that altered gravity results in observable impairments to speech motor control.

2 Materials & Methods

The North American Space Association (NASA) provides audio-logs for all missions through the public NASA audio archive (see: https://archive.org/details/nasaaudiocollection). The audio files from the STS-134 mission were selected because they feature substantial speech data from Mark Kelly across various points in the mission.

Approximately 90 seconds of audio from each condition (1. pre-launch interview, 2. in-space interview, 3. post-landing tarmac interview) was assessed using automated alignment and formant extraction via the Dartmouth Linguistic Automation suite (DARLA). Stopwords were omitted from the analysis along with unstressed vowels and tokens where the formant bandwidth exceeded 300Hz. As all samples came from a single speaker, formant normalization was not employed. All vowels for which a comparable number of tokens were obtained (4 or more in each condition) were included in the analysis. For each vowel assessed, F1 and F2 were compared independently using single-factor ANOVAs. Following application of the selection criteria, five vowels were included in the analysis: The low front vowel /ae/, the mid front vowel /ɛ/, the near high front vowel /ɪ/, the high front vowel /i/, and the low-mid back vowel /ʌ/. A 5 level Holm-Bonferroni correction was applied to the ANOVAs for F1 and F2 independently, with a rank 1 significance level set at 0.01.

3 Results

Results from the DARLA formant analysis are provided in Table 1 below. Table 1 provides the mean F1 and F2 for all three conditions (pre-flight, mid-flight, post-flight). The first column designates each vowel and the following columns specify the condition. Results for each condition are found in the rows below, with F1 values on the left, and F2 on the right, The final column provides the p-value as obtained by single-factor ANOVA for the vowel in each row along with the Holm-Bonferroni corrected significance threshold in parenthesis, with the p-value for comparisons of F1 on the left, and F2 on the right.

Table 1 : F1 values for each condition

	Pre-flight		Mid-flight		Post-flight		P-value	
	F1	**F2**	**F1**	**F2**	**F1**	**F2**	**F1**	**F2**
i	380	2132	368	2122	451	2071	<.0001 (.01)	=.49 (.025)
ɪ	447	1818	459	1753	518	1660	=.0002 (.012)	=.06 (.016)
ɛ	544	1765	620	1601	613	1551	= .027 (.025)	=.03 (.01)
æ	662	1626	661	1685	735	1617	= .08 (.05)	=.50 (.05)
ʌ	556	1194	672	1264	650	1348	= .11 (.016)	=.09 (.012)

* arian.shamei@ubc.ca
† gick@ubc.ca

Figure 1 below consists of a vowel plot illustrating the differences in vowel space between each condition. Arrows are used to illustrate the difference in position (height/backness) between the mean pre-flight and post-flight measurements. Pre-flight measurements are represented by a green circle, mid-flight measurements by a red triangle, and post-flight measurements by a blue square. The X axis of the plot displays the inverted range of F2, and the Y axis displays the inverted range of F1.

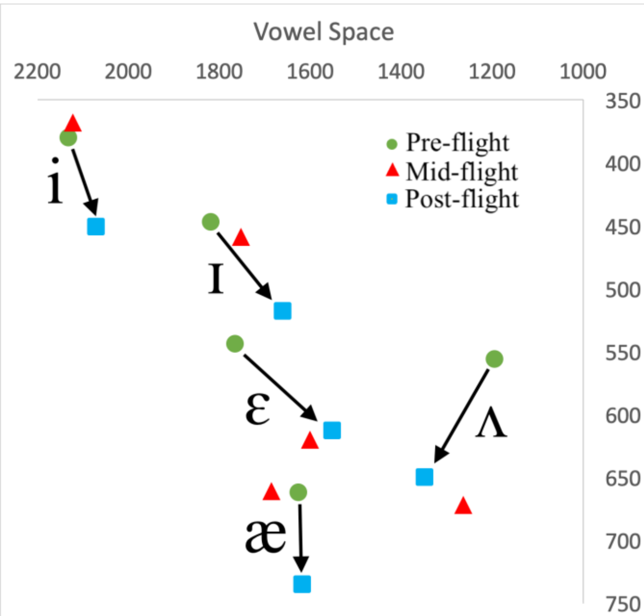

Figure 1: Cross-condition vowel space

4 Discussion

Formant values have been correlated to the height and backness of a speakers tongue, with the value of F1 bearing an inverse relationship with tongue height, and F2 relating to degree of backness [3]. Accepting this, a general effect on the vowel space can be observed between the pre-flight and post-flight conditions, with all vowels lowering and centralizing in the post-flight condition. Mid-flight vowels were variable, although some degree of reduction can be observed relative to the pre-flight condition.

Chris Hadfield's statement that he had grown accustomed to speaking with a weightless tongue in space is reflected in these results. We assume that if someone grew accustomed to a weightless tongue in space, the degree of passive muscle activation employed to keep the tongue elevated during speech would no longer be sufficient upon returning to earth's gravity. With muscle activations insufficient to counteract the forces of gravity, the tongue would be expected to fall short of its usual vowel targets, with the most substantial shortcomings reflected in vowels which are furthest from the resting configuration and require more effort to counteract the forces of gravity.

It is no surprise then that the F1 of the higher vowels (/i/ & /ɪ/), those which require the greatest degree of tongue movement against the forces of gravity, easily surpassed the Holm-Bonferroni corrected significance thresholds. The front mid-vowel /ɛ/, which had a p-value of .027, was very close to the Holm-Bonferroni threshold of .025, however the back mid-vowel /ʌ/ displayed significance (.011) despite the low significance threshold of .016. It is also unsurprising that the lowest vowel, /ae/, was furthest from achieving its unadjusted significance threshold of .05.

Although no measures of F2 produced significance levels below that of the Holm-Bonferroni corrected alpha, a general trend of vowel reduction towards the center can be observed between pre-flight and post-flight conditions, although mid-flight measurements did not show this trend consistently.

Unfortunately, not enough tokens containing the high back vowel /u/ and mid back vowel /o/ were obtained to compare to the high front vowels assessed. Thus a comparison of how backness interacts with vowel height was not possible in this preliminary analysis, but the authors plan to investigate this in future work.

5 Conclusion

The results of the formant analysis demonstrate that adaptation to changing gravity related to space travel may alter speech production. This further corroborates the anecdotal report by Chris Hadfield that astronauts learn to speak with a weightless tongue in space, and must adapt to Earth's gravitational forces upon return. As mechanical alteration to the speech production system is known to affect speech perception as well [4], it is reasonable to assume that astronauts' ability both to send and receive spoken information may be compromised during mission-crucial transitions in gravity.

Acknowledgments

The authors thank Carla Hudson Kam for alerting us to the Chris Hadfield comments that spurred this study. Work funded by NSERC RGPIN-2015-05099 to the second author.

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EEG-to-F0: Establishing Artificial Neuro-Muscular Pathway For Kinematics-Based Fundamental Frequency Control

Himanshu Goyal[*1,3], **Pramit Saha**[†2], **Bryan Gick**[‡3], **and Sidney Fels**[§2]
[1]Department of Computer Science, University of British Columbia
[2]Department of Electrical and Computer Engineering, University of British Columbia
[3]Department of Linguistics, University of British Columbia

1 Introduction

Speech-related Brain Computer Interfaces (BCI) are primarily targeted at finding alternative vocal communication pathways for people with speaking disabilities. However, most of the works in this field are centered around recognition of words from imagined speech, given a particular, small vocabulary [1–3]. Lesser efforts have been invested towards investigating voice synthesis from the information decoded from EEG signals [4, 5]. In this work, as an extension of the previous work [6], we aim at controlling the fundamental frequency of voice based on information derived from brain signals for an elbow movement task, through a biomechanical simulation toolkit ArtiSynth [6, 7]. To the best of our knowledge, this is the first attempt targeting such a brain-to-vocal frequency mapping incorporating a biomechanical control pathway.

