ISSN 0711-6659

canadian acoustics
acoustique canadienne

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

SEPTEMBER 2016
Volume 44 - - Number 3

SEPTEMBRE 2016
Volume 44 - - Numéro 3

Acoustical Consulting: Environmental - Consulting : bruit environnemental 23
Acoustical Consulting: Noise Control - Consulting : contrôle du bruit 47
Acoustical Measurements - Mesures acoustiques 59
Acoustical Standards - Normes acoustiques 69
Architectural Acoustics - Acoustique architecturale 81
Geoacoustic Inversion and Measurement - Inversion et mesures géoacoustique 105
Hearing Health and Aging - Santé auditive et vieillissement 119
Indigenous Speech Communication - Communication verbale indigène 133
Multisensory and Sensorimotor Integration - Intégration multisensorielle 145
Occupational Noise and Hearing Protection - Protection auditive 153
Ocean and Marine-Mammal Acoustics - Acoustique des océans et des mammifères 165
Other Topics and Discussion - Autres sujets et discussions 173
Sounds of Emotion - Sons des émotions 181
Speech Acquisition - Acquisition de la parole 191
Speech Production - Production de la parole 203
Speech Testing - Tests de la parole 215
Sustainable Building Acoustics - Acoustique du batiment durable 237
Vibroacoustics - Vibroacoustique 247
CAA Announcements - Annonces de l'ACA 261



canadian acoustics

acoustique canadienne

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

Association canadienne d'acoustique B.P. 74068 Ottawa, Ontario, K1M 2H9

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.

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 P.B. 74068 Ottawa, Ontario K1M 2H9 Canada - - - secretary@caa-aca.ca - Dr. Roberto Racca

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.

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 P.B. 74068 Ottawa, Ontario K1M 2H9 Canada - - - secretary@caa-aca.ca - Dr. Roberto Racca

EDITOR-IN-CHIEF - RÉDACTEUR EN CHEF

Jérémie Voix
ÉTS, Université du Québec
editor@caa-aca.ca

DEPUTY EDITOR - RÉDACTEUR EN CHEF ADJOINT

Umberto Berardi
Ryerson University
deputy-editor@caa-aca.ca

COPYEDITOR - RELECTEUR-RÉVISEUR

Cécile Le Cocq
copyeditor@caa-aca.ca

ADVISORY BOARD - COMITÉ AVISEUR

Jérémie Voix
ÉTS, Université du Québec

Frank A. Russo
Ryerson University

Ramani Ramakrishnan
Ryerson University

Bryan Gick
University of British Columbia

ADVERTISING EDITOR - RÉDACTEUR PUBLICITÉS

Bernard Feder
HGC Engineering
advertisement@caa-aca.ca



WELCOME TO VANCOUVER!

ACOUSTICS WEEK IN CANADA 2016

September 21-23, 2016

Organizing Committee

Conference Co-chairs:	Kathy Pichora-Fuller, University of Toronto
	Clair Wakefield, Wakefield Acoustics/RWDI
Assistant to Co-chairs & Website Manager:	Philip Tsui, Wakefield Acoustics/RWDI
Technical Program Co-chairs:	Murray Hodgson, U of British Columbia
	Bryan Gick, U of British Columbia
Exhibits/Sponsorship Co-chairs:	Mehrzad Salkhordeh, dB Noise Reduction
	Bernard Feder, HGC ENGINEERING
Event Coordination Chair:	Maureen Connelly, BCIT
Proceedings Preparation:	Cécile Le Cocq, Université du Québec (ÉTS)
	Jérémie Voix, Université du Québec (ÉTS)
Student Presentation Prize Coordination:	Hugues Nélisse, IRSST

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





BIENVENUE À VANCOUVER!

SEMAINE CANADIENNE DE L'ACOUSTIQUE 2016

21 au 23 Septembre 2016

Comité organisateur

Présidents de la conférence:	Kathy Pichora-Fuller, University of Toronto
	Clair Wakefield, Wakefield Acoustics/RWDI
Assistant aux co-présidents & site web:	Philip Tsui, Wakefield Acoustics/RWDI
Directeurs scientifiques:	Murray Hodgson, U of British Columbia
	Bryan Gick, U of British Columbia
Coordinateurs exposants:	Mehrzad Salkhordeh, dB Noise Reduction
	Bernard Feder, HGC ENGINEERING
Coordinateur de la conférence:	Maureen Connelly, BCIT
Préparation des actes:	Cécile Le Cocq, Université du Québec (ÉTS)
	Jérémie Voix, Université du Québec (ÉTS)
Prix pour communications étudiantes:	Hugues Nélisse, IRSST

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



Sponsors/Commanditaires

Platinum/Platine



Gold/Or



Silver/Argent



Other/Autres











Exhibitors/Exposants





















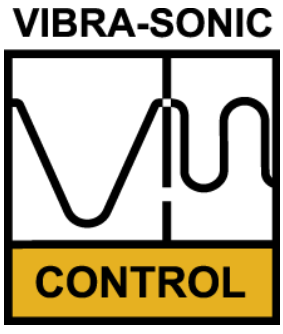












Éditorial invité
Guest Editor's note

Welcome to Vancouver!

We are very happy to welcome delegates to the 2016 Acoustics Week in Canada. Acoustics researchers, professionals, educators, and students from across the country are welcomed to Canada's scenic West Coast for three days of plenary lectures and technical sessions. The Canadian Acoustical Association Annual General Meeting will be held in conjunction with the conference, along with the Acoustical Standards Committee Meeting and an exhibition of acoustical equipment and services. There will be an opening reception on Wednesday September 21st, at the end of the first day of the conference, and the Awards Banquet will be held the next evening, on Thursday September 22nd. The World Congress of Audiology will be held in the same week from September 18th-21st.

The conference will be held at the Sutton Place Hotel, a short walk from the Robson Street shopping area and Stanley Park with its over 1000 acres of land where you can visit the Vancouver Aquarium or enjoy walks in the forest, on the beach or along over 20 km of the Seawall. Vancouver is "spectacular by nature". It is a truly unique and world-class city. It has a mild climate, is safe, clean and friendly. You will find a blend of cosmopolitan amenities, natural splendor and cultural attractions. There are stadiums, dining, shops, art galleries, museums and entertainment within walking distance or easily accessible by a public transportation system of "Sky Trains" and buses. Or you could rent a bicycle and use the network of specialized bike lanes to get around the city like a local.

There are lots of exciting options if you have time to extend your travels in conjunction with the conference. Vancouver is 1.5 hours from Whistler and 3 hours from Seattle. If you want to go further, then it is a 1.5-hour sail on BC Ferries through the Gulf Islands to visit Victoria on Vancouver Island. Or you could take a cruise to Alaska from Vancouver before the close of the season in early October.

We would like to thank all of the Organizing Committee for the hard work and dedication required to plan the conference.





Bienvenue à Vancouver!

Nous sommes très heureux d'accueillir les délégués au congrès de la Semaine canadienne d'acoustique 2016. Les chercheurs et les professionnels en acoustique, de même que les éducateurs et les étudiants de partout au pays sont invités sur la côte pittoresque de l'Ouest canadien pour trois jours de séances plénières et scientifiques portant sur l'acoustique. L'assemblée générale annuelle de l'Association canadienne d'acoustique aura lieu pendant le congrès, ainsi que la rencontre du comité de normalisation en acoustique et une exposition d'équipements et de services. Il y aura une réception à l'ouverture du congrès dans la soirée du mercredi 21 septembre ainsi qu'un banquet de la remise des prix, en soirée du jeudi 22 septembre. Le Congrès mondial en audiologie prendra place dans la même semaine, soit du 18 au 21 septembre.

Le congrès se tiendra à l'hôtel Sutton Place, situé à quelques pas des fameuses boutiques de la rue Robson, de même que du Parc Stanley où vous pouvez aussi visiter l'aquarium de Vancouver. Vous pourrez aussi aller faire de longues promenades en forêt, sur la plage ou encore sur le long des 20 km du Seawall et tout ça, à de très courtes distances de l'hôtel. En plus de sa nature mondialement reconnue, la ville de Vancouver jouit d'un climat bien tempéré. C'est une ville où l'on se sent en sécurité et entouré de gens sympathiques. Vancouver regorge d'attraits cosmopolites, tant sur le plan culturel que sur celui de ses richesses naturelles. On y trouve aussi un stade, des restaurants et cafés, des boutiques et galeries d'art, des musées de même que d'autres attraits et tous sont à des distances facilement faisables à la marche ou accessibles avec les transports publics, sans oublier le fameux «Sky Train». Il est aussi possible de louer un vélo et les multiples pistes cyclables de la ville vous permettront de vous promener en ville comme si vous étiez un résident permanent.

Il y aussi des endroits très intéressants à visiter à proximité de la ville. Par exemple, on peut se rendre à Whistler en 1 heure et demie, ou encore à Seattle en trois heures. Il y a aussi les îles du golfe et même l'ile de Vancouver où l'on peut se rendre en traversier, et ce à moins d'une heure et demie de la ville, sans oublier les croisières en Alaska qui sont fonction jusqu'au début du mois d'octobre.

Nous tenons à remercier tout le comité d'organisation pour le travail acharné et le dévouement nécessaires pour planifier la conference.

See you soon/ à bientôt
Kathy Pichora-Fuller and Clair Wakefield
Conference Chairs

Travel information

Hotel Information

The Sutton Place Hotel is located in the heart of downtown Vancouver on Burrard Street near Robson Street. It boasts a signature restaurant and wine merchant, and an elegance that few hotels can rival, with European charm and exceptional service.

Rooms booked as part of the conference group booking include the following:

- **Complimentary Internet:** wireless/high-speed internet in guestrooms and wireless internet in function rooms.
- **Health Club and Pool:** free access to the fitness facility and swimming pool for overnight guests.
- **The Sutton Shopper Program:** the Sutton Shopper Program, exclusive to the Sutton Place Hotel, will be offered to every guest with the group. This program enables guests to receive discounts between 10% - 30% at over fifty shops and services along the Robson Street shopping corridor.

Address: 845 Burrard St, Vancouver, V6Z 2K6
Telephone: 604-682-5511
Email: info_vancouver@suttonplace.com
On Line: **www.vancouver.suttonplace.com**





Informations voyageur

Information sur l'hôtel du congrès

L'hôtel Sutton Place est située au centre-ville de Vancouver sur la rue Burrard, tout près de la rue Robson. En plus d'un restaurant signature et d'un marchand de vin, cet hôtel au charme européen et d'une élégance inégalée offre un service exceptionnel.

Le tarif pour une chambre pendant le congrès inclut les services suivants:

- **Service internet gratuit:** l'internet sans fil et à haute vitesse est disponible dans toutes les chambres et les salles de réunion.
- **Centre de conditionnement et piscine:** les invités de l'hôtel bénéficient d'un accès gratuit aux équipements du centre de conditionnement de même qu'à la piscine.
- **Le programme "Sutton Shopper Program":** Ce programme, exclusif à l'hôtel, est offert à tous les délégués qui hébergent à l'hôtel et donne droit à des rabais de 10% à 30% dans plus de 50 boutiques et services offerts de la rue Robson.

Adresse: 845 Burrard St, Vancouver, V6Z 2K6
Téléphone: 604-682-5511
Email: info_vancouver@suttonplace.com
En ligne: **www.vancouver.suttonplace.com**



Getting to the hotel

From Vancouver International Airport (YVR)
YVR is about 13 km/8 miles from the hotel.
Taxi: flat rate is $31 from the airport to the hotel.
SkyTrain (http://www.translink.ca/):
Canada Line YVR Station (towards Waterfront Station) to Vancouver City Centre station (25 minutes; $7.75-$9.00) then walk .4 km south on Granville to Robson Street then west to Burrard St then south to hotel (map on right).
Driving from YVR:

- From YVT, follow the signs towards Vancouver.
- Proceed across Arthur Laing Bridge.
- Exit onto Granville Street.
- Continue on Granville Street.
- Proceed across Granville Street Bridge to downtown.
- Turn left onto Davie Street.
- Turn right onto Burrard Street.
- Turn left into the hotel entrance.

From East

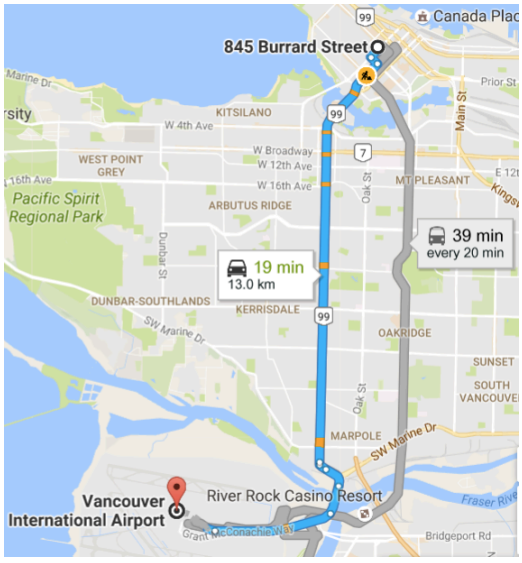
- From TransCanada Highway 1, take the Hastings Westbound Exit into downtown
- Turn left onto Burrard Street
- Turn right into the hotel.

From South

- From Highway 99, continue on Oak Street.
- Turn left onto 49th Street.
- Turn right onto Granville Street.
- Proceed across the Granville Street Bridge.
- Turn left onto Davie Street.
- Take a right onto Burrard Street.
- Hotel is on the left.

From North

- From Highway 99, take the Taylor Way Exit (99A)
- Proceed across Lions Gate Bridge and through Stanley Park.
- Continue onto Georgia Street.
- Turn right onto Burrard Street.
- Hotel is on the right.



Pour se render à l'hôtel

L'aéroport international de Vancouver (YVR)
La distance à parcourir est environ 13 km/8 miles.
Taxi: $31 Tarif forfaitaire pour un trajet.
SkyTrain (http://www.translink.ca/):
Canada Line YVR Station (vers Waterfront Station) à Vancouver City Centre Station (25 minutes; $7.75-$9.00), puis marcher .4 km vers le sud sur la rue Granville à la rue Robson Street, puis vers l'ouest à la rue Burrard, puis vers le sud a l'hôtel (la carte à droite).
En voiture de YVR:

- Depuis YVR, suivez les panneaux indiquant Vancouver.
- Traversez le pont Arthur Laing.
- Sortez et engagez-vous sur la rue Granville.
- Continuez sur la rue Granville.
- Traversez le pont Granville.
- Tournez à gauche sur la rue Davie.
- Tournez à droite sur la rue Burrard.
- L'hôtel se trouve à gauche.