2 Overview of proposed technique

The fundamental frequency (F0) of human voice is generally controlled by changing the vocal fold parameters (including tension, length and mass), which in turn is manipulated by the muscle exciters, activated by neural synergies. In order to begin investigating the neuromuscular to F0 control pathway, we simulate a simple biomechanical arm prototype (instead of an artificial vocal tract) that tends to control F0 of an artificial sound synthesiser based on the elbow movements. The intended arm movements are decoded from the EEG signal inputs (collected simultaneously with the kinematic hand data of the participant) through a combined machine learning and biomechanical modeling strategy.

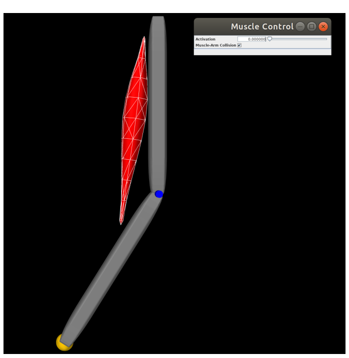


Figure 1: Simulated single-muscle arm model in ArtiSynth

*h.goyal@alumni.ubc.ca
†pramit@ece.ubc.ca
‡gick@mail.ubc.ca
§ssfels@ece.ubc.ca

3 Data collection and preparation

We record EEG data as shown in Fig. 2, corresponding to hand movements using a 10 electrode dry EEG headset with bluetooth connectivity, manufactured by Avertus. It has a sampling frequency of 1000 Hz per electrode at 20 bits/sample. We take data from the electrodes FP1, FP2, F7, F8, T3, T4, T5, T6, O1 and O2 with FCz as reference and FPz as the ground electrode. We particularly choose this headset because of the ease of use and portability of the equipment, which makes our strategy practically implementable. We collect 500 EEG data samples from single participant: 250 each for the hand going up and down, both with the eyes closed and the head fixed. We do not perform any pre-processing on the data and split the data into train:test as 70:30. We also simultaneously collect the angular displacement values or the elbow angles correspondingly with the EEG.

Figure 2: Sample of 10 channel EEG data collected (0.001 s)

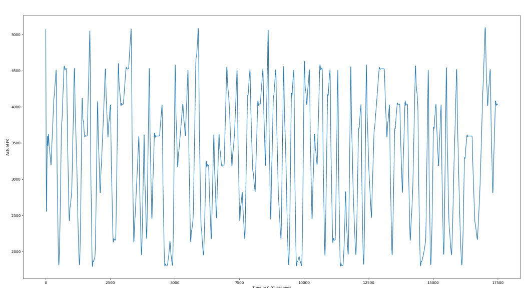
4 Proposed methodology

In order to achieve this EEG to F0 mapping, we first attempt to map the EEG signals corresponding to the elbow movements to the discretized muscle activation of the simulated one-muscle arm model (in the biomechanical toolkit ArtiSynth as shown in Fig. 1) that would create same elbow angular displacement. The temporal resolution of the muscle activations in ArtiSynth is .01 second. Hence, we set a time-window of 0.01 second for EEG signal as well, thereby, each EEG sample (to be mapped to muscle activation) becomes a 10 x 10 matrix . Each matrix is fed to a random forest based classifier with min sample leaf = 1 and min sample split = 2, number of estimators = 10 and random state = 42. This is a 10-class classification task, with the output classes being {0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7, 0.8, 0.9 and 1} which represent the discretized muscle activations between 0 and 1. Using inverse biomechanical modeling in ArtiSynth, we calculate the required muscle activation corresponding to targeted angular movement and optimise the loss function between the intended muscle activation (from EEG signal) and simulated muscle activation (from ArtiSynth) to achieve the best possible mapping from EEG to the muscle activation.

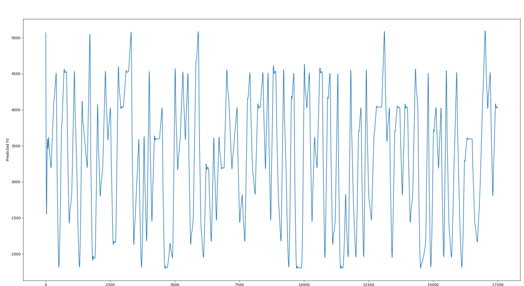
This muscle activation is loaded into the one-muscle biomechanical model of the hand in the ArtiSynth. The resultant angular movement is recorded and the elbow joint angle is then linearly mapped to F0 between 1500 Hz and 5150 Hz. We use the F0 value mapped from the actual kinematic hand data as the ground truth and compare the F0 estimated from brain signal to evaluate the performance of our method.

5 Results

In this section, we present our step-wise performance for the test data. In the first step, *i.e.*, mapping EEG to muscle activations, the overall mean accuracy (the percentage of times, the predicted value matches the actual value) achieved is 85%, with a root mean square error (rmse) of 0.04. After the next step, *i.e.* mapping muscle activations to angles, the cumulative mean accuracy is 68.8% with rmse of 3.4 degrees. The final rmse of the F0 estimation (ranging from 1.5 KHz to 5.15 KHz) is 102.7 Hz. For qualitative comparison of the final predicted F0 with actual F0, a randomly chosen sample is shown in Fig. 3.



(a) Actual F0 values



(b) Predicted F0 values

Figure 3: Comparison of actual and predicted (from EEG) fundamental frequency values

6 Conclusion

We report a novel successful mapping scheme from EEG to fundamental frequency, ranging from 1.5 KHz to 5.15 KHz, with a negligible rmse of 102.7 Hz. This is an initial prototype which proved to be satisfactory in testing the idea of EEG-to-muscle activation-to-kinematics-to-acoustics mapping. Instead of using the complex biomechanical space of the vocal folds, as an initial work in this direction, we tested the concept using simple elbow rotation and single-muscle arm model to simulate the movement, where upwards (or downwards) elbow motion represented increase (or decrease) in frequency. Further work will be directed towards incorporating the ideas using multiple muscle biomechanical arm model, extending it to a vocal fold model as well as replacing the random forest-based classifier with neural network models to investigate any increase in performance accuracy.

Acknowledgments

This work was funded by the Natural Sciences and Engineering Research Council (NSERC) of Canada and Canadian Institutes for Health Research (CIHR).

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MEASURING THE DISPERSION OF DENSITY IN HEAD AND NECK CANCER PATIENTS' VOWEL SPACES: THE VOWEL DISPERSION INDEX

Matthew C. Kelley*[1] and Daniel Aalto†[2,3]
[1]Department of Linguistics, University of Alberta
[2]Department of Communication Sciences and Disorders, University of Alberta
[3]Institute for Reconstructive Sciences in Medicine, Misericordia Hospital

1 Introduction

Head and neck cancer patients who undergo surgical treatment present a variety of physiological and anatomical changes [1–3]. Because of these changes, the patients' speech is impacted. To create intelligible speech, these patients must adopt new communication strategies. The present paper introduces a measure of vowel dispersion that can be used in analyzing patient communication strategies. The measure is termed the "vowel dispersion index," and it is an extension of Story & Bunton's work on vowel space density [4].

Changes in patients' speech may be reflected in their F1-by-F2 vowel space. Per Lindblom's H&H theory [5], hyperarticulated speech will use formant configurations that tend toward the extremes of the vowel space. Lindblom has described this style of speech as more effortful, but also easier for a listener to interpret. On the other hand, hypoarticulated speech uses formant configurations that tend toward the center of the vowel space [6]. Following cancer treatment, the head and neck cancer survivors may need to enhance acoustic contrastiveness in their speech. It is reasonable to expect that they will produce speech closer to the extremes of the available vowel space than the central region if they are to be understood. Indeed, it has been found that the area of patients' vowel space is reduced after surgery with partial restoration after rehabilitation [7].

Working with vowel space requires choosing a representation that can be reasoned about. One such representation is the convex hull, which Story & Bunton use to examine vowel space density [4]. In their work, vowel space density is a reflection of what proportion of time a speaker spends in a given region of the vowel space for running speech. The vowel dispersion index is an extension based off Story & Bunton's work on vowel space density. It provides an indication of how diffuse a patient's vowel productions are.

The remainder of the present paper is as follows. First, the method of calculating the vowel dispersion index is briefly described. Next, several illustrative examples of the vowel dispersion index and the vowel space density for a patient are presented. Finally, the paper concludes with a discussion of the patients' results and potential applications of the vowel dispersion index.

2 Method

To calculate the vowel dispersion index, the vowel space density from Story & Bunton is calculated as in [4]. First, the formant tracks are calculated from a recording. Second, at each point in a grid of possible formant values, the number of points within a given radius is counted, giving an indication of the density in that section. Third, the density values are scaled to the invterval between 0 and 1. And, fourth, a convex hull is calculated around the points that are at a value of 0.25 or higher.

Once the vowel space density has been determined, the vowel dispersion index can be calculated. To find it, the magnitudes of all the discrete gradient values are summed over the vowel space. The result is what's generally known as the total variation, and here is referred to as the vowel dispersion index. When considering the vowel space density as a topographic map, where dense areas represent high elevations and sparse areas represent low elevations, the vowel dispersion index indicates how hilly the area is.

A high vowel density index indicates that there are many elevation changes, and it represents diffuse vowel productions. In turn, diffuse vowel productions show a more diverse use of the vowel space, indicating that the speaker is hyperarticulating. A low value indicates that there are few elevation changes, and the vowel produces are clustered together. In such a case, a speaker is using little of the vowel space and could be said to be hypoarticulating.

3 Results

The vowel space density, vowel dispersion index, and vowel space area are presented here for two patients. A waiver of consent was obtained to use their data and the protocol was approved by the Health Research Ethics Board of Canada Cancer Committee (HREBA.CC-18-0689). At different stages in their treatment, they recorded the zoo passage, which is the recording that was analyzed for the present study. The patients were recorded before their treatment, 1 month post-treatment, 6 months post-treatment, and 1 year post-treatment. The values for the vowel dispersion index and the vowel space area were calculated in a normalized space, but the images are presented in the Hertz scale for ease of interpretation.

A female patient's speech may be seen in Figure 1. Note the single large peak in the upper-right quadrant of the vowel space pre-treatment, indicating a locus of where the majority of the vowel productions occurred. There are also singular large peaks in the space at 1 and 6 months post-treatment, suggesting the patient is using similar speaking strategies to their pre-operation strategies at 1 and 6 months after treatment. However, at 1 year after their operation, they show a

*mckelley@ualberta.ca
†aalto@ualberta.ca

visually greater degree of dispersion. The vowel dispersion index is also higher at the 1 year mark, suggesting that the patient is using more of the overall vowel space. Such a strategy may indicate that the patient is heightening the acoustic contrastiveness of the segments in their speech to compensate for their altered anatomy and physiology.

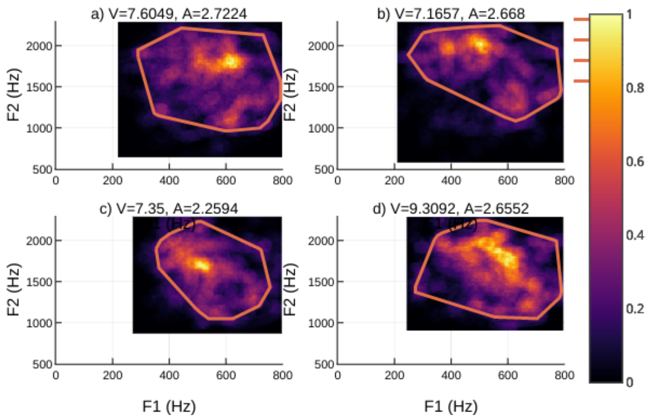

Figure 1: A female patient's vowel space densities. Black is low density, while yellow is high density. a) Pre-treatment. b) 1 month post-treatment. c) 6 months post treatment. d) 1 year post-treatment.

A male patient's speech may be seen in Figure 2. The pre-operation, 1 month post operation, and 1 year post operation recordings are visually similar, with single masses of density surrounded by lower density regions. The mass in the 1 month post operation recording is smaller, however, permitting more variation in the rest of the space, which is reflected numerically in the vowel dispersion index. The 6 months post operation recording stands out as showing greater dispersion, which is also reflected in the vowel dispersion index.

Note from these two examples how the vowel dispersion index and vowel space area are related to each other. While some degree of correlation is expected between the two measures, they are conceptually distinct. The vowel dispersion index offers an additional dimension to vowel space size in describing speech production acoustics.

4 Discussion & Conclusion

The present paper introduced the vowel dispersion index. It indicates the degree to which a speaker's vowel productions are diffuse in the vowel space throughout running speech. For head and neck cancer patients, it can be used to study communication strategies they employ in compensation for anatomical changes after their treatment. It is another tool for researchers in objectively studying the speech of these patients.

But, the vowel dispersion index is not limited to the study of head and neck cancer patients' speech. It could be used to study reduction patterns in speech, like those discussed in [5] and [6]. Additionally, it could be used to investigate the degree of acoustic contrast a speaker is employing in challenging situations, such as speech in noise, speech with objects in the mouth, or speech where a part of the mouth is numbed.

Future work should focus on validating the vowel dispersion index against other known variables, such as intelligibility for head and neck cancer patients and vowel centralization for reduction studies. Other forms of representing the vowel space density could be studied as well, such as Gaussian mixtures. The vowel dispersion index is a quantitative tool for speech science.

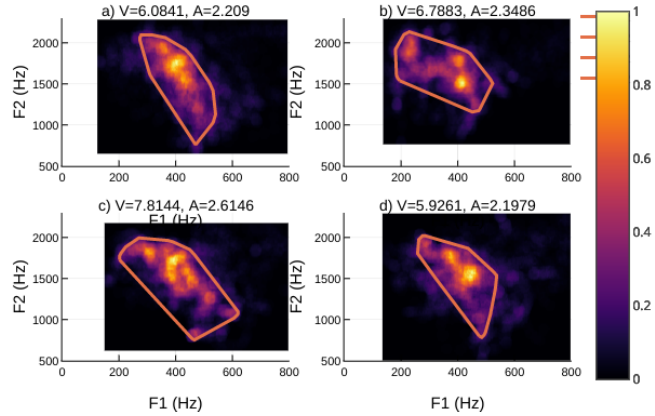

Figure 2: A male patient's vowel space densities. Black is low density, while yellow is high density. a) Pre-treatment. b) 1 month post-treatment. c) 6 months post treatment. d) 1 year post-treatment.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Investigation Of Tongue Biomechanics In English Vowels (/i/,/æ/) Using Virtual Computer Simulation

Noor Hisham Al-Zanoon, Daniel Aalto

The tongue plays a central role in speech production, and this has prompted a vast amount of research on its biomechanics. However, understanding biomechanics underlying tongue movement is inherently difficult due to its speed, and location in the mouth. Previous studies have used various experimental techniques (e.g., imaging and kinematic tracking) to estimate relationships between speech productions and tongue biomechanics. When combined with experimental data, virtual 3D computer simulations of the tongue have the potential to provide critical and experimentally difficult to observe information on tongue biomechanics. ArtiSynth is an open-source simulation platform that uniquely supports the combination of multibody and finite element (FEM) models. ArtiSynth provides a generic jaw-tongue model that fully couples the jaw and hyoid (rigid bodies) with an FEM tongue model. Tongue motion trajectories are achieved through specific muscle activation patterns. To determine the required muscle activation patterns, we propose the use of tongue position data collected from electromagnetic articulography (EMA). We have collected data from a single subject producing two English vowels embedded in word targets (/bit/, /bæt/). Sensors were placed on the tongue tip, dorsum, right and left lateral positions. Collected motion trajectories include tongue position from rest to the target words. The tongue position in the jaw tongue model will be modulated based on the EMA data. Markers will be virtually placed on the tongue model to match the sensor placement in the EMA. We will then use inverse simulation to determine the required excitation patterns to produce the tongue trajectories.To evaluate the performance of the virtual tongue mode, resynthesized trajectories will be compared to the kinematic tracking . Overall, our aim is to develop a workflow for translating experimental data to virtual computer simulation. This workflow is critical for unlocking the potential of virtual computer models as an investigative tool in tongue biomechanics.