En venant de l'est

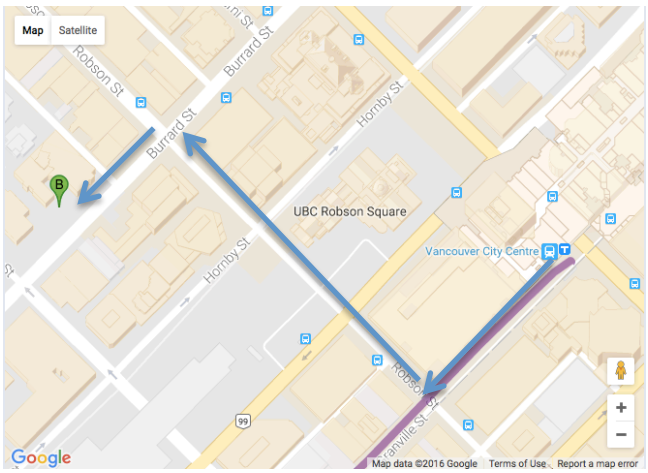
- Depuis l'autoroute TransCanada Highway 1, prenez la sortie Hastings Westbound direction centre-ville.
- Tournez à gauche sur la rue Burrard.
- L'hôtel se trouve à droite.

En venant du sud

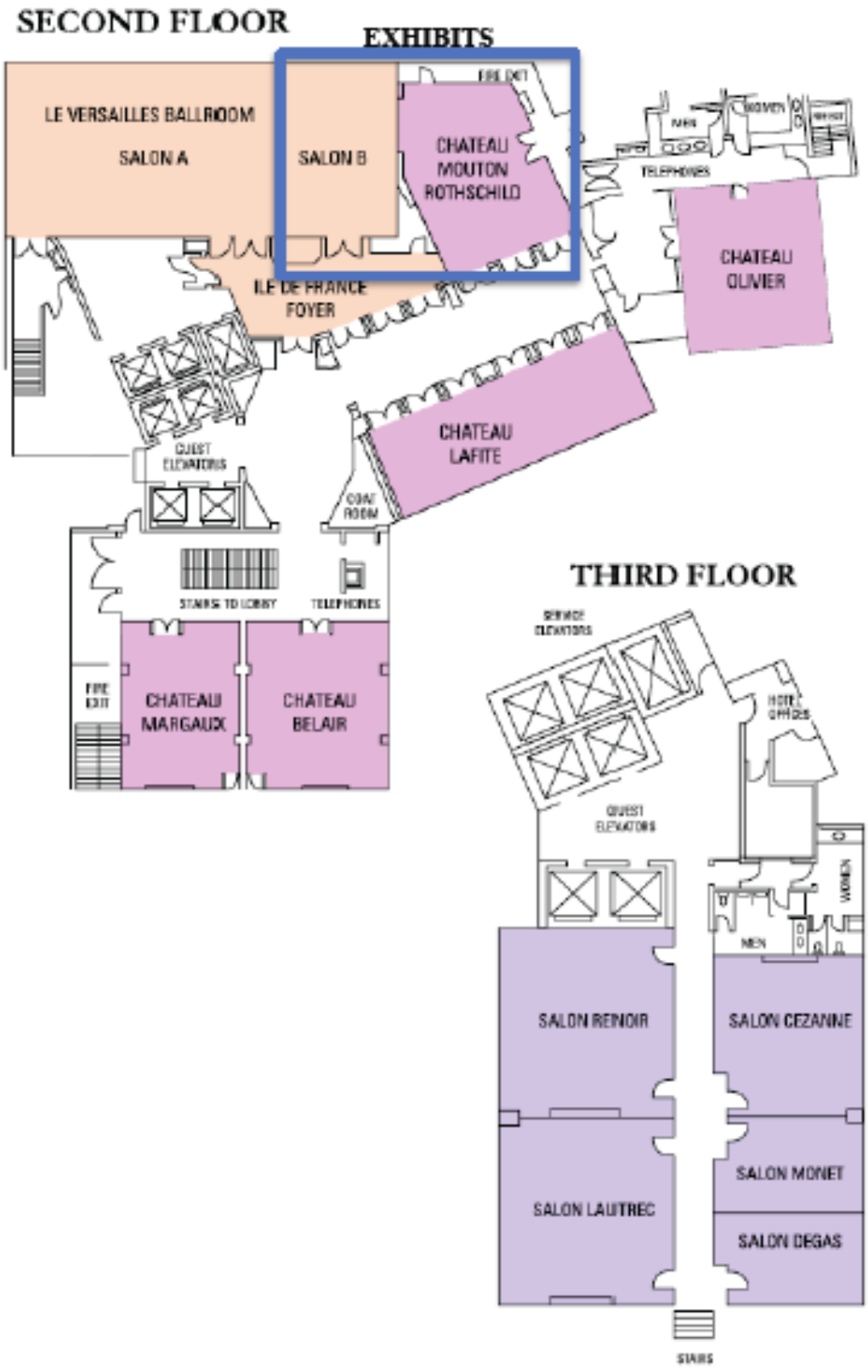
- Depuis l'autoroute 99, continuez sur la rue Oak.
- Tournez à gauche sur la rue 49th.
- Tournez à droite sur la rue Granville.
- Traversez le pont de la rue Granville.
- Tournez à gauche sur la rue Davie.
- Prenez à droite sur la rue Burrard.
- L'hôtel se trouve à gauche.

En venant du nord

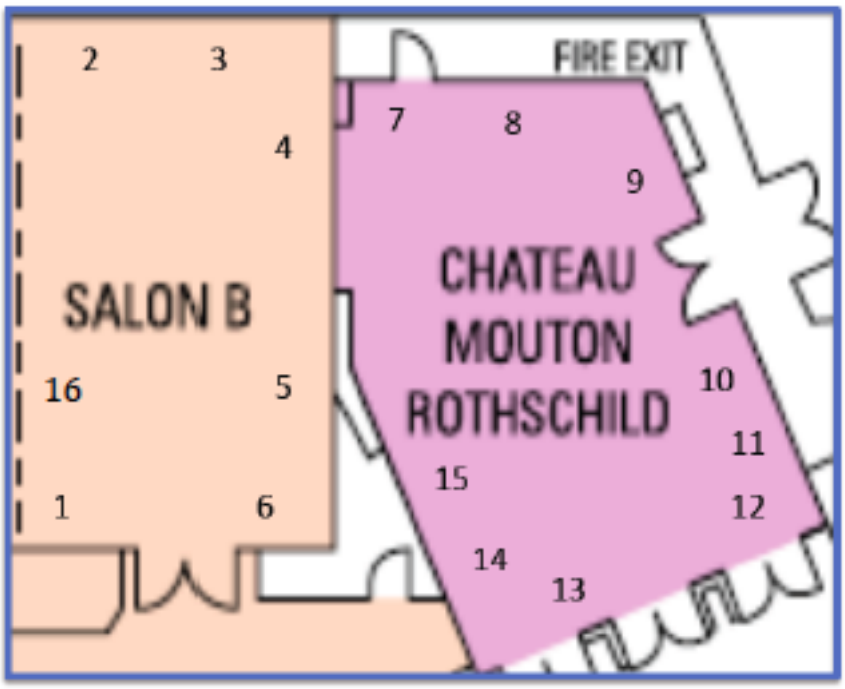
- Sur l'autoroute 99, prenez la sortie Taylor Way (sortie 99A)
- Traversez le pont Lions Gate et passez par Stanley Park.
- Continuez sur la rue Georgia.
- Tournez à droite sur la rue Burrard.
- L'hôtel se trouve à droite.



Hotel map/carte



EXHIBITS



1. SVS Canada
2. Vibra-Sonic
3. Pyrok
4. Acoustic Solutions
5. BKL Consultants
6. Proximify
7. Regupol
8. **ARC**ACOUSTICS
9. Johns Manville
10. Brüel &Kjær
11. ROXUL
12. Rockfon
13. Pliteq
14. Dalimar
15. Sound-rite
16. JASCO

PROGRAM - PROGRAMME

TUESDAY SEPTEMBER 20 / MARDI 20 SEPTEMBRE						
ROOM	SALON LAUTREC		CHATEAU MARGAUX			
	MEETING (3:00 start)	CHAIR	MEETING (7pm start)	CHAIR		
	Board of Directors Meeting	Russo	Standards Committee Meeting	Kelsall		

WEDNESDAY SEPTEMBER 21 / MERCREDI 21 SEPTEMBRE							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
7:30	Registration - Foyer						
8:20-8:30	Welcome	Opening Comments	Wakefield				
8:30-10:00	Invited Plenary (Versaille A)	**Age-Related Hearing Loss: Interdisciplinary Teams to Advance Research and Practice in Acoustics**	Dubno				
8:45-10:15	Forum (Chateau Mouton Rothchild)	**Advancing architectural and building acoustics with the Building Envelope / Building Science Community**	Connelly				
10:00-10:30	Coffee Break	Sponsored by Pliteq					
10:30-12:00	Contributed Papers	**Hearing Health as a Key Component of Healthy Aging**	Dupuis	**Vibroacoustics**	Nélisse	**Acoustical Consulting: Noise Control**	Bliss
10:30-10:45		Association between sensory loss and social outcomes: A preliminary report	Parfyonov, Mick, Pichora-Fuller & Wittich	Laboratory and field measurements of nail guns' noise emission	Gaudreau, Laville, Marcotte & Boutin	Laboratory impact and airborne sound attenuation in cross-laminated timber construction using resiliently mounted ceilings and poured concrete toppings.	Byrick
10:45-11:00		Addressing social isolation in hearing rehabilitation	Reed	Quantification of footstep noise transmission in the interior of an apartment building	Magalhães	Noise control for high plume exhaust fans: A case study	VanDelden, Downey & Downey
11:00-11:15		Improving communication for persons with dementia	Mamo, Mayhew, McNabney, Oh & Lin	Comparison of two methods of transfer path analysis applied to snowmobile for noise source identification	Guenfoud, Robin, Panneton, Desrochers & Belgacem	Effectiveness of noise reducing asphalt pavements	Penton
11:15-11:30		Promoting hearing and communication health in long-term residential care through the use of communication apps	Small, Wilson , Mihailidis & Cochrane	A data processing method for noise measurements of snowmobiles and their sub-systems on test bench	Méjane, Panneton, Desrochers & Oddo	Noise mitigation of small PTAC systems	Penton, Arnold & Walters
11:30-11:45		Audiologic care for clients with multiple medical comorbidities: Modifications to clinical practice and inclusion of family members and caregivers	Dupuis , Lemke, Reed & Pichora-Fuller	Understanding sources and sound radiation of a snowmobile track	Beuvelet, Panneton, Desrochers & Oddo	Selecting suitable noise control for mine return air raise systems	Dobson & Gastmeier
11:45-12:00		Acoustic factors related to emotional responses to sound	Picou	Optimization study and panel parameter study for noise radiation reduction of an aircraft panel excited by turbulent flow	Sonnenberg & Rocha	Predicting exhaust sound power levels of general purpose boilers	Stevens & Gastmeier

WEDNESDAY SEPTEMBER 21 / MERCREDI 21 SEPTEMBRE							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
12:00-1:00	Lunch	Sponsored by **ARC**OUSTICS					
1:00-2:30	Contributed Papers	**Speech Production**	Gick & Vatikiotis-Bateson	**Sustainable Building Acoustics**	Connelly	**Acoustical Consulting: Environmental Part 1**	Salkhordeh
1:00-1:15		Individual variations and gradience in English palatalization	Sung	Sustainability in acoustic design	Borch & Martin	Effects of traffic density on measured sound levels – A case study	Matthew & Amarra
1:15-1:30		Dental overjet, articulatory compensation and acoustics of speech	Cheng, Doe, Lew, Henderson, Schellenberg & Gick	Acoustical investigation of laneway house facades	Lin & Connelly	Noise levels from heavily travelled roads for use with environmental noise regulations in British Columbia and Alberta	Daltrop, Faszer & Young
1:30-1:45		Schlieren study of external airflow during the production of nasal and oral vowels in French	Rowell, Noguchi, Bernhardt, Herdman, Gick & Schellenberg	Increasing sound insulation of window with add-on systems & various insulating glass unit configurations	Mak & Connelly	Monitoring road traffic sound levels	Gastmeier & Paul
1:45-2:00		(Canadian) raise your vowels in song	Schellenberg & Ozburn	Investigation of airflow and sound transmission through vented-acoustical panels for natural ventilation	Higgins, Shankar & Hodgson	Modular expansion joint noise in B.C.	Marriner & Wakefield
2:00-2:15		Affricate contrasts in Eastern Oromo	Percival	Sound absorption and scattering properties of living walls: Applications to room acoustics	Connelly, Bolbolan, Akbarnejad & Daneshpanah	Comparison of aircraft noise exposure from NEF contours and field measurement results	Gu & Bliss
2:15-2:30		Japanese EFL learners' production of pronouns and articles in English: Evidence for L2 prosodic structures	Yamane, Yoshimura & Fujimori	Achieving better sound insulation of CLT partitions	Sabourin, Hoeller & Zeitler	Acoustical verification testing of ground run-up enclosure at Vancouver International Airport	Bliss, Cheng, Min, Reeves & McMullan

WEDNESDAY SEPTEMBER 21 / MERCREDI 21 SEPTEMBRE							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
2:30-3:00	Coffee Break	Sponsored by Jade Acoustics					
3:00-4:30	Contributed Papers	**Indigenous Speech Communication**	Bird	**Acoustical Measurements**	Gastmeier	**Acoustical Consulting: Environmental Part 2**	Wakefield
3:00-3:15		Acoustic characteristics of preaspirated stops in Plains Cree	Schmirer	Acoustical effects of wind screens on infrasound measurements	Busch, Yung, Thorbahn & Swallow	Community noise investigation of a "low level thrumming" attributed to the operation of mine	Williamson
3:15-3:30		Articulatory conflict resolution strategies among L1 and L2 Sencoten speaker	Bird	Identifying localized noise sources during industrial vehicle passbys	Reniak	Considerations regarding an acoustic criterion for wind turbine acceptability	Palmer
3:30-3:45		The realisation of obstruents across speakers of Hul'q'umi'num'	Bird, Gerdts & Leonard	Time domain source localization technique based on generalized cross correlation and generalized mean	Padois, Doutres, Sgard & Berry	A review of the BC wind power project acoustic assessment guidelines	Martin & Bliss
3:45-4:00		Shifty vowels: Variation in dialectal lowering in Gitksan	Fortier	Development of the second generation NRC acoustic spectrum control system for high intensity noise testing	Grewal, Chen, Alavi, Wickramasinghe & Lawrie	Comparison between modelled and measured sound levels for a southern Alberta wind farm	de Haan & Senden
4:00-4:15		Ultrasound overlay videos and their application in Indigenous language learning and revitalization	Bliss, Burton & Gick	Colt C8A2 carbine impulsive noise auditory hazard assessment through testing in a reverberant environment	Ghinet, Price, Grewal, Chen & Wickramasinghe	Meaningful measurements to assess transformer noise	de Haan & Senden
4:15-4:30		Acoustic characteristics of obstruents in Huehuetla Tepehua	Puderbaugh				
5:00-6:30	Reception - Chateau Lafite	Sponsored by **ARC**OUSTICS					