Coarticulation In Speech And Smile Movements: An Electromyographic And Video Analysis

Terrina W Chan, Grace Purnomo, Yadong Liu, Bryan Gick

Temporally overlapping facial expressions and speech movements impose conflicting demands on articulators. For example, lip spreading associated with smiling is incompatible with bilabial closures such as /m/, /b/, or /p/. Anecdotal evidence suggests this conflict may at least sometimes resolve as a labiodental stop variant. The simplest model of coarticulation - one of unmediated superposition [Gick et al., 2013, POMA 060207] - suggests that the outcome of this conflict should be determined by summing overlapping muscle activations. In such a model, different degrees of activation for smile and stop closure force should be expected to yield distinct outputs. Previous research suggests that closures for /m/, /b/, and /p/ vary increasingly in intraoral pressure [Lubker & Parris 1970, JASA 47: 625] and therefore muscle force [Gick et al. 2012, JASA 131: 3345] which are variables that may influence how articulatory conflict is resolved. An alternative proposal to an unmediated superposition model of coarticulation suggests active inhibition, wherein one activation set is deliberately inhibited to favour a particular outcome. This paper describes a study of a particular case of coarticulatory conflict – specifically, bilabial production during smiled speech — to examine the plausibility of the two models of coarticulation. We present video and electromyography (EMG) data gathered from twelve subjects tasked to read aloud 31 sentences containing one bilabial consonant, under three facial conditions (neutral, smiling, and laughing). EMG electrodes were placed on the orbicularis oris and zygomaticus major to measure degrees of muscle activation. Video analysis compares visible degree of dentalization, while EMG analysis compares muscle voltage between the orbicularis oris and the zygomaticus major to detect the presence of inhibition during this articulatory conflict.

An Acoustic Analysis Of Vowel Production In Spontaneous Speech

Michael Kiefte, Terrance M. Nearey

A large database of speech samples from the Canadian Maritimes was collected, processed, and analyzed with the primary aim of examining vowel-inherent spectral change in vowel formants. The analysis of spontaneous speech presents a number of challenges when compared to citation forms. For example, surrounding consonants have a large influence on vowel formant frequencies. To overcome this problem, a statistical procedure was developed to estimate onset and coda consonant effects on vowel formant frequencies. Estimates of these vowel-target formant frequencies were then determined after factoring out consonant-context effects. Formant-frequency values were estimated for all stressed vowels in a database of thirty-five hours of recorded speech from 223 speakers. This talk describes the procedure developed for this analysis as well as the results. In addition, comparisons will be made between citation and spontaneous speech as well as between different dialect areas within the region.

Canadian Prairie Dialects: An Exploration Of Alberta And Saskatchewan Vowels

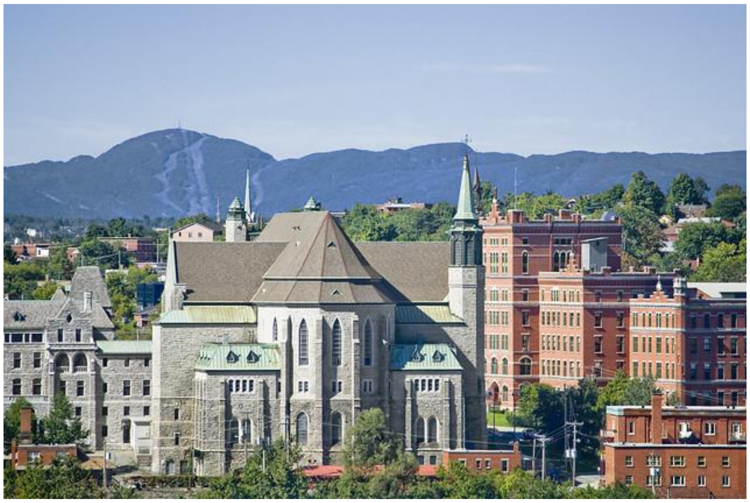
Bryce Jacob Wittrock, Benjamin Tucker

Previous research has documented the English spoken in Western Canada as generally homogenous with minor variation along a spectrum from British Columbia to Northern/Western Ontario (Labov et al., 2005). This description is investigated on a narrower geographic scale, building a corpus of Southern Alberta and Saskatchewan English (SASE) and comparing the acoustic properties of SASE vowels to those of Edmonton English (EE) as observed in Thomson 2007. The corpus consists of 24 informal interviews as well as word list and passage readings. First and second formants were extracted from the word list (careful speech) context and measurable acoustic differences were found between this data and the EE description. Both speaker groups were found to have similar distributions, however /?/ and /?/ as well as /?/ and /?/ are closer to each other in SASE than in EE, suggesting possible mergers. SASE fronts /u/ and /?/, aligning with shifts noted in the western and southern United States (Clopper, 2005) but also fronts /?/, distancing itself from the Northern Cities Shift which instead describes a backing of /?/ (Labov, 1998). We also statistically explore differences in the vowel space using vowel overlap methods (Kelley et al., in prep).

Announcement

ACOUSTICS WEEK IN CANADA

Sherbrooke (Québec) October 7-9, 2020



View of Mont-Orford from downtown Sherbrooke

Acoustics Week in Canada 2020 will be held on October 7-9, in Sherbrooke, Québec.

You are invited to be part of this three-day conference featuring the latest developments in Canadian acoustics and vibration. Sherbrooke is well known in acoustics for the Groupe d'Acoustique de l'Université de Sherbrooke (GAUS) founded in 1984.

The conference will be an excellent opportunity to visit or rediscover the GAUS during the International Year of Sound!

The keynote talks and technical sessions will be framed by a welcome reception, conference banquet, Acoustical Standards Committee meeting, technical tour and an exhibition of products and services related to the field of acoustics and vibration.

Take a few days before or after the conference to enjoy the area and the cultural activities! Especially have a look to the beautiful surrounding nature during Fall colors with Mont-Bellevue downtown and the nearby 'Mont-Orford' National Park. Three other parks can also be found within a radius of 100 km.

Various demos and activities will be held at the Groupe d'Acoustique de l'Université de Sherbrooke (GAUS) and at Université de Sherbrooke campus - A series of innovative workshop activities will be a part of the program; we are open to proposals along this line (challenges, measurements, simulations).