THURSDAY SEPTEMBER 22 / JEUDI 22 SEPTEMBRE (EXHIBITS/L'EXPOSITIONS - 9am to 5pm - Versailles B/Rothschild)							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
7:30	Registration - Ile de France Foyer						
8:20-8:30	Welcome	Opening Comments	Pichora-Fuller				
8:30-10:00	Invited Plenary	**Hearing Loss Prevention in 2016 and Beyond**	Morata				
10:00-10:30	Coffee Break	Sponsored by B&K					
10:30-12:00	Contributed Papers	**Speech Testing Part 1**	Ellaham & Lagacé	**Architectural Acoustics Part 1**	Hodgson	**Occupational Noise and Hearing Protection**	Voix
10:30-10:45		An ecological approach to assess auditory perception.	Bergeron et al.	Acoustical properties of homogeneous planar surfaces in room acoustics: A theoretical review	Yousefzadeh & Hodgson	Custom molded silicon earplugs: Effect of material properties on acoustic attenuation and mechanical skin contact	Viallet & Voix
10:45-11:00		Intelligibility of older talkers on the NU6 test	Goy, Pichora-Fuller & van Lieshout	Sound field diffuseness in oblique-shaped reverberation rooms with different test configurations	Hasan & Hodgson	Ear canal deformations by various earplugs: An in situ investigation using MRI	Benacchio, Varoquaux, Wagnac, Doutres, Callot, Bendahan & Sgard
11:00-11:15		Speech auditory brainstem responses and words-in-noise recognition scores: Is there a link?	Lagacé, Koravand, Thompson & Lteif	Objective acoustical quality in healthcare office facilities	Hodgson	Effect of ear canal occlusion on loudness perception	Bonnet, Voix & Nélisse
11:15-11:30		Effects of modality and linguistic materials on memory	Pattison, Pichora-Fuller & Smith	Subjective acoustical quality in healthcare office facilities	Hodgson	Pilot study on individual dose-response relationship evaluated through otoacoustic emission measurements in controlled noise exposure: Influence of circadian rhythm	Nadon, Bockstael, Botteldooren & Voix
11:30-11:45		A comprehensive approach for assessing the effectiveness of frequency-lowering hearing aids and electric acoustic stimulation (EAS) cochlear implant for treating people with a severe-to-profound high-frequency hearing loss	Hotton & Bergeron	DL2 requirements for speech intelligibility and privacy in rooms	Ng & Hodgson	Investigation of aircrew noise exposure onboard the NRC Dassault Falcon aircraft through flight testing	Ghinet, Price & Wickramasinghe
11:45-12:00		Laboratory assessment of daily-life speech understanding	Kalluri, Xia & Hafter	Scale-model investigation of the effect and optimal design of sound absorbers in an open-plan office	Mahmud & Hodgson	Using CSA Standard Z107.56 to investigate noise exposure of call center employees	Daltrop & Bessey

THURSDAY SEPTEMBER 22 / JEUDI 22 SEPTEMBRE (EXHIBITS/L'EXPOSITIONS - 9am to 5pm - Versailles B/Rothschild)							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
12:00-1:00	Lunch	Sponsored by Johns Manville					
1:00-2:30	Contributed Papers	**Speech Testing Part 2**	Lagacé & Ellaham	**Architectural Acoustics Part 2**	Hodgson	**Ocean and Marine-Mammal Acoustics**	Dosso
1:00-1:15		Listening to speech in noise : Evaluation and training	Jutras, Owliaey, Lafontaine, Pinsonnault-Skvarenina & Gagné	Optimizing ceiling systems and lightweight plenum barriers to achieve Ceiling Attenuation Class (CAC) ratings of 40, 45 and 50	Madaras & Heuer	Automatic detection and visualization of marine mammal sounds in British Columbia, Canada	Mouy, Mouy & Hannay
1:15-1:30		A software platform to administer the Canadian Digit Triplet Test	Ellaham, Giguère, Lagacé & Pichora-Fuller	Acoustics in Alberta schools - Overview and challenges	Moquin & Buchan	Passive acoustic localization and density estimation of fin whales using a single hydrophone	Rideout
1:30-1:45		Intelligibility, response time and listening efficiency of native and non-native listeners in various acoustical conditions	Lam & Hodgson	The dynamic response of composite floating floors in buildings	Magalhães	Blind channel estimation of underwater acoustic waveguide impulse responses using marine mammal vocalizations	Rideout, Nosal & Host-Madsen
1:45-2:00		What do forced alignment likelihood scores tell us about the aligned speech?	Mrigen, Brenner & Tucker	Acoustical design for 21st century learning	Marks	Interaction between learned and naturally selected features in the design of killer whale calls and their functions	Yurk
2:00-2:15		Discrimination of aspiration noise in breathy vowels	Labuschagne & Ciocca	Noise control issues in a steel-framed institutional building	Wakefield	Efficient estimation of range-dependent seabed properties from large data volumes of a towed source and receiver-array system	Dettmer, Quijano, Dosso, Holland & Mandolesi
2:15-2:30				Exterior noise ingress in Greater Vancouver residential buildings: Prediction versus field measurements	Mak & Bliss	An automated real-time vessel sound measurement system for calculating monopole source levels using a modified version of ANSI/ASA S12.64-2009	Hannay, Mouy & Li

THURSDAY SEPTEMBER 22 / JEUDI 22 SEPTEMBRE (EXHIBITS/L'EXPOSITIONS - 9am to 5pm - Versailles B/Rothschild)							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
2:30-3:00	Coffee Break	Sponsored by HGC ENGINEERING					
3:00-4:30	Contributed Papers	**Multisensory & Sensorimotor Integration**	Lau & Campos	**Acoustical Standards**	Kelsall	**Geoacoustic Inversion & Acoustical Measurements**	Racca
3:00-3:15		Auditory, tactile and bimodal synchronization to the beat: A test of the auditory-motor enhancement hypothesis	Russo, Ammirante & Patel	CSA Z1007: A new management standard from the Canadian Standards Association for the management of hearing loss prevention programs	Goldberg, Kelsall, Behar & Malek	Bayesian inversion of time-difference-of-arrival data to localize bowhead whales in the Chukchi Sea	Warner, Dosso & Hannay
3:15-3:30		Rethinking multimodal integration: The importance of event unity	Schutz	CSA Z107.56 and safe use of music players	Kelsall	Geoacoustic inversion of reflection data in a water tank	Shi, Dosso & Sun
3:30-3:45		Sensorimotor integration of vestibular, proprioceptive, and visual information about head motion in the interpretation of dynamic auditory cues for front/back sound localization	Macpherson & Kim	CSA Z107.6 audiometric testing for use in hearing loss prevention programs: A new title for a new edition	Brown, Giguère & Sharpe	Gradient representations in seabed geoacoustic inversion by Bernstein polynomials	Quijano, Dosso, Holland & Dettmer
3:45-4:00		Using spatialized sound to enhance self-motion perception in virtual environments and beyond: Auditory and multi-modal contributions	Riecke	Update on communication headset noise measurements in the workplace	Nassrallah, Ellaham, Giguère & Dajani	Advances in acoustic monitoring	Kinart & Gastmeier
4:00-4:15		Preliminary findings from investigations of the relationship between hearing and segmental orientation during walking	Lau, Cinelli, Zabjek, Ellaham & Campos	Changing standards for building acoustics regulation	Quirt	The potential of remote real-time noise monitoring	de Haan & Senden
4:15-4:30		Quantifying the time-varying relation between speech production and postural control at different levels of vocal effort	Fuhrman, Vilela-Barbosa & Vatikiotis-Bateson	Standards for modelling road traffic noise in Western Canada	Borch & Bliss	The evolution of environmental ground borne vibration measurement, assessment and reporting over the last 50 years	Katsiroumpas & Matthew
7:00-11:00	Banquet - Versaille A	Sponsored by Pliteq					
	Awards						
	"Performing in Flatland" (Zappi, Vasudevan & Fels)	Sponsored by Pliteq					

FRIDAY SEPTEMBER 23 / VENDREDI 23 SEPTEMBRE							
ROOM		**VERSAILLES A**		**CHATEAU MARGAUX**		**CHATEAU OLIVIER**	
TIME	**SESSION**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**	**TITLE**	**CHAIR/AUTHOR(S)**
7:30	Registration - Ile de France Foyer						
8:20-8:30	Welcome	Opening Comments	Pichora-Fuller				
8:30-10:00	Invited Plenary	**Interdisciplinary Approaches for Advancing articulatory Speech Theory and Synthesis**	Fels & Gick				
10:00-10:30	Coffee Break	Sponsored by JASCO Applied Sciences					
10:30-12:00	Contributed Papers	**Speech Acquisition**	Yeung	**Sounds of Emotion**	Russo & Nespoli	**Other Topics and Discussion**	Wakefield
10:30-10:45		Does training to perceive L2 English vowels in one phonetic context transfer to other phonetic contexts?	Thomson	Perception of emotional speech by listeners with hearing aids	Goy, Pichora-Fuller, Singh & Russo	The lasso and elastic-net algorithms for predictive sound quality models using large pool of predictive metrics and factors	Benghanem, Gauthier, Leblanc & Berry
10:45-11:00		Perceptual learning of speech sounds: A bias for 'sipper' over 'zipper'?	Babel, Lawler & Norton	Skin conductance responses to emotional speech in hearing-impaired and hearing-aided listeners	Nespoli, Singh & Russo	Effect of variability in micro-geometry of polyurethane foams on the double wall transmission loss	Gholami, Doutres & Atalla
11:00-11:15		Audio-aerotactile integration in infant speech perception	Keough, Kandhadai, Werker & Gick	Perception of emotion in music by hearing-impaired and hearing-aided listeners	Russo & Fanelli	The case for a Canadian trade association for acoustic consulting companie s	de Santis & Meszaros
11:15-11:30		Infants' use of temporal and phonetic information in the encoding of audiovisual speech	Danielson, Tam, Kandhadai & Werker	There's something about that groove: Rhythm improves detection of audio but not vibrotactile asynchronies	Lauzon, Russo & Harris	Discussion of de Santis proposal	Wakefield
11:30-11:45		Does swallowing bootstrap speech learning?	Mayer, Roewer-Despres, Stavness & Gick	Emotional communication in music: Implications for understanding links between between speech and music	Schutz		
11:45-12:00		Breathing position and the perception of consonants	Yeung, Scott & Granjon	Environmental sound and its relation to human emotion	Truax		
12:00-1:00	Lunch	Sponsored by ROXUL					
	Student Presentation Awards						

Plenary Lectures
Séances plénières et scientifiques

Wednesday September 21
Mercredi 21 septembre



Judy R. Dubno from the Medical University of South Carolina will talk about age-related hearing loss and how interdisciplinary teams can advance research and practice in acoustics.

Judy R. Dubno de la Medical University of South Carolina viendra parler sur la perte d'audition associée au vieillissement et comment les équipes interdisciplinaires peuvent faire avancer la recherche et les pratiques en acoustique.

Abstract: A hallmark of acoustical science and practice is its interdisciplinary focus. Indeed, teams of acoustical scientists and practitioners working together toward the same goal can include physicists; electrical, mechanical, and biomedical engineers; computer scientists; speech scientists and linguists; psychologists; audiologists; biologists, physicians, and neuroscientists; architects; and musicians. This diversity of expertise has benefited the science and practice that is addressing the growing problem of age-related hearing loss. Hearing loss is among the most common chronic conditions of aging, ranking first among males and fourth among females. Studies of hearing across the lifespan reveal that changes in auditory function can begin in middle age. The prevalence and severity of this condition will increase as the population ages, but only about 20% of adults who require intervention seek help for their condition. This presentation will focus on how interdisciplinary teams of acoustical scientists and practitioners are leveraging their diverse expertise to improve the design of acoustical spaces, enhance communication technologies for speech recognition and music appreciation, reveal population trends in health economics of hearing loss, protect hearing by controlling noise levels, and conduct basic and clinical research to improve diagnosis, prevention, and treatment of age-related hearing loss. An interdisciplinary team of acousticians is best able to challenge this complex disorder of aging that contributes to poor communication abilities and reduced quality of life for millions of older adults. Work supported by NIH/NIDCD

Bio: **Judy R. Dubno**, Ph.D., is a Professor and Director of the Hearing Research Program in the Department of Otolaryngology–Head and Neck Surgery at the Medical University of South Carolina in Charleston. Her research, which is supported by grants from the National Institutes of Health (NIH)/National Institute on Deafness and Other Communication Disorders (NIDCD), focuses on auditory perception and speech recognition in adverse listening conditions and how perception changes with age, hearing loss, hearing aids, and training. She previously served on the Advisory Council of the NIH/NIDCD, four Institute of Medicine/National Academies of Science consensus committees, as President and Secretary-Treasurer of the Association for Research in Otolaryngology, and as President of the Acoustical Society of America. She is a Fellow of the Acoustical Society of America and the American Speech-Language-Hearing Association and the recipient of the James Jerger Career Award for Research in Audiology.

Thursday September 22
Jeudi 22 septembre



Thais Morata from the National Institute for Occupational Safety and Health (NIOSH) and the Center for Disease Control (CDC) will talk about hearing loss prevention in 2016 and modern ways to communicate to the public about hearing health and noise.

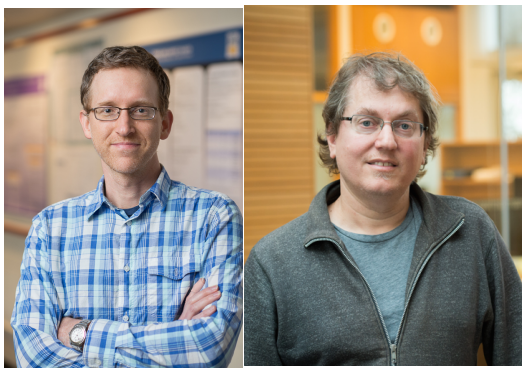
Thais Morata de la National Institute for Occupational Safety and Health (NIOSH) and et du Center for Disease Control (CDC) viendra parler de la prévention de la perte auditive en 2016 et des méthodes modernes de parler en publique de la santé auditive et du bruit.

Abstract: Noise and hearing loss place a significant burden on society, a burden that can be measured. One way to measure this impact is to calculate disability-adjusted life years (DALYs). This is a number of healthy years lost due to a disease or other health condition. For a condition like hearing loss the DALYs calculation takes into account life limitations caused by hearing loss as a lost portion of a healthy year of life. It is an approach that can be used to quantify the impact of hearing loss on critical intangibles, such as communication and mental health. NIOSH recently used DALYs to estimate the impact of hearing loss on quality of life. The results denote that prevention of hearing loss is a battle that

requires expanding and modernizing preventive initiatives and even requires the integration of occupational safety and health protection with health promotion. Today we have fewer obstacles and more tools than were available in past decades, such as technology that allows noise measurements with smartphones, hearing protection fit testing and faster and less expensive means to communicate with the public. Still, we still need to evaluate the effectiveness of our actions related to noise control, hearing loss and tinnitus prevention. We need to incorporate plans for research and intervention evaluations that will result in evidence-based recommendations and practices. This session will cover factors affecting our hearing health and initiatives that can impact not only individual, but society's decisions, and promote change to improve lives.

Bio: **Thais C. Morata** earned a doctoral degree in Communication Sciences and Disorders from the University of Cincinnati in 1990. Currently she is a Research Audiologist at the National Institute for Occupational Safety and Health (NIOSH, Cincinnati) and the Coordinator of the National Occupational Research Agenda Manufacturing Sector Council, a network of partners and stakeholders who collaborate through activities encompassing the entire research continuum (conceiving, planning, conducting, translating, disseminating and evaluating research). Dr. Morata created and directs the Safe-in-Sound Excellence in Hearing Loss Prevention Awards™. She is a Founding Associate Editor for the International Journal of Audiology (from the International Society of Audiology), and a Founding Editor of the Cochrane Work Review Group. Her pioneering work in the area of noise interactions in the workplace has influenced not only NIOSH priorities and policy, but has affected national and international occupational safety and health policies. More recently, she is devoting time to the goals of improving the communication of science to the public through new media, and promoting the evaluation of intervention effectiveness and adoption of evidence-based health practices.

Friday September 23
Vendredi 23 septembre



Bryan Gick and Sid Fels from the University of British Columbia will talk about "Interdisciplinary approaches for advancing articulatory speech theory and synthesis".

Bryan Gick et Sid Fels de l'University of British Columbia vont venir parler des approches interdisciplinaires pour l'avancement des théories articulatoires de la parole et de leur synthèse.

Abstract: For years, articulatory synthesis research has been largely overshadowed by frequency domain and concatenate sample-based speech synthesis techniques. While successful in some domains (e.g., voice-based databases), these techniques still cannot produce natural looking and sounding speech from text from an arbitrary speaker. Natural looking and sounding speech technology is one of the next major milestones in voice-based interaction for natural user interfaces. Through a team of interdisciplinary researchers, we have been steadily working towards creating the necessary platform to overcome basic problems in speech production and, we believe, represents the next major advance in speech synthesis technology. We will discuss our progress on articulatory speech synthesis using 3D biomechanical models of the vocal tract and the advances in understanding of speech production and synthesis produced from it.

Bio: **Bryan Gick** (Professor and Head, UBC Linguistics; PhD Yale 1999) is Co-chair of the UBC Language Sciences Initiative and Senior Researcher at Yale's Haskins Laboratories (New Haven, CT). With ongoing collaborations spanning a half-dozen UBC faculties and papers in such venues as *Nature* and *Neuroscience*, Gick is widely known for his research on speech production, perception, and motor control of speech. He is most widely known for his work in multimodal speech perception, for his work in the biomechanics and neuromuscular control of speech, and for his work developing ultrasound imaging as a tool for biofeedback in pronunciation learning and teaching.

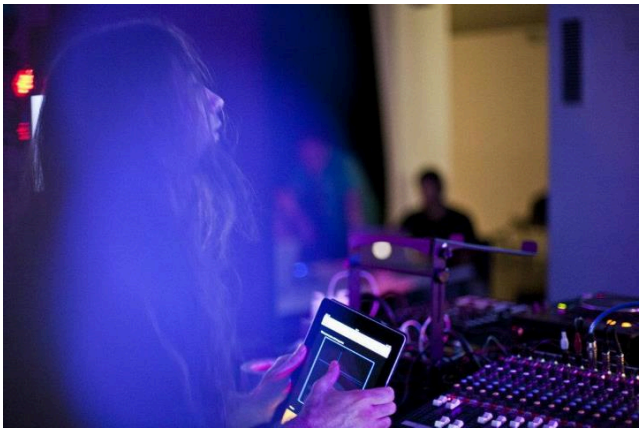
Sid Fels has been a Professor in Electrical and Computing Engineering at UBC since 1998 and a Distinguished University Scholar since 2004. He was the Director of the Media and Graphics Interdisciplinary Centre (MAGIC) at UBC from 2001 to 2012). He earned a PhD and MSc in Computer Science at the University of Toronto (1994) and a BASc in Electrical Engineering at the University of Waterloo (1988): He was a visiting researcher at the ATR Media Integration & Communications Research Laboratories in Kyoto, Japan (1996-1997) and worked at Virtual Technologies Inc. in California (1995). He has with over 250 publications and is internationally known for his work in human-computer interaction, neural networks, biomechanical modeling, and new interfaces for musical expression and interactive arts. He is the academic leader on the biomechanical modeling efforts for the Parametric Human project and leads a collaborative health research project on modeling speaking and swallowing.

Performing in Flatland

Victor Zappi, Arvind Vasudevan, Sidney Fels

Performing in Flatland is an audio/visual performance inspired by the physics of sound and, in particular, by the theory of pressure wave propagation in space.

Sound is part of our everyday lives, we are constantly immersed in it and our brain is capable of retrieving an impressive amount of information from it. Our relationship with sound is an audio/visual experience. In *Performing in Flatland*, we try to explore this paradigm more deeply, by means of coupling sound and music with a realistic visual representation of how these waves propagate and interact with obstacles.





Victor Zappi (above), Arvind Vasudevan (lower left), Sidney Fels (lower right)

Throughout the piece, a virtual representation of a two dimensional physical space domain is projected in front of the audience, where the music played by the performer is visualized as coloured waves travelling on the surface of the screen. Through a digital music interface, the performer interactively manipulates the pixels that compose the propagation domain, drawing reflecting walls, resonant chambers and more complex dynamic geometric structures. These elements realistically reflect and absorb the sound injected into the domain, producing complex spectral modifications that are used by the performer as primary means of composition. The interaction between the sound sources and the geometry of the domain is also visualized on the screen as clear changes of waves' trajectories and recursive motion patterns, generating a multimodal representation of the involved acoustic phenomena. Furthermore, during the performance these geometries are coupled with different self-oscillating excitation models, to create and play 2D virtual instruments inspired by real aerophones and human voice production.

Performing in Flatland is the premiere of a novel digital technology that uses GPU parallel computing to simulate pressure wave propagation in real-time, both sonically and visually. Pressure and velocity values are calculated in the 2D domain using an FDTD solver, implemented in a shading language. Each FDTD cell corresponds to a pixel of a dynamic texture that is updated on the GPU and rendered on screen. The simulation runs at audio rate; at each cycle, the pressure values of a chosen position on the texture are sampled and packed into a buffer, which is then sent to the audio interface to provide a continuous sound stream.

The system was originally developed in the context of research in acoustics and speech. For the performance it has been transformed into an audio/visual instrument and is used together with electronic music controllers as a means of artistic expression.

We would like to thank Dr. Andrew Allen and Dr. Nikunj Raghuvanshi for their help and support. Without their contribution this work would not be possible.

INTERDISCIPLINARY APPROACHES FOR ADVANCING ARTICULATORY SPEECH THEORY AND SYNTHESIS

Sidney Fels[1] and Bryan Gick[2, 3]
[1] Dept. of Electrical and Computer Eng., University of British Columbia, 2356 Main Mall, Vancouver, BC, Canada, V6T1Z4
[2] Dept. of Linguistics, University of British Columbia, 2613 West Mall, Vancouver, BC, Canada, V6T1Z4
[3] Haskins Laboratories, 300 George St., New Haven, CT, USA 06511

1. INTRODUCTION

Articulatory synthesis research has long been dominated by frequency domain and concatenate sample-based speech synthesis techniques. While successful in some domains (e.g., voice-based databases), these techniques still cannot produce natural looking and sounding speech from text for an arbitrary speaker. Natural looking and sounding speech technology is one of the next major milestones in voice-based interaction for natural user interfaces. Through a team of interdisciplinary international researchers, we have been steadily working towards creating the necessary platform to overcome basic problems in synthetic speech production. Our approach is three fold: 1. Create a Functional Reference for Anatomical Knowledge (FRANK) template model for use in studying articulatory synthesis, 2. Develop new methods for voice synthesis that couple to the platform, and 3. Use the biomechanically driven articulatory synthesizer to explore and develop theories of speech production. Our team draws from electrical, computer and mechanical engineering, linguistics, computer science, dentistry, radiology and surgery to assemble the needed components for advancing the state of the art.

2. The FRANK Model

We have created a modular biomechanical model of the head and neck referred to as a Functional Reference ANatomical Knowledge (FRANK) template [3]. It consists of multiple components governed by a hybrid physical simulation technique that combines multibody physics with 3D finite element analysis. Figure 1 shows an illustration of some of the components of FRANK. Components for FRANK have evolved as we have completed different experiments on speech [2][3][4][6][9][10][11][12][19][20][21][14], swallowing [15] and mastication [13] By combining the models from these studies and tailoring them to fit well together, we generate a generic 3D biomechanical model of the muscles and bones of the human head and neck suitable for modeling the biomechanics of speech.

FRANK is built using a biomechanical modeling toolkit called ArtiSynth [6][18] that has been specifically designed to support modeling of human anatomy and biomechanics. ArtiSynth is an open-source[a] platform that allows for mixed multibody and FEM simulation, supporting both contact and constraints. This enables efficient simulation of a large number of connected dynamic hard and soft tissues such as those involved in the upper airway.

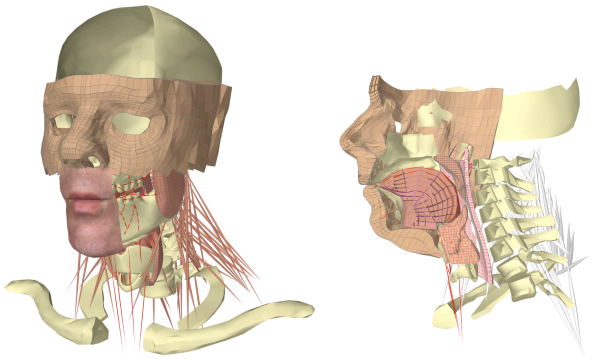


Figure 1: Overview of FRANK: oblique view showing external structures and midsagittal view showing some of the FEM soft tissues. FRANK includes an airway surface mesh connecting to the tissue models to provide an air-tight tube suitable for acoustic simulation.

When used in an inverse simulation setting, kinematic data can drive FRANK, as the model attempts to estimate the muscle activations needed to control actions when speaking. The template model uses a combination of FEM, rigid body, and spring-like components, allowing its functional resolution to be tailored to the needs of specific applications by trading off between speed, complexity and accuracy. By making FRANK modular, FEM elements can be introduced where high fidelity is required, while simpler, reduced degree-of-freedom models can be used where low-fidelity approaches provide sufficient scaffolding or accuracy, such as with line based muscles or rigid body bone models.

3. Acoustic Simulation

FRANK supports modeling of the 3D biomechanics of the upper airway. To be consistent with our approach to modular methods for simulation, we are developing multiple synthesis modules to combine with FRANK. Currently, we support 1D source filter models [17] with calculations of the acoustic transfer function of the vocal tract using JASS [23]. For the glottis, we are comparing the classic two mass model [16] with higher degree of freedom models of the vocal folds [22][5]. We are also investigating a 2D finite difference approach [24] based on [1]. Thus far, our comparison illustrates how mapping the 3d structure of the

[a] ArtiSynth is available for research use at http://www.artisynth.org.

vocal tract to lower dimensional representations for acoustic simulation results in poor estimates of frequency response. However, low dimensional representations of frequency response can be tuned to give good performance; though, the connection to physical geometry is compromised.

4. Speech Research Using FRANK

Model simulation using ArtiSynth coupled with speech production data show that the body offers only a small inventory of reliable, biomechanically robust actions for speech sounds, and that these postures have special biomechanical properties, whether viewed at the lips, the tongue, the larynx, the oropharyngeal isthmus, or the velopharyngeal port [2][9][10][12][20]. These findings align with research in neurophysiology and computation of motor control in which movement is built upon inventories of semi-closed neuromuscular structures, or "modules". We have developed a modular theory for speech (e.g., [8][7]) whereby these modules emerge through use as part of a learner's strategy to optimize the biomechanics of speech production, building on phylogenetically encoded motor structures [19]. This model provides robust and highly predictive results with no brain, no experience, and no anatomically pre-defined body parts/articulators (e.g., "lips", "jaw", etc.). The result is a theory of embodied phonetics built on an inventory of emergent, highly specialized and reliable functionally defined body structures that are harnessed for communication, each of which is used to serve a specific phonetic function (e.g., lip rounding, vocal fold approximation).

5. DISCUSSION AND CONCLUSIONS

Our large interdisciplinary team continues to create a platform for speech research using a 3D biomechanical articulatory speech synthesizer. By including team members with expertise in all areas related to speech we can improve each aspect and corroborate results using assumptions made to help capture the complexity of speech production. To date, our FRANK model incorporates over seven years of different modeling efforts and improvements of physics simulation techniques. As the effort is open source, we invite researchers to participate in this effort to continually add modules and improvements to the platform.