Venue and Accommodation

The conference will be held at the Hotel Delta by Marriott in Sherbrooke. A block of rooms in the hotel will be available at a special rate of 155$/night. This rate is extended to stays two days prior and two days after the conference, and each room can be shared across up to 4 people. Complimentary city bus passes will be offered to all the participants to promote the use of public transport during the conference. A shuttle is also available to provide a direct link between International Montréal Trudeau Airport and the conference venue. Please refer to the conference website for further details and registration: https://awc.caa-aca.ca/index.php/AWC/AWC20

Plenary, Technical and Workshop Sessions

Plenary, technical and workshop sessions are planned throughout the conference. Each day will begin with a keynote talk of broader interest and relevance to the acoustics community. Technical sessions are planned to cover all areas of acoustics including:

AEROACOUSTICS / ARCHITECTURAL AND BUILDING ACOUSTICS / BIO-ACOUSTICS AND BIOMEDICAL ACOUSTICS / MUSICAL ACOUSTICS / NOISE AND NOISE CONTROL / PHYSICAL ACOUSTICS / PSYCHO- AND PHYSIO-ACOUSTICS / SHOCK AND VIBRATION / SIGNAL PROCESSING / SPEECH SCIENCES AND HEARING SCIENCES / STANDARDS AND GUIDELINES IN ACOUSTICS / ULTRASONICS / UNDERWATER ACOUSTICS

A General Public Session

A general public session is currently planned on the afternoon of the last conference's day and linked to the International Year of Sound, a global initiative to highlight the importance of sound and related sciences and technologies for all in society (https://sound2020.org/). This event will be held on Université de Sherbrooke campus and opened to scholars and to the population. The organizing committee welcomes any proposal for this session, a rare occasion of explaining our everyday job and implications for society.

Exhibition and Sponsorship

The conference offers opportunities for suppliers of products and services to engage the acoustic community through exhibition and sponsorship.

The tabletop exhibition facilitates in-person and hands-on interaction between suppliers and interested individuals. Companies and organizations that are interested in participating in the exhibition should contact the Exhibition and Sponsorship coordinator for an information package. Exhibitors are encouraged to book early for best selection.

Anechoic room and wind-tunnel opening at GAUS

The conference will be offering sponsorship opportunities of various conference features. In addition to the platinum, gold and silver levels, selected technical sessions, social events and coffee breaks will be available for sponsorship. Additional features and benefits of sponsorship can be obtained from the Exhibition and Sponsorship coordinator and on the conference website. Demos can also be organized at GAUS.

Students

Students are strongly encouraged to participate. Students presenting papers will be eligible for one of three 500$ Best Presentation Student prizes to be awarded. Conference travel bursaries will also be available to those students whose papers are accepted for presentation.

For Registration Details,

For registration details please refer to the conference web site: https://awc.caa-aca.ca/index.php/AWC/AWC20

Contacts

Conference Chair:
Olivier Robin
(Olivier.Robin@USherbrooke.ca)

Technical co-Chairs:
Patrice Masson and
Sebastian Ghinet
(Patrice.Masson@USherbrooke.ca)
(Sebastian.Ghinet@nrc-cnrc.gc.ca)

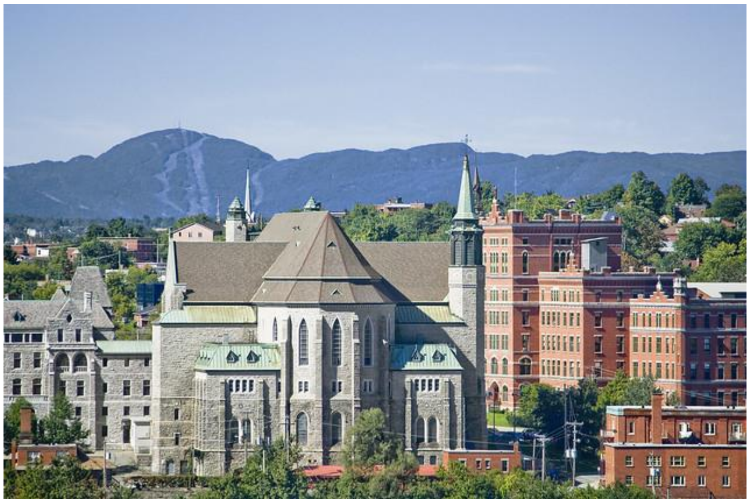
Exhibits and Sponsorships:
Julien Biboud
(Julien.Biboud@mecanum.com)

Enjoy the Mont Bellevue in the center of Sherbrooke during Fall

Appel à communication

SEMAINE CANADIENNE D'ACOUSTIQUE

Sherbrooke (Québec) 7-9 Octobre 2020


Vue du Mont-Orford depuis le centre-ville de Sherbrooke

La Semaine canadienne d'acoustique 2020 se tiendra du 7 au 9 octobre 2020 à Sherbrooke, Québec.

Nous vous invitons à prendre part à cette conférence de trois jours sur les derniers développements en matière d'acoustique et de vibrations au Canada. Sherbrooke est reconnue en acoustique pour le Groupe d'Acoustique de l'Université de Sherbrooke (GAUS) fondé en 1984.

La conférence sera le moment idéal pour visiter ou redécouvrir le GAUS durant l'Année Internationale du Son !

Les exposés principaux et les séances techniques seront encadrés par une réception de bienvenue, un banquet, une réunion du comité des normes acoustiques, une visite technique et une exposition de produits et services liés au domaine de l'acoustique et des vibrations.

Prenez quelques jours avant ou après la conférence pour profiter de la région et des activités culturelles ! Découvrez la nature environnante durant la flambée des couleurs d'automne, avec la proximité du Parc National du Mont-Orford. Trois autres parcs nationaux sont accessibles dans un rayon de 100 km.

Diverses démonstrations et activités seront organisées au sein du Groupe d'Acoustique de l'Université de Sherbrooke (GAUS) et sur le campus principal de l'université de Sherbrooke. Des ateliers participatifs seront intégrés dans le program-me; nous sommes ouverts à toute proposition (concours, mesures, simulations).

Lieu et hébergement

La conférence aura lieu au Centre de congrès de l'Hôtel Delta Sherbrooke par Marriott. Un bloc de chambres dans l'hôtel sera disponible à un tarif spécial de 155$ par nuit (valable deux jours avant et deux jours après la conférence, et chaque chambre peut être partagée par 4 personnes au maximum). Des passes de bus seront offertes à tous les participants afin de favoriser l'usage du transport en commun durant la conférence. Une navette directe entre l'aéroport international Trudeau de Montréal et le lieu de la conférence est également accessible sur demande. Veuillez consulter le site Web de la conférence pour plus de détails et pour l'inscription: http://awc.caa-aca.ca/AWC/AWC20

Séances plénières, techniques et ateliers

Des séances plénières, techniques et des ateliers sont prévus tout au long de la conférence. Chaque journée débutera par une plénière d'un intérêt et d'une pertinence plus larges pour la communauté de l'acoustique. Des sessions techniques sont prévues pour couvrir tous les domaines de l'acoustique, y compris :

AÉROACOUSTIQUE / ACOUSTIQUE DU BÂTIMENT ET ARCHITECTURALE / BIOACOUSTIQUE / ACOUSTIQUE BIOMÉDICALE / ACOUSTIQUE MUSICALE / BRUIT ET CONTRÔLE DU BRUIT / ACOUSTIQUE PHYSIQUE / PSYCHOACOUSTIQUE / CHOCS ET VIBRATION / LINGUISTIQUE / AUDIOLOGIE / ULTRASONS / ACOUSTIQUE SOUS-MARINE / NORMES EN ACOUSTIQUE

Une session grand public

Une session grand public est planifiée en après-midi du dernier jour de la conférence, et liée à l'année internationale du son, une initiative globale destinée à illustrer l'importance du son et de ses sciences et technologies dans la société (https://sound2020.org/). Cet évènement se déroulera sur le campus de l'Université de Sherbrooke et sera ouvert aux scolaires et à la population. Le comité organisateur est ouvert à toute proposition pour cette session, une rare occasion d'expliquer notre travail et ses implications pour la société.

Exposition et parrainage

La conférence offre aux fournisseurs de produits et de services la possibilité de faire participer la communauté acoustique par l'exposition et le parrainage.

L'exposition sur le plateau facilite l'interaction en personne des fournisseurs et des personnes intéressées. Les entreprises et organisations désirant participer à l'exposition doivent contacter le coordonnateur de l'exposition et du parrainage pour obtenir un dossier d'information. Les exposants sont encouragés à réserver tôt pour obtenir de meilleures opportunités.

Salle anéchoïque et soufflerie au GAUS

La conférence offrira des possibilités de parrainage de divers évènements de la conférence. Outre les niveaux platine, or et argent, des séances techniques, des événements sociaux et des pauses café seront disponibles pour le parrainage. Les commanditaires peuvent placer leur logo sur le site Web de la conférence dans les 10 jours suivant leur parrainage. Les caractéristiques et avantages supplémentaires du parrainage peuvent être obtenus auprès du coordonnateur des expositions et des commandites ou sur le site Web de la conférence. Des démonstrations pourront aussi être organisées au GAUS.

Les étudiants

Les étudiants sont fortement encouragés à participer. Les étudiants qui présenteront seront admissibles à l'un des trois prix de 500 $ pour les meilleures présentations. Des subventions de voyage seront également offertes aux étudiants dont les communications sont acceptées pour présentation.

Plus d'informations

Pour plus d'information sur l'inscription veuillez consulter le site Web de la conférence : http://awc.caa-aca.ca/AWC/AWC20.

Contacts

Président de la conférence :
Olivier Robin
(Olivier.Robin@USherbrooke.ca)

Présidents techniques :
Patrice Masson and
Sebastian Ghinet
(Patrice.Masson@USherbrooke.ca)
(Sebastian.Ghinet@nrc-cnrc.gc.ca)

Exposants et commandites :
Julien Biboud
(Julien.Biboud@mecanum.com)

Appréciez le Mont Bellevue au centre de Sherbrooke durant l'automne

CANADIAN ACOUSTICS TELEGRAM ANNOUNCEMENTS - ANNONCES TÉLÉGRAPHIQUES DE L'ACOUSTIQUE CANADIENNE

Looking for a job in Acoustics?

There are many job offers listed on the website of the Canadian Acoustical Association!

You can see them online, under http://www.caa-aca.ca/jobs/

August 5th 2015

International Symposium on Room Acoustics - ISRA2019

The "International Symposium on Room Acoustics" in Amsterdam. ISRA 2019 is a satellite symposium to the ICA conference that takes place September 15 – 17 2019.

- You are cordially invited to participate in the conference and to submit your contributions. The agenda features interesting structured sessions but all submissions on the topic of room acoustics are very much appreciated. The existing session cover the following topics: 1- Developments in prediction techniques 2- Experimental methods in room acoustics 3- Music rehearsal rooms and stage acoustics 4- Performer's adaptation to room acoustics 5- Room acoustic simulations as a tool for performance-based design 6- Room acoustic perception 7- Curved Architecture in Acoustics 8- Design fundamentals and strategies for concert halls and large auditoria 9- Metrics vs. Quality – What are we missing? - It goes without saying that distinguished keynote lectures and a concert in Concertgebouw Amsterdam are a central part of the conference's agenda. - You can find detailed information at www.isra2019.eu.

February 15th 2019

Acoustics Week in Canada 2019

We are pleased to announce that the 2019 Acoustics Week in Canada meeting will be in Edmonton, Alberta on October 9-11, 2019.

- We are pleased to announce that the 2019 Acoustics Week in Canada meeting will be in Edmonton, Alberta on October 9-11, 2019. Abstract submission is now open with abstracts due on June 14, 2019. We are also pleased to announce our three plenary speakers: Hildegard Westerkamp (Soundscape Composition and Acoustic Ecology), Sonya Bird (Speech Acoustics and Indigenous Languages), and Michelle Vigeant (Acoustics & Architectural Engineering). Please find the call for papers here https://awc.caa-aca.ca/index.php/AWC/index/manager/files/AWC19/Call_for_papers_Combined.pdf. More information can be found online at https://awc.caa-aca.ca/. We look forward to seeing you in Edmonton!

April 1st 2019

Acoustics Week in Canada 2020

AWC 2020 will be held October 7 – 9, 2020 in Sherbrooke (Québec) with Dr. Olivier Robin as General Chair, as well as Prof. Patrice Masson and Dr. Sebastian Ginet as Scientific Chairs. https://awc.caa-aca.ca/index.php/AWC/AWC20

May 3rd 2019

Acoustics Week in Canada 2021

AWC 2021 will be held in St-John's (Newfoundland) with Profs. Benjamin Zendel and Len Zedel as co-chairs. https://awc.caa-aca.ca/index.php/AWC/AWC21

May 3rd 2019

2020: International Year of Sound

The International Year of Sound (IYS 2020) is a global initiative to highlight the importance of sound in all aspects of life on earth and will lead towards an understanding of sound-related issues at the national and international level.

- Inspired by the achievements of La Semaine du Son (The Week of Sound), and following naturally as an important contribution to UNESCO Resolution 39 C/49 25 September 2017 on "The Importance of Sound in Today's World: Promoting Best Practices", the International Commission for Acoustics (ICA) is mobilizing its Member Societies and International Affiliates to promote best practices in sound during the year of 2020 to create an International Year of Sound (IYS 2020). For more info, visit http://sound2020.org/ -

May 3rd 2019

ASTC Workshop by the National Research Council Canada

The National Research Council Canada will be hosting a workshop to give an overview of new tools to calculate the ASTC.

This workshop will be held during the 2019 Acoustics Week in Canada meeting in Edmonton, Alberta on October 9-11, 2019. - More info available from https://awc.caa-aca.ca/public/conferences/2/AWC19/ASTC_workshop_flyer.pdf

June 11th 2019

À la recherche d'un emploi en acoustique ?

De nombreuses offre d'emploi sont affichées sur le site de l'Association canadienne d'acoustique !

Vous pouvez les consulter en ligne à l'adresse http://www.caa-aca.ca/jobs/

August 5th 2015

International Symposium on Room Acoustics (ISRA 2019)

Le « Colloque International sur l'acoustique des salles» (plus connu sous le nom de « International Symposium on Room Acoustics » ou encore ISRA) qui aura lieu à Amsterdam du 15 au 17 septembre 2019 à la suite de la réunion de l'ICA (International Congress on Acoustics).

Vous êtes cordialement invités à nous soumettre vos contributions pour cette conférence. Des réunions intéressantes et structurées sont déjà au programme, mais tous les apports au thème de l'acoustique des salles sont les bienvenus. Les réunions couvrent les thèmes suivants : 1- Developments in prediction techniques 2- Experimental methods in room acoustics 3- Music rehearsal rooms and stage acoustics 4- Performer's adaptation to room acoustics 5- Room acoustic simulations as a tool for performance-based design 6- Room acoustic perception 7- Curved Architecture in Acoustics 8- Design fundamentals and strategies for concert halls and large auditoria 9- Metrics vs. Quality – What are we missing? - Le programme inclut aussi plusieurs allocations majeures sur le thème de l'acoustique, et offre la possibilité d'assister à un concert dans le hall principal du Concertgebouw. - Vous pouvez retrouver des informations détaillées sous le lien suivant : http://www.isra2019.eu