REFERENCES

[1] Allen A. and Raghuvanshi, (2015) N., Aerophones in Flatland: Interactive Wave Simulation of Wind Instruments, ACM ToGS (SIGGRAPH), 34(4).

[2] Anderson, P., Fels, S., Stavness, I. and Gick, B. (2016). Intrinsic and extrinsic portions of soft palate muscles in VP and oropharyngeal constriction: A 3D modeling study. *Can. Ac*.

[3] Anderson, P., Harandi, N. M., Moisik, S., Stavness, I. and Fels, S. (2015). A comprehensive 3D biomechanically-driven vocal tract model including inverse dynamics for speech research. *Interspeech 2015*. Dresden, Germany.

[4] Derrick, D., Stavness, I. & Gick. B.(2015).Two phonological segments, one motor event: Evidence for speech-motor disparity from Eng flap production. *J. Ac. Soc. Am.* 137, 1493-1502.

[5] Fariborz, A., DBerry, D. and Titze, I. (2000). A finite-element model of vocal-fold vibration, *JASA* 108.6: 3003-3012.

[6] Fels, S., Gick, B., Jaeger, C., Vogt, F. and Wilson, I. (2003). User-centered design for an open source 3-D articulatory synthesizer. In M. J. Solé, D. Recasens & J. Romero (eds.) *Proc. XVth Int. Congr. Phonet. Sci.*, Barcelona, Spain. 179-184.

[7] Gick, B and Stavness, I. (2013). Modularizing speech. *Front. Psych.: Cog. Sci*. 4, 977.

[8] Gick, B. (2016). Ecologizing dimensionality: prospects for a modular theory of speech production. *Ecol. Psy*. 28(3), 176-181.

[9] Gick, B., Allen, B., Stavness, I. (in press) Speaking tongues are actively braced. *J. Speech, Lang. & Hear. Res*.

[10] Gick, B., Anderson, P., Chen, H., Chiu, C., Kwon, H. B., Stavness, I., Tsou, S., Fels, S. (2014). Speech function of the oropharyngeal isthmus: a modelling study. *Comp. Meth. in Biomech. and Biomed. Eng.: Imag. & Visualization*, 2, 217-222.

[11] Gick, B., Stavness, I. and Chiu, C. (2013). Coarticulation in a whole event model of speech production. Proc *JASA* 19. 060207, 4.

[12] Gick, B., Stavness, I., Chiu, C. and Fels, S. (2011). Categorical variation in lip posture is determined by quantal biomechanical-articulatory relations. Can. Acoust. 39. 178-179.

[13] Hannam, A.G., Stavness, I., Lloyd, J.E., Fels, S.S., Curtis, D. and Miller, A. (2010). A comparison of simulated jaw dynamics in models of segmental mandibular resection versus resection with alloplastic reconstruction, JProsthetic Dentistry, 104(3):191-8.

[14] Harandi, N., Woo, J., Stavness, I., Stone, M., Fels, S. and Abugharbieh, R. (2015). Subject-Specific Biomechanical Modelling of the Oropharynx: Towards Speech Production, May:1-11, CMBBE:I&V.

[15] Ho, A., Affoo, R., Nicosia, M., Inamoto, Y., Eichii, S, Green S., and Fels, S. (2016). Computer Simulation of a Liquid Bolus for Studying Hyposalivation from Dynamic 320-slice CT images, Proc. of the Dysphagia Research Society (DRS2016), Feb 25-27.

[16] Ishizaka K and Flanagan, JL (1972). "Synthesis of voiced sounds from a two-mass model of the vocal cords," *Bell system technical journal*, vol. 51, no. 6, pp. 1233–1268.

[17] Kelly JL. and Lochbaum, CC, (1962). "Speech synthesis," in *Proc. Fourth Int. Congr. Acoust.*, Paper G42, pp. 1-4.

[18] Lloyd, J. E., Stavness, I. and Fels, S. (2012). Artisynth: A fast interactive biomechanical modeling toolkit... In Soft tissue biomech modeling for comp assisted surgery:355-394. Springer.

[19] Mayer, C., Roewer-Despres, F., Stavness, I. and Gick, B. (in press). Does swallowing bootstrap speech learning? *Can. Ac.*

[20] Moisik, S. and Gick, B. (2013). The quantal larynx revisited. *J. Ac. Soc. Am. – Proc. Meet. Acoust.* 19. 060163, 8 pp.

[21] Stavness, I., Gick, B., Derrick, D. and Fels, S. (2012). Biomechanical modeling of English /r/ variants.*JASA*.131:355-360.

[22] Story, B, and Titze. I. (1995). Voice simulation with a body-cover model of the vocal folds, *JASA* 97.2:1249-1260.

[23] van den Doel K. and U. Ascher, U., (2008) "Real-time numerical solution of webster's equation on a nonuniform grid," IEEE Trans. ASLP,16(6):1163–1172.

[24] Zappi, V. and Vasudevan, A. and Fels, S., Towards real-time two-dimensional wave propagation for articulatory speech synthesis, JASA: 139: 2010-2010 (2016).

ACKNOWLEDGEMENTS

The work is funded by NSERC, CIHR (CPG-121024), Autodesk Research and nVidia. We thank: ArtiSynth team, Oral, Pharyngeal and Laryngeal modeling team, Swallowing Research Group, Speech Synthesis team, and the Parametric Human Project.

ACOUSTICAL CONSULTING: ENVIRONMENTAL - CONSULTING : BRUIT ENVIRONNEMENTAL

Acoustical Verification Testing Of Ground Run-Up Enclosure At Vancouver International Airport
Mark Bliss, Mark Cheng, Rachel Min, Ron Reeves, Ryan McMullan 24

A Review Of The Bc Wind Power Project Acoustic Assessment Guidelines
Brigette Martin, Mark Bliss 26

Noise Levels From Heavily Travelled Roads For Use With Environmental Noise Regulations In British Columbia And Alberta
Shira Daltrop, Andrew C. Faszer, Victor W. Young 28

Meaningful Measurements To Assess Transformer Noise
Henk De Haan, Virgini Senden 30

Comparison Between Modelled And Measured Sound Levels For A Southern Alberta Wind Farm
Henk De Haan, Virgini Senden 32

Monitoring Of Road Traffic Sound Levels
Bill Gastmeier, Sheeba Paul 34

Comparison Of Air Craft Noise Exposure In Between Nef Contour Prediction And 24-Hour Measurement Results
Eric Gu, Mark Bliss 36

Modular Expansion Joint Noise In B.c.
Duane Ernest Marriner, Clair Wakefield 38

Effects Of Traffic Density On Measured Sound Levels – A Case Study
Ian Matthew, Anthony Amarra 40

Considerations Regarding An Acoustic Criterion For Wind Turbine Acceptability
William Keith Gregory Palmer 42

Community Noise Investigation Of A ”low Level Thrumming” Attributed To The Operation Of Mine
Andrew Williamson 44

ACOUSTICAL VERIFICATION TESTING OF GROUND RUN-UP ENCLOSURE AT VANCOUVER INTERNATIONAL AIRPORT

M. Bliss*[1], M. Cheng†[2], R. Min‡[2], R. Reeves^[3], R. McMullan#[3]
[1]BKL Consultants Ltd., Vancouver BC
[2]Vancouver Airport Authority, Richmond BC
[3]URS Aviation, California USA

1 Introduction

The Vancouver Airport Authority (Airport Authority) manages the Vancouver International Airport (YVR) on behalf of Transport Canada under a long-term ground lease. The Airport Authority is responsible for noise management activities, and has a comprehensive noise management program to meet this responsibility. In 2010, Airport Authority approved a capital project to construct a ground run-up enclosure (GRE) to further reduce noise disturbances from run-ups, i.e. engine testing after maintenance.

A GRE is a three-sided structure into which aircraft are positioned while performing a run-up. Propeller aircraft are associated with the majority of run-ups at YVR, and the Airport Authority selected the dimensions of the GRE to safely accommodate power-in/power-out operations of the most common aircraft types expected to use the facility.

While a number of potential sites for the GRE on the airfield were identified, siting and orientating is a challenge due to the many factors that must be considered, and trade-offs between acoustical and non-acoustical factors often need to be made. In general, the main factors include: aerodynamic usability; acoustical performance; proximity and ease of access to maintenance facilities; airport zoning (height) restrictions; and interferences with radio and navigation signals.

To help with site selection, URS Aviation created a noise model using CadnaA® with a GRE meeting the general dimensional requirements and then completed prediction modeling using run-up noise profiles collected for the Saab SF 340 and Beechcraft 1900 aircraft. This resulted in selecting a site on the south side of the airport, off Apron I.

The design and construction of the GRE was then awarded to Blast Deflectors Inc. (BDI), of Reno Nevada, with a requirement to meet a specified arithmetic average insertion loss (IL) of 15 dBA at directions 60°-300° at a distance of 400 feet from the operating engine (0° is along the axis of the tested aircraft) to be determined based on measurements made at 60°, 90°, and 135° relative to the aircraft axis. The calculation of the acoustical performance was to be determined using the methodology outlined in the ANSI/ASA Standard Method S12.8-1998, *Methods for Determining the Insertion Loss of Outdoor Noise Barriers.*[1]

* bliss@bkl.ca
† mark_cheng@yvr.ca
‡ rachel_min@yvr.ca
^ ron.reeves@longbeach.gov
# ryan.mcmullan@longbeach.gov

2 Method

2.1 Testing Set Up

Construction of the GRE was completed in January 2012, and the acoustical verification testing was completed by staff from ESA and BKL Consultants Ltd. on January 17, 2012, between the hours of 9 pm and 12 am.

During the measurements, 1-second Leqs and ⅓-octave spectral data were captured using Type 1 Larson-Davis Laboratories (LDL) Model 824 sound level analyzers. Calibrations were performed prior to the measurements using a Brüel & Kjær (B&K) Type 4230 or LDL Model CAL200 acoustical calibrator. With the exception of the reference noise level measurement, all measurements were completed at a height of approximately 1.5 meters above the ground.

Due to the location of adjacent buildings and hangars, only two ground-level receiver measurement sites were suitable for the acceptance testing: 50° at 400’ from the aircraft (MP1) and 125° at 250’ from the aircraft (MP2). While not the most ideal, this was the best possible measurement set-up.

Measurements were also completed at a reference location (REF) 90° and approximately 60’ from the closest aircraft engine. The REF location was positioned at a height of approximately 55’ to minimize the effect of the GRE wall on measurements. See Figures 1 and 2 for the acceptance testing measurement locations relative to the aircraft inside and outside the GRE.

As shown in the figures, an acoustical barrier, approximately 8’ in height and treated with 4” thick open-face fiberglass absorptive material on the microphone side, was positioned behind the MP2 location to mitigate reflection noise from the neighboring buildings in close proximity.

Measurements were completed for three aircraft: Saab 340, Beechcraft B1900, and Falcon 900B. Each aircraft performed three full power one-minute run-ups inside and outside the GRE.

2.2 Atmospheric Conditions

Atmospheric conditions were recorded continuously using a Davis Portable Weather Station, positioned at the REF measurement location, in accordance with ANSI/ASA S12.8. The standard specifies that measurements are not valid for wind speeds in excess of 5 m/s, or for changes in

wind class between measurements of the same aircraft inside and outside the GRE.

Specifically, changes in the wind direction during the Beechcraft 1900 run-ups resulted in all measurements for this aircraft being excluded for use in the verification study. While some measured data for the Saab 340 and the Falcon 900B were also affected, enough unaffected data was collected to proceed with insertion loss calculations.

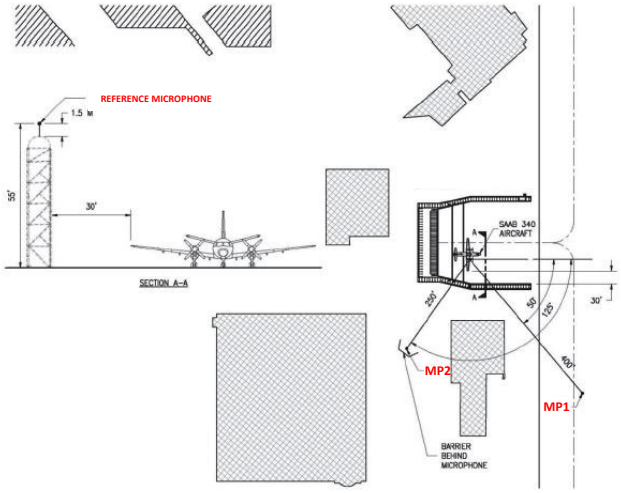


Figure 1: Measurement Locations for Aircraft Inside GRE

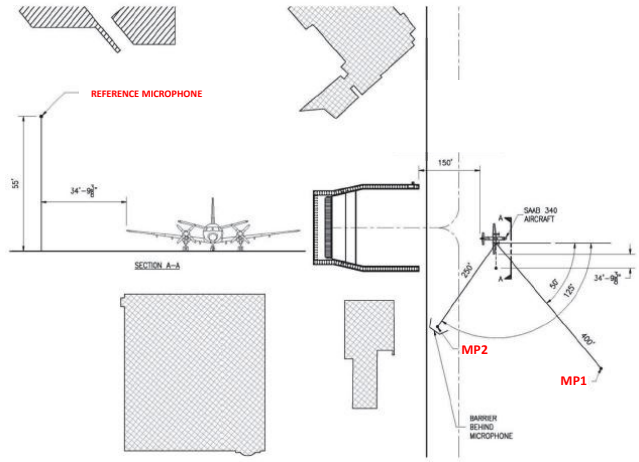


Figure 2: Measurement Locations for Aircraft Outside GRE

3 Results

Barrier insertion loss (IL) was calculated for each study aircraft using the following:

$$IL = (L_{AR} - L_A) - (L_{BR} - L_B)$$

Where L_{AR} is the average reference level (at REF) after barrier insertion (i.e. inside the GRE), L_A is the average receiver level at MP1 or MP2 after barrier insertion, L_{BR} is the average reference level before barrier insertion (i.e., outside GRE), and L_B is the average receiver level before barrier insertion. Measured levels and calculated IL for the two studied aircraft are summarized in Table 1.