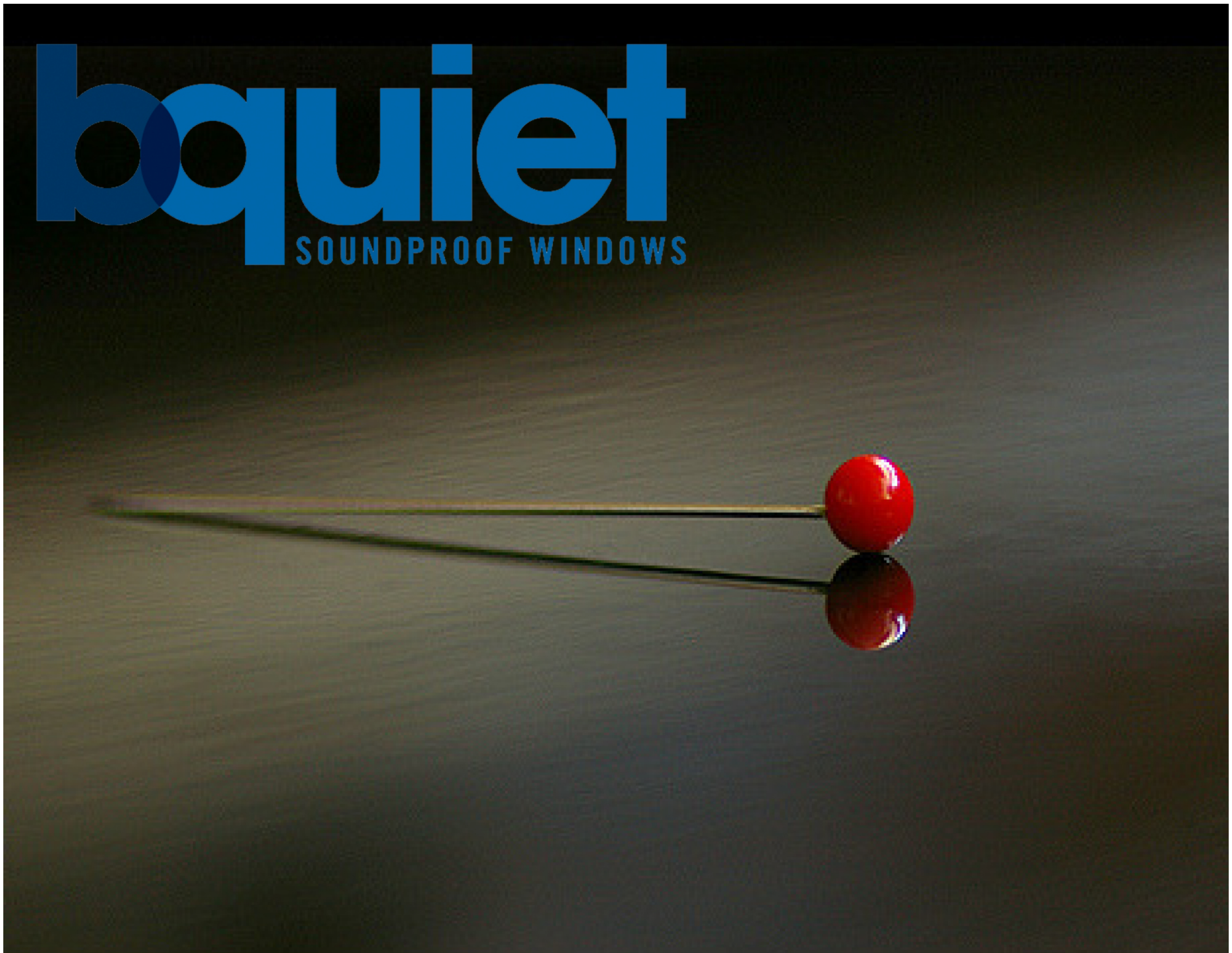
February 15th 2019

Semaine canadienne de l'acoustique 2019

La Semaine canadienne de l'acoustique 2019 se tiendra à Edmonton, en Alberta, du 9 au 11 octobre 2019.

Nous sommes heureux d'annoncer que la conférence de la Semaine canadienne de l'acoustique 2019 se tiendra à Edmonton, en Alberta, du 9 au 11 octobre 2019. La soumission des résumés est ouverte et les résumés doivent nous parvenir le 14 juin 2019. Nous avons également le plaisir d'annoncer nos trois conférenciers pléniers: Hildegard Westerkamp (Soundscape Composition and Acoustic Ecology), Sonya Bird (Speech Acoustics and Indigenous Languages), et Michelle Vigeant (Acoustics & Architectural Engineering). Veuillez trouver l'appel à contributions ici https://awc.caa-aca.ca/index.php/AWC/index/manager/files/AWC19/Call_for_papers_Combined.pdf. Plus d'informations peuvent être trouvées en ligne à https://awc.caa-aca.ca/.

April 1st 2019


bquiet
SOUNDPROOF WINDOWS

To: ICA Member Societies
and International Affiliates

Dear Colleagues,

As you know, the International Commission for Acoustics has started preparations for the celebration of an International Year of Sound for the 2020 (IYS 2020). The IYS 2020 will not be included in the UNESCO and the UN's official list of International Years but will have a similar format, with events centrally organized by an IYS 2020 Liaison Committee, events organized by the ICA National Societies and International Affiliates and possibly events organized by the Week of Sound (WoS) a French non-governmental organization which had the initiative to approach to convince UNESCO to approve the Resolution 39 C/49 25 September 2017 “THE IMPORTANCE OF SOUND IN TODAY’S WORLD” . The IYS 2020 will make reference to the necessity for promoting best practices in the framework of this resolution and we have already informed UNESCO about that.

The ICA and the WoS will create a Liaison Committee to coordinate the IYS 2020. For the time being the ICA representatives are Michael Taroudakis (President) and Marion Burgess (Past President). In addition, an ICA/IYS 2020 steering committee will be formed to coordinate all the events to be organized by the ICA Member Societies and will have representatives from all the regions.

In order to prepare the events of the IYS 2020 in the most efficient way, we are asking each of the ICA Members and our International Affiliates to appoint one representative to be in direct contact with the ICA/IYS 2020 steering committee. The contact person will have the responsibility to communicate all the planned events by his/her organization to the ICA/IYS 2020 steering committee and also to convey and discuss with them any ideas or suggestions about the events and activities to be included in the IYS 2020.

The attached file describes the main idea of the events to be included in the IYS 2020.

Please send the name and contact details of your representative to the ICA Secretary General Mike Stinson (ICASecGen@icacommission.org). If you have any questions or comments, please contact me (ICAPresident@icacommission.org).

I hope that with your collaboration, the importance of sound in today's world will reach every part of our planet in 2020.

With my best regards

Michael Taroudakis
President of the ICA

INTERNATIONAL YEAR OF SOUND 2020

National/International Coordinators

Education and Outreach on Sound for Society and the World

Culture – Creativity – Nature – Health – Science – Technology

Development – Education – History

Outcome of the UNESCO Charter of Sound

MISSION

The International Commission for Acoustics has decided to declare the Year 2020 as the International Year of Sound (IYS 2020). The IYS 2020 will not take the form of an official International year sponsored by UNESCO and the UN, but will have a structure similar to such an official International Year with many events to be organized centrally by the IYS 2020 steering committee or regionally by the ICA members Societies and the Week of Sound, which will be a partner to ICA in his celebration.

The International Year of Sound follows naturally as an important contribution to the UNESCO Charter of Sound. The year will be a global initiative to highlight the importance of sound in all aspects of life on earth and will work towards an understanding of sound-related issues at the international level.

ACTIVITIES/EVENTS IYS 2020

These will fall into three main categories:

- Centrally organized broad area events/outcomes funded by ICA and sponsors.
- Those organized and funded by ICA Member societies and organisations.
- Those organized by the Week of Sound (WoS) funded in the normal manner by the WoS (also referred to as La Semaine du Son).

LIAISON AND STEERING COMMITTEES

The ICA and the WoS will create a Liaison Committee to coordinate the IYS 2020. For the time being the ICA representatives are Michael Taroudakis (President) and Marion Burgess (Past President).

Also, an ICA IYS 2020 Steering Committee will be formed to coordinate all the activities of the IYS 2020, which will include events managed by ICA Member Societies and supporting Organisations. Members of this Committee will be representatives from all the ICA Regions.