Table 1: Measured sound levels and IL (dBA)

Aircraft	Site	L_{AR}	L_A	L_{BR}	L_B	IL
Saab SF 340	MP1	112	111	111	117	7
	MP2	112	77	111	98	22
Falcon 900B	MP1	108	93	108	109	16
	MP2	108	73	108	99	26

4 Discussion

The IL performance at MP1 likely suffered due to the reduced shielding provided by the south wall of the GRE: the angle of this measurement site (50°) relative to the aircraft heading was outside of the range specified for the effective performance of the GRE (60°-300° relative to the aircraft heading of 0°). The IL performance of the GRE at MP2 was likely improved due to the reduced distance between the barrier (GRE) and the site (250 feet instead of 400 feet specified). This reduced barrier-to-receiver distance produced an increased sound path length difference, increasing the barrier's noise attenuation performance. Although the measurement sites were not as specified or desired, they were the best available given the project site layout. Additionally, the acoustical shortcomings of site MP1 were largely negated by the acoustical benefits of site MP2.

Average IL performance of the GRE for the two ground-level measurement sites was 15 dB and 21 dB for the Saab SF 340 and Falcon 900B, respectively. As such, the IL performance of the GRE satisfied the minimum specified 15 dBA IL performance criteria.

5 Conclusion

Since its opening in January 2012, over 70% of the high power run-ups of propeller and business jet aircraft maintained on the south side of the airport have been performed inside the GRE. In addition, the GRE has become the primary location for run-ups performed during the noise sensitive hours between 10 pm and 7 am, with approximately 90% of high power run-ups performed during this time period performed inside the facility. There have been no reports of run-ups being adversely impacted due to adverse winds and the overall feedback from the maintenance operators has been very positive.

While the surrounding buildings at the site posed challenges to meet measurement requirements contained in the ANSI/ASA S12.8 standard, the GRE was observed to meet the acoustical insertion loss required in the design specification.

Acknowledgments

Thank you to Pacific Coastal Airlines and Pattison Airways for providing crew and aircraft for use during the testing.

References

[1] American National Standards Institute (ANSI). 1998. Methods for Determining the Insertion Loss of Outdoor Noise Barriers. Reference No. ANSI S12.8-1998. New York, Acoustical Society of America.

A REVIEW OF THE BRITISH COLUMBIA WIND POWER PROJECT ACOUSTIC ASSESSMENT GUIDELINES

B. Martin[*], EIT and M. Bliss[†], P.Eng.
BKL Consultants Ltd., #308-1200 Lynn Valley Road, North Vancouver, BC V7J 2A2

1 Introduction

The BC Ministry of Forests, Lands and Natural Resource Operations (FLNRO), Ministry of Energy and Mines and the Environmental Assessment Office (EAO) jointly published the guideline Best Practice for Wind Power Project Acoustic Assessment (BC Guideline) in 2012 [1]. This document makes recommendations in three areas: interpretation of the Wind Policy criteria, requirements of assessment reports and predictive modelling techniques.

As it would be difficult to compare the BC Guideline to every published windfarm noise assessment guideline worldwide, this paper provides a discussion and critique of the requirements presented in this guideline when contrasted to the Institute of Acoustics (IOA) document *A Good Practice Guide to the Application of ETSU-R-97 for the Assessment and Rating of Wind Turbine Noise* (IOA Guideline) published in 2013 [2]. The IOA Guideline has been referenced as an industry best practice guideline. A case study has been used to illustrate how predicted noise levels could vary when following the BC Guideline recommendations compared to the IOA Guideline.

2 Method

2.1 Literature Review

The purpose of the literature review is to compare the recommendations of the BC Guideline with other assessment frameworks that are used in other areas of the world.

The IOA Guideline has been used as a contrasting framework, as this document aims to summarize the findings of past research undertaken to present a current good practice in the application of the ETSU-R-97 assessment methodology for all wind turbine developments above 50 kW.

2.2 Case Study

To demonstrate the impact that different assessment methodologies can have on the predicted noise levels for a project, the requirements of these two different assessment methodologies have been applied to the same case study project.

[*] martin@bkl.ca
[†] bliss@bkl.ca

3 Results

3.1 Literature review

Some key differences between the BC Guideline and the IOA Guideline can be summarized as follows:

Correction Factors to Address the Limitations of Using the ISO 9613-2 Noise Prediction Methodology for Wind Farm Noise Predictions

The ISO 9613-2 [3] prediction methodology sets out the standard's limitations. Of relevance for wind farm projects is that ISO 9613-2 is designed for noise sources that are no higher than 30 metres above the ground and that receiver locations are no more than 1,000 metres from the noise source. Given the height of modern wind turbines is typically much greater than 30 metres (the turbine hub height used for the case study was 135 metres), this scenario rarely occurs for wind farm projects. While the BC Guideline recognizes that wind turbine projects do not fall within the stated limitations of the ISO 9613-2 prediction methodology, it does not specifically address how the ISO9613-2 standard should be applied to address these limitations.

The IOA Guideline addresses the ISO 9613-2 limitations by recommending appropriate maximum attenuation values for screening due to intervening terrain. According the IOA Guideline, attenuation due to screening should be limited to 2 dBA. In addition, the screening calculation must consider screening that would occur if the noise source was at rotor tip height rather than the hub height which would typically be used to determine the screening attenuation for a noise source.

Recommended Ground Absorption Coefficients

The BC Guideline and the IOA Guideline both discuss the risks of using soft ground for wind farm projects, given the limitation of ISO 9613-2 (discussed above), however the guidelines specify different maximum ground absorption coefficients. The BC Guideline specifies a maximum value of G=0.7 in contrast to the IOA Guideline maximum value of G=0.5. IOA Guideline recommendations appear to be based on the findings of studies which compared the measured noise level to predicted noise levels for as-built wind farms.

Wind Turbine Sound Power Levels

The BC Guideline states that the sound power level of the turbines is to be supplied by the turbine manufacturer and that uncertainty factors should be discussed.

The IOA Guideline provides more details on how sound power level data should be considered. Specifically, where G=0.5 is used for the model, the IOA states the following:

- Sound power levels that are determined based on considering the average of several individual IEC 61400-11 test results **(Declared sound power levels)** can be used directly.
- Sound power levels that are **warranted or specified manufacturer sound power level data** can be used provided a margin to account for uncertainty has been included. Where uncertainty has not been incorporated into the specified level, an uncertainty factor of +2 dBA should be added.
- Sound power levels that are determined based on an individual IEC 61400-11 test results **(Tested sound power level)** can be used when a correction reflecting the reported test uncertainty is incorporated. Where no data on uncertainty or test reports are available, a factor of +2 dBA should be added.

The benefits of the IOA approach mean that a clear view of the uncertainty can be presented. By applying the uncertainty value to the sound power level of the wind turbines, the risk of exceeding the criteria is reduced.

Consideration of Concave Ground Profiles for IOA Guideline

The IOA Guideline recommends that an additional correction factor of +3 dBA should be added for noise propagation "across a valley" i.e. a concave ground profile where the ground falls away significantly between the turbine and receiver when G=0.5 is used for the model (a correction of +1.5 dBA is recommended for G=0.0 models). A suggested criterion as to when to apply this correction is:
$h_m \geq 1.5 \times (\text{abs}\ (h_s - h_r)/2)$

This correction is recommended to account for the reduced ground effect expected and potential for additional reflection path that may exist when this specific topography occurs on site. This scenario is not specifically discussed in the BC Guideline.

3.2 Case Study

Based on the literature review, the key modelling differences between the BC Guideline and the IOA Guideline identified are summarized in Table 1.

Table 1: Key Differences in Recommended Prediction Methodology

Parameter	BC Guideline	IOA Guideline
Maximum ground absorption coefficient	0.7	0.5
Maximum attenuation due to intervening terrain	25 dBA	2 dBA

These two prediction methodologies were applied to a wind farm project to illustrate how predicted noise level could vary. The wind farm project consisted of 8 wind turbines, with the closest residential receivers approximately one kilometre from the nearest turbine. The wind turbines were proposed to be sited at an elevated level, with the residential receivers approximately 50 metres lower in elevation. The results for the receiver with the highest predicted noise level is shown in Table 2 below.

Table 2: Noise Levels for Case Study using the BC Guideline and IOA Guideline Prediction Methodologies

	BC Guideline	IOA Guideline
Highest predicted level	31.9 dBA	35.5 dBA

The average difference in noise level for receivers was an increase of +3.5 dBA when using the IOA Guideline compared to the BC Guideline. The differences ranged from 2.6 dBA to 6.4 dBA.

4 Conclusion

Reviewing and contrasting the BC Guideline and IOA Guideline for the wind farm noise assessment illustrate some differences in assessment methodologies. Some of these differences have to potential to result in significant variations in the noise prediction results for the same project. Further work should be undertaken to compare the predicted noise levels using the BC Guideline prediction methodology to noise levels experienced on site. In absence of this data, it is important to consider the risk that following the BC Guidelines may unknowingly result in noise that exceeds the BC Wind Policy noise criteria. The uncertainty of the noise predictions and possibility that turbines may have to be shut-down or operated in low-noise modes to enable compliance with the noise criteria should be considered as part the risk assessment procedure for wind farm projects in BC.

References

[1] Ministry of Forests, Lands and Natural Resource Operations the Ministry of Energy, Mines and Natural Gas and the Environmental Assessment Office. 2012. Best Practice for Wind Power Project Acoustic Assessment British Columbia. Victoria, Ministry of Forests, Lands and Natural Resource Operations the Ministry of Energy, Mines and Natural Gas and the Environmental Assessment Office.

[2] Institute of Acoustics. 2013. A Good Practice Guide to the Application of ETSU-R-97 for the Assessment and Rating of Wind Turbine Noise. Hertfordshire, Institute of Acoustics

[3] International Organisation for Standardization (ISO). 1996. Acoustics - Attenuation of Sound During Propagation Outdoors - Part 2: General Method of Calculation. Reference No. ISO 9613-2:1996. Geneva, International Organisation for Standardization.

NOISE LEVELS FROM HEAVILY TRAVELLED ROADS FOR USE WITH ENVIRONMENTAL NOISE REGULATIONS IN BRITISH COLUMBIA AND ALBERTA

Shira Daltrop *[1], Andrew C. Faszer †[2] and Victor W. Young ‡[2]
[1]Golder Associates Ltd., Suite 200 - 2920 Virtual Way, Vancouver, BC, Canada, V5M 0C4
[2] Golder Associates Ltd., 102, 2535 - 3rd Avenue S.E., Calgary, Alberta, Canada, T2A 7W5

1 Introduction

Noise Impact Assessments (NIAs) for oil and gas facilities in British Columbia (BC) are typically conducted in accordance with the BC Oil & Gas Commission's (BC OGC) Noise Control Best Practices Guideline [1]. NIAs for industrial facilities in Alberta are typically conducted in accordance with Alberta Utilities Commission (AUC) Rule 012 [2] or Alberta Energy Regulator (AER) Directive 038 [3]. The BC OGC Guideline, AUC Rule 012, and AER Directive 038 are very similar with respect to assessment methodology and compliance limits. All three documents require that noise be assessed cumulatively and, in particular, require that the contribution of heavily travelled roads be included when testing noise compliance for industrial facilities. The three documents have slightly different definitions of a heavily travelled road; Directive 038 has the most specific definition, requiring at least 10 vehicles per hour during the nighttime period. All three documents endorse the same desktop technique for estimating A-weighted Ambient Sound Levels (ASLs) at various distances from heavily travelled roads. This paper compares ASL values estimated using the regulatory desktop technique and noise levels calculated using a widely-accepted computer model of road traffic noise. Noise levels are compared for various receptor distances and various traffic levels.

2 Method

2.1 Computer Noise Model

A computer noise model was developed to predict traffic noise levels using commercially available software. There are a number of different traffic noise standards available; the Traffic Noise Model (TNM) developed by the Federal Highway Administration (FHWA) was selected for use in this study.

The Type 7810 Predictor® software (version 9.11), developed by Softnoise GMBH and distributed by Brüel & Kjær, implements TNM 2.5 and was used to develop predictive noise models for this study. The TNM standard characterizes road source emissions based on traffic volume, vehicle speed, road width, pavement type, and vehicle category distribution (i.e., car, heavy truck). The TNM standard accounts for noise attenuation related to ground cover

The computer model for this study predicted daytime and nighttime noise levels at distinct receptor distances and for a variety of hourly traffic volumes. As these predicted noise levels are from traffic only, the noise contribution from non-road sources must be added to be comparable to regulatory ASLs.

2.2 Ambient Sound Levels

The ASL represents background noise levels in the absence of energy resource and power generating facilities. The BC OGC Guideline, AUC Rule 012, and AER Directive 038 all use the same method for determining the relevant ASL for a given receptor. Based on research conducted by the Environment Council of Alberta, the average rural ASL in Alberta is approximately 35 dBA during the nighttime period. Daytime ASLs are 10 dB higher than nighttime ASLs and therefore the daytime rural ASL is 45 dBA. The guidelines indicate that these default ASL values are appropriate for low population areas and far from transportation infrastructure, but provide a specific method for adjusting ASL values for population density or transportation infrastructure. Table 1 presents ASLs for a number of different dwelling densities and distances to a road.

Table 1: Ambient Sound Levels for varying distances to heavily travelled roads and dwelling densities.

Distance (m)	Dwelling density per quarter section of land[1]					
	<9 dwellings		9-160 dwellings		>160 dwellings	
	Day	Night	Day	Night	Day	Night
<30	55	45	58	48	61	51
30-500	50	40	53	43	56	46
>500	45	35	48	38	51	41

[1] Quarter section of land is equivalent to an area with radius 451 m and the affected dwelling at the centre.

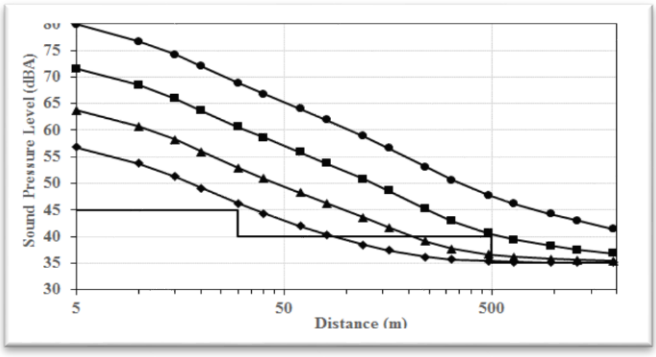
The ASLs prescribed for a receptor greater than 500 m from a road, which corresponds to the ASL for an area unaffected by road noise, were logarithmically added to the modelled traffic noise levels discussed in Section 2.1 to capture non-road related noise.

3 Results

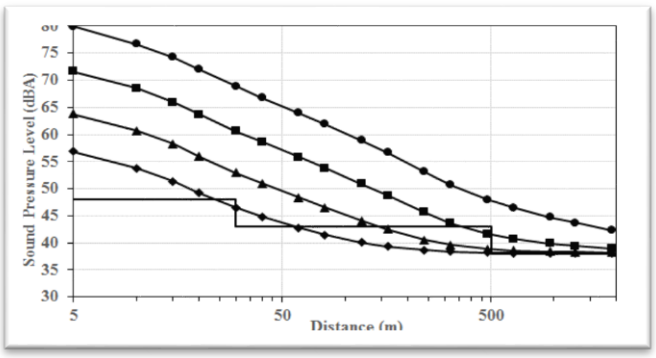
Predicted nighttime noise levels for three different dwelling densities and various traffic volumes are shown in Figure 1, alongside the relevant regulatory ASL. A vehicle speed of 80 km/hr and a vehicle category distribution of 80% cars and 20% heavy trucks was used for this modelling.

* sdaltrop@golder.com
† afaszer@golder.com
‡ vyoung@golder.com

a)



b)



c)

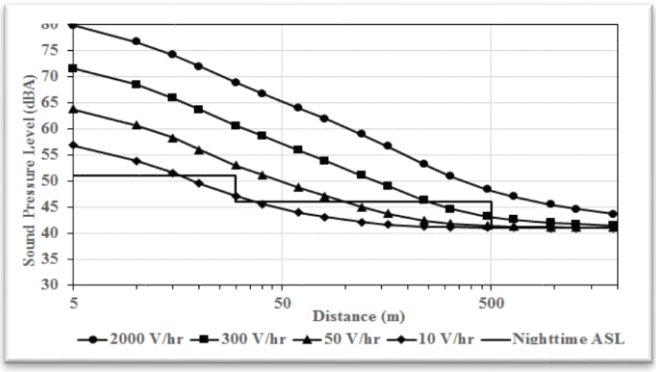


Figure 1: Predicted nighttime noise levels for various traffic volumes and the following dwelling densities per quarter section: a) <9 dwellings; b) 9-160 dwellings; and c) >160 dwellings.

ASLs are predicted to be within 5 dBA of noise levels from a road with 10 vehicles per hour (except very close to the road), consistent with Directive 038's definition of a heavily travelled road at night. The predicted sound pressure levels are higher than the prescribed ASLs for high traffic volumes and short distances to the road. As the dwelling density increases, the differences between the predicted noise levels and the ASLs become smaller.

Figure 1 is based on a vehicle speed of 80 km/hr; changing the vehicle speed has a substantial effect on noise levels near to the road. Increasing the speed to 110 km/hr increases the predicted noise level by an average of 3.7 dBA at receptors closer than 20 m to the road. Decreasing the speed to 50 km/hr decreases the noise level by an average of 3.5 dBA at receptors closer than 20 m to the road.

Traffic noise was predicted for daytime and nighttime periods for three road classes with high traffic volumes: expressway, major arterial, and minor arterial. These road types were quantified based on the parameters presented in Table 2. Table 2 also presents the predicted noise levels for a dwelling density of less than nine dwellings per quarter section.

Table 2: Modelled noise levels for different types of high-traffic roads assuming <9 dwellings per quarter section.

Parameter	Expressway		Major Arterial		Minor Arterial	
	Day	Night	Day	Night	Day	Night
Vehicle/hr	7200	1680	1800	420	840	196
Speed (km/hr)	80	80	60	60	50	50
% Heavy Truck	25	45	20	20	20	20
Distance (m)	Predicted Noise Levels (dBA)					
10	84	79	74	68	70	64
30	78	74	69	62	64	58
100	70	66	60	53	56	49
250	58	54	**51**	**44**	**49**	**41**
500	**54**	50	**49**	**41**	**47**	**39**
1000	51	46	**47**	**39**	**46**	**37**

Bold indicates prediction within 5 dBA of the regulatory ASL

4 Discussion

Noise modelling indicates that regulatory ASLs underestimate noise levels at receptors near roads with high traffic counts. Figure 1 indicates that nighttime ASLs are representative of roads with 10 vehicles per hour and with a vehicle speed of 80 km/hr; roads with higher traffic volumes and higher vehicle speeds result in higher noise levels that are underestimated by regulatory ASL values. Table 2 indicates that predicted noise levels for typical types of heavily travelled roads are greater than the regulatory ASL for almost all distances. At distances less than 250 m, predicted noise levels are greater than regulatory ASLs by more than 5 dBA.

5 Conclusion

Predicted noise levels for varying traffic volumes were compared with ASL values prescribed in BC and Alberta noise regulations. ASLs were found to be representative of a road with a low traffic count (10 cars per hour during the nighttime) travelling 80 km/hr. The ASLs are not representative of noise levels from roads with higher traffic counts or higher vehicle speeds; both of these factors should be taken into account when establishing existing conditions as part of regulatory NIAs.

References

[1] BC OGC. Noise Control Best Practices Guideline. 2009.

[2] AUC. Rule 012: Noise Control. 2013.

[3] AER. Directive 038: Noise Control. 2007.

MEANINGFUL MEASUREMENTS TO ASSESS POWER TRANSFORMER NOISE

Henk de Haan*[1] and Virgini Senden †[2]
dBA Noise Consultants Ltd, RR1 Site 14 Box 55, Okotoks, AB (403) 836 8806

1 Introduction

Electric power transformers provide a link between the power transmission system and the distribution system. They transform the high voltage power (e.g. 138 kV) from the transmission system to the lower voltage (e.g. 25 kV) of the distribution system. These transformers are situated where electric power is used – frequently, close to people. Due to their hum and the noise of cooling fans they may cause intrusive noise [1] levels. The noise emitted by transformers can be characterized as continuous noise.

Utilities are frequently required to consider transformer noise when planning or expanding a substation, when residential developments are planned near existing substations. Standard CAN/CSA-C88-M90 [1] is typically used as a reference document when selecting a transformer. This standard defines the maximum acceptable average noise level in "decibels" for transformers of a given capacity and construction. Noise levels are to be determined in accordance with Standard "IEEE[2] C57.12.90-2010 [2]. That standard applies to measurements around transformers in a laboratory or factory setting or transformers installed in a substation. It produces a single number sound power level (PWL) and includes an elaborate measurement procedure in the near sound field around a transformer.

This paper aims to compare the measurement procedure and results according to IEEE standard C57.12.90-2010 to other relevant standards such as ISO 3744 [3] or measurements at some distance, assuming the transformer as a point source.

2 Sources and Frequencies of Transformer Noise

Transformer hum is mainly caused by vibrations in the core of the transformer (caused by magnetostriction) and, typically to a lesser extent by vibrations in the windings surrounding the core, caused by electromagnetic forces [4].

If the core is poorly supported, or in very silent transformers, the noise contribution from the windings may become more significant. In North America, the fundamental frequency for core noise is 120 Hz and can be recognizable for observers nearby and sometimes is tonal according to Standard ISO 1996-1 2003 [5] or ANSI S12.9-2005 part 4 [6]. Tonal noise can be perceived as more intrusive and can be penalized according to the standard. The noise from cooling fans is typically broadband and needs to be added to the core noise. The increase in the average noise level due to the cooling fans from a transformer can be approximately 3 - 5 dB.

Since the fans are located to the side of the tank, the tank may act as a noise barrier for fan noise, especially if the radiators are placed on one side of the transformer. Taking advantage of this barrier effect when positioning a transformer may reduce noise levels for nearby residents, or prevent the need to install low-noise cooling fans on the transformer.

3 Why Deviate from Standard C57.12.90-2010?

Standard C57.12.90-2010 may not be the most suitable to follow for noise measurements around installed transformers when assessing environmental noise impacts for a number of reasons:

A large number of individual measurements around and close to a transformer are prescribed. Individual readings are collected at 1 m horizontal intervals around the transformer. If the height of the tank or enclosure is 2.4 m or more, measurements are conducted at 1/3 and 2/3 of the tank or enclosure height. Otherwise, measurements are conducted at 1/2 height of tank and enclosure. See Figure 1 for an illustration of the measurement positions.

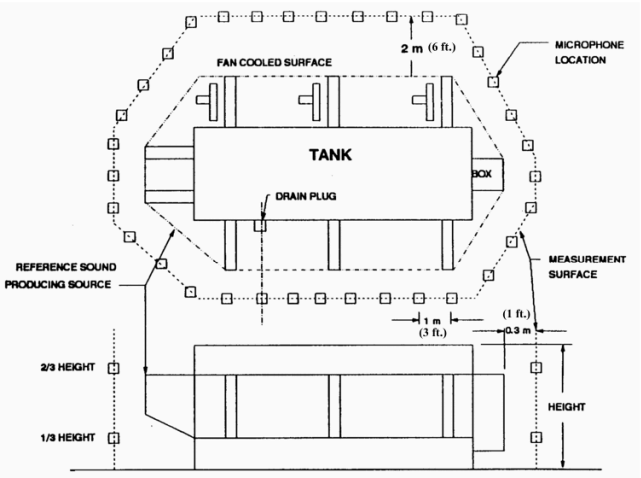


Figure 1. Measurement Positions According to IEEE C57.12.90

The prescribed procedure can easily result in hundreds of individual measurements. Assuming 30 seconds are required to collect each sample, the total uninterrupted measurement time is approximately 2 hours. It would be much quicker to allow for the microphone to be traversing over the measurement surface, resulting in an average sound pressure reading for that surface. This well-known procedure is allowed according to a more general standard

* henk@dbanoise.com
† virgini@dbanoise.com

[1] Noise is defined as unwanted sound.

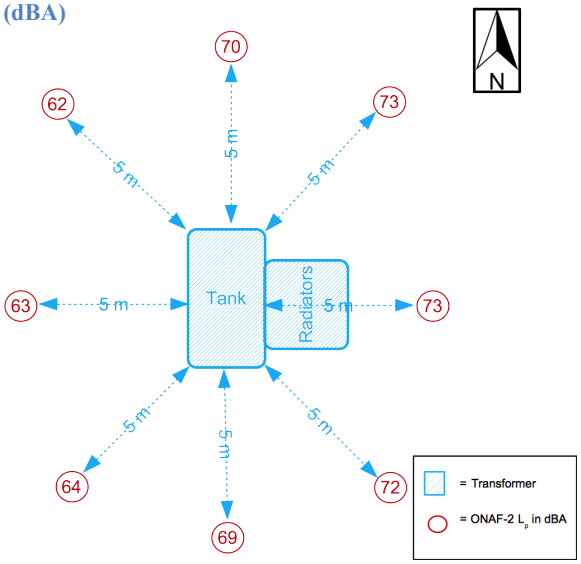
[2] The Institute of Electrical and Electronics Engineers

ISO 3744, frequently applied in noise measurements for all kinds of machinery.

The collected broadband or spectral data is energetically averaged to a single broadband or spectral number, representing the average sound pressure L_p around the transformer. This is the sound pressure L_p that standard CAN/CSA-C88-M90 refers to. From that average sound pressure, the average sound power is calculated without taking directivity into account. Directivity however may affect the noise impact. Standard ISO 3744 contains a procedure to assess directivity by comparing the measured sound pressure to the average sound pressure. It would also be possible to conduct measurements in the far sound field around a transformer at fixed distances (e.g. a circle at a fixed distance, with 8 equidistant measurements at a 45° spacing) to assess directivity. This is the preferred method. Practical considerations are background noise caused by other equipment in the substation and obstacles that may be challenging to overcome however. Under those circumstances, measurements according to ISO 3744 could be considered.

Measurements are conducted at a distance of 0.3 m from a reference surface around the transformer, except when cooling fans are running. If fans are running, the distance is 2 m. from any portion of radiators, fans etc. For frequencies up to approximately 500 to 1,000 Hz this is in the near sound field around the transformer. The large number of samples can be expected to compensate for potential deviations from the average sound pressure caused by local pressure variations, especially in low and mid frequency bands. However, considering transformer hum (120 Hz) and that the noise is experienced in the far sound field, it would be much simpler to conduct measurements also in the far field near the transformer. For a frequency of 120 Hz, this distance can be calculated to be up to approximately 4 m [7]. A very practical rule of thumb is that in order to characterize a noise source as a point source, the distance to the source should be at least double the maximum dimension of that source. See Figure 2 for an illustration of this approach. Measured sound pressure levels in dBA are included in Figure 2 for ONAF-2[3] operating conditions. These results illustrate the directivity around a transformer as a result of the noise barrier function of the tank.

Figure 2: Far Field Measurement Positions and Results (dBA)



The required operating conditions according to [2] are that the transformer is energized at the rated voltage and frequency with no load and the tap changer at the principal tap. The "no-load" condition is typically not achieved when measuring operating transformers in a substation, and in such a scenario the tap changer may be at a different setting, depending on grid requirements. "No-load" indicates that the transformer windings do not produce noise. This may prevent a straight comparison between an operating transformer and factory-tested data for the same transformer. It should be noted however that winding noise is typically marginal compared to core noise.

4 Conclusion

Summarized, for operational transformers the procedure according to C57.12.90 is not preferred as it is elaborate, does not provide directional information, is measured in the near field and the load setting requirements are typically not achieved in an operational substation. Practical alternatives are equidistant and evenly spaced far field noise measurements or near-field measurements according to ISO 3744-2010.

References

[1] CAN/CSA-C88-M90 (reaffirmed 2014), Power Transformers and Reactors

[2] IEEE C57.12.90-2010 [2], Standard Test Code for Liquid-immersed Distribution, Power and Regulating Transformers

[3] ISO 3744-2010 Acoustics — Determination of sound power levels and sound energy levels of noise sources using sound pressure — Engineering methods for an essentially free field over a reflecting plane

[4] Handbook of Acoustical Measurements and Noise Control, 3rd edition, Cyril M. Harris, reprint by the Acoustical Society of America – 1998, Woodbury USA

[5]ISO 1996-1 2003, Acoustics, Description, measurement and assessment of environmental noise — Part 1: Basic quantities and assessment and procedures

[6] ANSI S12.9-2005/Part 4, Quantities and Procedures for Description and Measurement of Environmental Sound – Part 4: Noise Assessment and Prediction of Long-term Community Response

[7] Handbook of Noise and Vibration Control, 2007, Malcolm J. Crocker, Published by John Wiley and Sons, Inc. Hoboken, New Jersey USA

[3] Oil Natural Air Forced. Typically, all cooling fans are running under ONAF-2 operating conditions.