STRUCTURE FOR ICA MEMBER ORGANISATIONS ACTIVITIES/EVENTS

Each member organisation/society is asked to nominate a coordinator to be the primary contact with the ICA IYS steering committee

The coordinator will discuss with the organization/society the events/activities that can be undertaken during 2020 and will help to promote one or more aspects of acoustics.

Once the activities are decided upon, the coordinator will provide a concise summary plus dates to the IYS steering committee for endorsement as an official IYS activity.

Each activity endorsed will have the authority to use the IYS 2020 logo and be included in the official IYS 2020 website calendar and other promotion.

All funding for the event/activity must be provided for the activity by the member society or organization and no central funding will be provided

The steering committee will provide some promotional material to the coordinator. The steering committee will also promote the activity internationally as appropriate.

At the completion of the activity, the coordinator will be responsible for providing a concise report plus photos and links to supplementary material. This will be loaded onto the IYS 2020 website as a future resource.

TYPES OF ICA MEMBER ORGANISATIONS ACTIVITIES/EVENTS

All activities that relate to the mission of the IYS 2020 would be relevant. While commercial sponsorship is encouraged, and hence there would be some advertisement, the coordinator is responsible to ensure that the activity is not solely aimed to promote the company or particular products.

This IYS 2020 is the opportunity to promote to the world the importance of sound to all aspects of our life. Organisations are encouraged to consider outreach activities and to be innovative.

The following are some suggestions but it is up to each organisation to consider what may be appropriate for their region/resources

Activities related to the annual meeting or conference.

Activities related to relevant "days" throughout the year such as

- International Noise Awareness Day - Wednesday, 29 April 2020.
- World Hearing Day - 3 March 2020.
- International Mother Language Day - Friday 21 February 2020.

Activities related to education.

These will naturally take longer but the outcome will be long lasting.

The package of activities could include production of any material (digital/video/audio) addressing a particular topic or area or age group.

Also it could include specific events dedicated to teachers and students in collaboration with educational institutes and especially with priparay and secondary level schools.

Activities addressed to the general public

The activation of the Public Media is essential in conveying the message of the International Year of sound to the general public. The National Societies and International Affiliates are encouraged to use all possible means of communication with the community to explain the important of sound for our lives to all the citizens.

TYPES of ICA ORGANIZED AND CENTRALLY FUNDED EVENTS:

The following is a provisional list of centrally organized events for the IYS 2020

- Design of promotional material (posters and leaflets) to be distributed electronically to ICA members and International Organizations. The ICA members will include this material in their web-sites. After printing, these leaflets will be available for distribution to all the events related to the IYS. Banners for the IYS 2020 will be produced to be displayed in the IYS 2020 focal events as well any other events of the ICA and The Week of Sound.
- Organization of the opening ceremony of the IYS 2010 in Paris in 2020. The details and the program of the opening event will be set later this year.
- Organization of world-wide competitions for students of primary and secondary schools respectively. The subject of the competitions will be decided by the IYS 2020 Liaison Committee.
- Development of a video to promote the objectives of the IYS 2020. The film should be of short duration, 5 min max, and will be shown to all conferences and events coordinated by ICA, and its Member Societies and associations as well as to the events of The Week of Sound. This film can also be used as promotional material for the IYS 2020.
- Development of a video for use in education on the importance of sound in our world and to provide guidance on the career opportunities.

MAJOR INTERNATIONL CONFERENCES ASSOCIATED WITH THE IYS 2020

The organisers of at least the following major events during 2020 will be asked to include some form of activity/event in recognition of IYS 2020. Additional International Conferences on Acoustics may be added in this list based on the approval of the ICA Steering Committee.

- FORUM ACUSTICUM (the EAA main conference) Lyon, France, 20-24 April 2020.
- 179th Meeting of the ASA Chicago, Illinois 11-15 May 2020.
- ICSV 27, Prague, Czech Republic, July 2020.
- Internoise 2020, Seoul, Korea, 23-26 August 2020.
- FIA 2020 - 12° Iberoamerican Congress on Acoustics, Florianopolis, Brazil, 27-20 September 2020.
- 180th Meeting of the ASA Cancun, Mexico 9-13 November 2020.

Why publish in Canadian Acoustics?

Because, it is...

- A respected scientific journal with a 40-year history uniquely dedicated to acoustics in Canada
- A quarterly publication in both electronic and hard-copy format, reaching a large community of experts worldwide
- An Open Access journal, with content freely available to all, 12 months from time of publication
- A better solution for fast and professional review providing authors with an efficient, fair, and constructive peer review process.

Pourquoi publier dans Acoustique canadienne ?

Parce que, c'est...

- Une revue respectée, forte de 40 années de publications uniquement dédiée à l'acoustique au Canada
- Une publication trimestrielle en format papier et électronique, rejoignant une large communauté d'experts à travers le monde
- Une publication "accès libre" dont le contenu est disponible à tous, 12 mois après publication
- Une alternative intéressante pour une évaluation par les pairs, fournissant aux auteurs des commentaires pertinents, objectifs et constructifs

Application for Membership

CAA membership is open to all individuals who have an interest in acoustics. Annual dues total $120.00 for individual members and $50.00 for student members. This includes a subscription to ***Canadian Acoustics***, the journal of the Association, which is published 4 times/year, and voting privileges at the Annual General Meeting.

Subscriptions to *Canadian Acoustics* *or* Sustaining Subscriptions

Subscriptions to ***Canadian Acoustics*** are available to companies and institutions at a cost of $120.00 per year. Many organizations choose to become benefactors of the CAA by contributing as Sustaining Subscribers, paying $475.00 per year (no voting privileges at AGM). The list of Sustaining Subscribers is published in each issue of ***Canadian Acoustics*** and on the CAA website.

Please note that online payments will be accepted at http://jcaa.caa-aca.ca

Address for subscription / membership correspondence:

Name / Organization ____________________

Address ____________________

City/Province ____________________ Postal Code __________ Country ____________________

Phone ____________________ Fax ____________________ E-mail ____________________

Address for mailing Canadian Acoustics, if different from above:

Name / Organization ____________________

Address ____________________

City/Province ____________________ Postal Code __________ Country ____________________

Areas of Interest: (Please mark 3 maximum)

1. Architectural Acoustics
2. Engineering Acoustics / Noise Control
3. Physical Acoustics / Ultrasound
4. Musical Acoustics / Electro-acoustics
5. Psychological / Physiological Acoustic
6. Shock and Vibration
7. Hearing Sciences
8. Speech Sciences
9. Underwater Acoustics
10. Signal Processing / Numerical Methods
11. Other

For student membership, please also provide:

__________ __________ ____________________

(University) *(Faculty Member)* *(Signature of Faculty Member)* *(Date)*

I have enclosed the indicated payment for:
[ ] CAA Membership $ 120.00
[ ] CAA Student Membership $ 50.00

Corporate Subscriptions (4 issues/yr)
[ ] $120 including mailing in Canada
[ ] $128 including mailing to USA,
[ ] $135 including International mailing

[ ] Sustaining Subscription $475.00
(4 issues/yr)

Please note that the preferred method of payment is by credit card, online at http://jcaa.caa-aca.ca

For individuals or organizations wishing to pay by check, please register online at *http://jcaa.caa-aca.ca* and then mail your check to:

Executive Secretary, Canadian Acoustical:
Dr. Roberto Racca
c/o JASCO Applied Sciences
2305–4464 Markham Street
Victoria, BC V8Z 7X8 Canada

Formulaire d'adhésion

L'adhésion à l'ACA est ouverte à tous ceux qui s'intéressent à l'acoustique. La cotisation annuelle est de 120.00$ pour les membres individuels, et de 50.00$ pour les étudiants. Tous les membres reçoivent ***l'Acoustique Canadienne***, la revue de l'association.

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