COMPARISON BETWEEN MODELLED AND MEASURED SOUND LEVELS FOR A SOUTHERN ALBERTA WIND FARM

Henk de Haan* **and Virgini Senden†**
dBA Noise Consultants Ltd, RR1, Site 14, Box 55 Okotoks AB T1S 1A1

1 Introduction

As an approval condition by the Alberta Utilities Commission (AUC) for a 23 wind turbine generator (WTG) expansion with substation a Comprehensive Sound Level Survey (CSL) was conducted. Three receptor locations were included, where the noise[1] impact was predicted to be within 1 dB of the nighttime (10 PM – 07 AM) threshold of 40 dBA. A CSL with a duration of 38 days was conducted. This article will summarize some of the lessons learnt.

The wind farm (WF) is located in Southern Alberta on an elevated plain. Other WFs are located in the vicinity. The area is mostly used for cattle grazing. There are no major highways or railroads in the area, nor are there frequent aircraft flyovers; roads serve mostly local traffic. The WTGs installed in the expansion are Vestas V90-3 MW, with a hub height of 80 m and a rotor diameter of 90 m.

The power output and noise production from a WTG are correlated. At the kick-in wind speed, the rotor will start to turn and produce both power and noise. Both will increase up to a certain wind speed.

Noise propagation is related to multiple atmospheric conditions. Wind direction is a major influence; being downwind from a noise source typically results in (close to) the highest noise levels for a specific operational setting.

2 Method

2.1 Modelled Noise Impact

Noise Impact According to AUC Rule 012.

In Alberta, noise from power-related facilities is regulated by the AUC. The AUC published Rule 012, *Noise Control* (Rule 012) [1] that prescribes thresholds, assessment methods and reporting requirements. Summarized, the noise impact is defined as the cumulative total of the (assumed) ambient sound level and the noise from all energy-related facilities (Power and Oil & Gas) that might affect noise levels at a noise sensitive receptor. This cumulative noise impact should not exceed a Permissible Sound Level (PSL) under summertime conditions. For rural Alberta, the nighttime (10 PM – 07 AM) PSL is typically 40 dBA, and the daytime (7 AM – 10 PM) PSL is 50 dBA. The assumed ambient sound level is 5 dB less than the PSL, thus 35 dBA for the nighttime period and 45 dBA for the daytime period. The noise impact from a proposed facility needs to be assessed under representative conditions and according to international standards such as ISO 9613 [2]. That standard was applied in this assessment. "Representative" was interpreted as requiring that all WTGs should be modelled as simultaneously operating at their maximum operational settings under summertime atmospheric conditions, and propagation is calculated as all receptors being downwind from all modelled noise sources.

The predicted nighttime noise impact at the receptors considered are included in Table 1. Also included are the downwind directions from the two closest WTGs towards each receptor.

Table 1: Predicted Nighttime Noise Impact (dBA)

Receptor	Ambient	Facilities	Noise Impact	Downwind Direction
1	35	37	39	NNW or SW
2	35	37	40	NNE or SE
3	35	38	40	E or SE

Comprehensive Sound Level Survey

Rule 012 also contains provisions for CSLs. Key requirements were that a CSL must be conducted under representative summertime conditions, and that at least three hours of valid data must be acquired in both the daytime and nighttime period. It is not explained in the Rule where this requirement is based upon. The current version of Rule 012 contains provisions allowing the use of statistical methods to assess whether or not sufficient data was collected for ambient noise measurements, conducted to assess wind noise at various wind speeds. Other jurisdictions include different requirements for the amount of data; to assess compliance, Ontario initially requires at least one hour of measurements, yielding 20 minutes of data to be included in the assessed noise level [3]. Only if this program indicates non-compliance a more extensive measurement program is requested. It is not stated in Rule 012 whether data should be collected in a consecutive period, or in the same 24-hour period. "Representative conditions" were understood to include operation of the closest or second closest WTGs for each receptor at or near their maximum operational setting, with the receptor being downwind from the closest WTGs. Based on historical weather data late July was selected as providing the best chance to experience the required wind directions. The program was to include a full week of measurements.

Typically, noise data (L_{eq}) is collected in one-minute intervals. Simultaneously weather data is collected and audio recordings are made. Data is analyzed for valid data that fall within the right weather and operating conditions and erroneous one-minute samples (e.g. people talking nearby, loud aircraft fly-overs) are removed from analysis.

* henk@dbanoise.com
† virgini@dbanoise.com

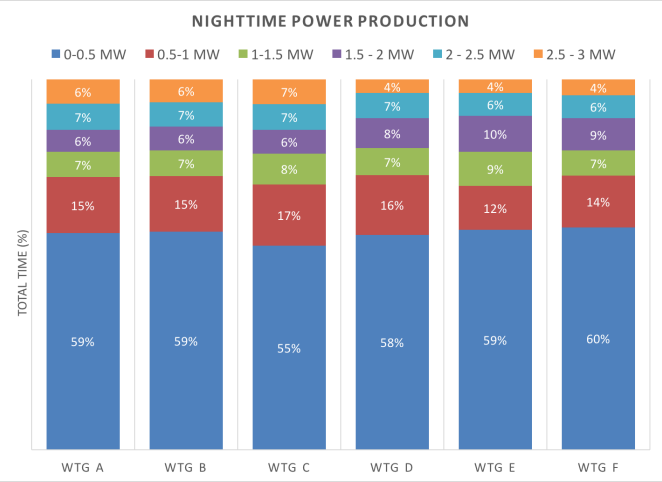
[1] Noise is defined as unwanted sound

The L_{eq} is calculated for the remaining samples for the valid duration only.

2.2 Program Duration

Analysis of the collected data after the first week revealed that insufficiently representative (wind direction and power output) data was available. The field program was therefore extended in to a total of 38 days. Analysis of the power output data of the WTGs over the full 38-day period revealed that the WTGs operated most of the time below their peak output. See Figure 1.

Figure 1 Nighttime Power Production in 500 kW Intervals



Power output between 2 kW and 3 kW during nighttime hours only occurred 10%-13% of the time; 55% - 60% of the time the WTGs have an output of less than 500 kW.

3 Results

After 38 days, not a single 24-hour period was available yielding 3 hours of valid data in the nighttime period or daytime period, but several multiple day periods were available that would yield sufficient data within each of those periods for either the closest or next closest WTG. The highest measured noise levels for the nighttime period are included in Table 2, together with the power output of the upwind WTG and compared to the calculated noise impact. The measured daytime noise impact is comparable within 2 dB of the nighttime noise impact.

The project was audible under downwind conditions for receptor 1, intermittently audible for receptor 2 and not audible at all for receptor 3. Results demonstrate compliance with PSL with a substantial margin. Measured results are well below the predicted noise impact.

Table 2: Comparison Measured Predicted Nighttime Noise Impact (dBA)

Receptor	Measured Noise Impact	Power Output (kW)	Predicted Noise Impact	Downwind from Closest or Next-closest WTG
1	35	1500 - 2250	39	Next Closest
2	35	50 - 1100	40	Next Closest
3	37	50 - 1500	40	Next Closest

4 Discussion

It turned out to be very challenging to simultaneously meet the conditions of conducting measurements downwind at a specified location and having the closest WTGs operate at or near maximum power output for at least three hours. Downwind conditions from either the closest or second closest were achieved between for 2% and 9% during nighttime hours. In future revisions of Rule 012, it could be considered to allow for measurements at an alternative and comparable location when testing for compliance. This would allow to take advantage of actual wind conditions instead of having to wait for a specific and maybe rare wind direction.

The program revealed that the WTG's only operated near maximum power levels (2,500 kW – 3000 kW) for approximately 5-6% of the time during the survey. It could be considered to take the typical wind resource of the season into account when assessing WTG noise; Rule 012 is aimed at summertime conditions.

It is unclear why a minimum of three hours of valid data is required. As an alternative a statistical assessment of the gathered data could be considered. Such a statistical method is already included in Rule 012 for ambient noise measurements with varying wind speeds in preparation for a C class adjustment to the PSL for WTGs, taking wind noise into account.

When predictions according to ISO 9613-1996 indicate compliance with a narrow margin, considering wind direction in prediction may provide more realistic results.

References

[1] AUC Rule 012, Noise Control, version valid from March 24, 2009 to February 22, 2010.

[2] ISO 9613-1996, Attenuation of Sound During Propagation Outdoors – Part 2: general method of Calculation.

[3] Compliance Protocol for Wind Turbine Noise, Guideline for Acoustic Assessment and Measurement, Ontario Ministry of the Environment.

MONITORING ROAD TRAFFIC SOUND LEVELS

William J. Gastmeier, MASc[†], PEng and Sheeba Paul, MEng, PEng[*]
HGC Engineering, 2000 Argentia Road, Plaza 1, Suite 203, Mississauga, Ontario, Canada L5N 1P7

1 Introduction and Background

In the 1980's the Ontario Ministry of the Environment, now the Ministry of the Environment and Climate Change (MOECC) in cooperation with the National Research Council (NRC) developed a set of assessment criteria and Acoustical Engineering analysis procedures to assist Municipalities in ensuring that sufficient mitigation of traffic noise is properly incorporated into new housing developments. These procedures were universally adopted by Municipalities across the Province and remain in use almost unchanged after 30 years.

Occasionally homeowners express complaints and dissatisfaction that road traffic is audible inside their homes and that it interferes with their enjoyment of their property and can disturb sleep. Since the original research supporting these procedures was partly based on sleep disturbance, and estimates of the percentage of persons who may be awakened or annoyed, it is of interest to Engineering Consultants, Municipalities, Land Developers and Home Builders to investigate these complaints.

This article describes one such complaint investigation and involves simultaneous monitoring of traffic sound levels inside and outside a multi-storey residential dwelling located beside an arterial roadway in a medium density row townhouse development. Analysis is conducted to confirm the accuracy and applicability of the Engineering prediction methods used to design building envelopes for residential dwellings located near busy roadways across the Province.

2 Criteria and Assessment Procedures

Guidelines for acceptable levels of road traffic noise impacting residential developments are given in MOECC Publication NPC-300[1] and are listed in Table I below. The values in Table I are energy equivalent (average) sound levels [L_{EQ}] in units of A-weighted decibels [dBA].

Table I: MOECC Road Traffic Noise Criteria (dBA)

Area	Daytime $L_{EQ\ (16\ hour)}$	Night-time $L_{EQ\ (8\ hour)}$
Outside Bedroom Windows	55 dBA	50 dBA
Inside Living/Dining Rooms	45 dBA	45 dBA
Inside Bedrooms	45 dBA	40 dBA

[†] bgastmeier@hgcengineering.com
[*] spaul@hgcengineering.com

Daytime refers to the period between 07:00 and 23:00, while nighttime refers to the period between 23:00 and 07:00.

Predictions of future road traffic sound levels use ORNAMENT, an analysis method implemented in STAMSON 4.1 software available from the MOECC.

A central air conditioning system as an alternative means of ventilation to open windows is required for dwellings where future nighttime sound levels outside bedroom windows are predicted to exceed 60 dBA. At that level of noise impact, which is common near busy roadways, building envelope components such as walls, windows and doors must be designed to achieve the indoor sound level criteria.

Building envelope components are designed using calculation methods were developed by the National Research Council (NRC)[3]. They are based on the predicted future sound levels at the building facades and the ratio of the area of the facade components to the floor area of the adjacent room.

3 Building Envelope Design

A partial site plan showing the location of the subject townhouse unit and its relationship to Busy Street is shown below.

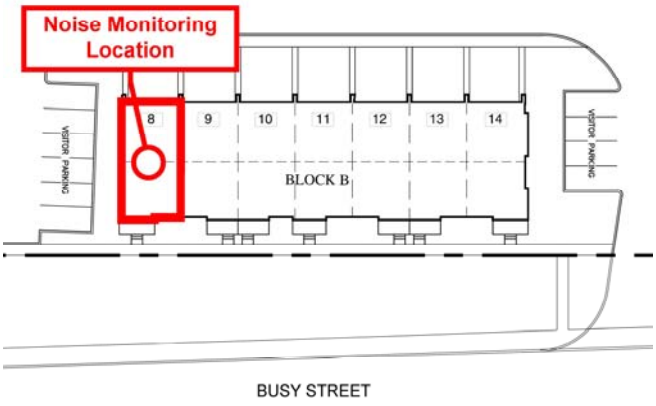


Figure 1: Site Plan

The results of the STAMSON predictions originally conducted for the Noise Impact Study required by the Municipality before Site Plan Approval indicated future daytime/nighttime sound levels of 68 dBA $L_{EQ(16\ hour)}$ daytime and 62 dBA $L_{EQ\ (8\ hour)}$ nighttime at the building façade of the

subject townhouse unit. Using the NRC calculations methods a glazing construction with an STC-30 rating was specified, considering that the exterior walls of the subject townhouse units would be constructed with a brick veneer with stone/EIFS accents.

4 Sound Level Monitoring

Automated sound level measurements were simultaneously conducted at four locations: outside the building façade at ground level, inside in the first floor den, inside in the second floor living room and inside in the third floor bedroom. All of the measurement locations had direct exposure to Busy Street. The Sound Level Meters are maintained in yearly laboratory calibration and were field calibrated before and after the measurements.

An elevation drawing of the row of townhouses showing the monitor locations is provided below.

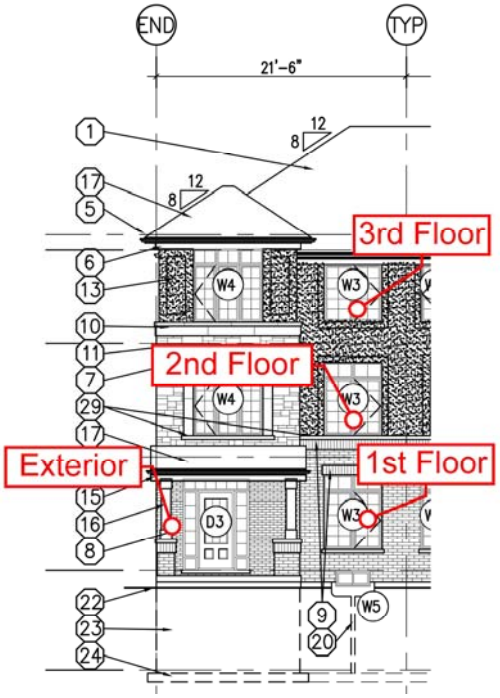


Figure 2: Measurement Locations

During the site visits it was confirmed that the appropriate glazing had been installed. Test data from the window manufacturer confirmed greater STC values than required.

The results of the monitoring are provided in the following figure and summarized in Table 2.

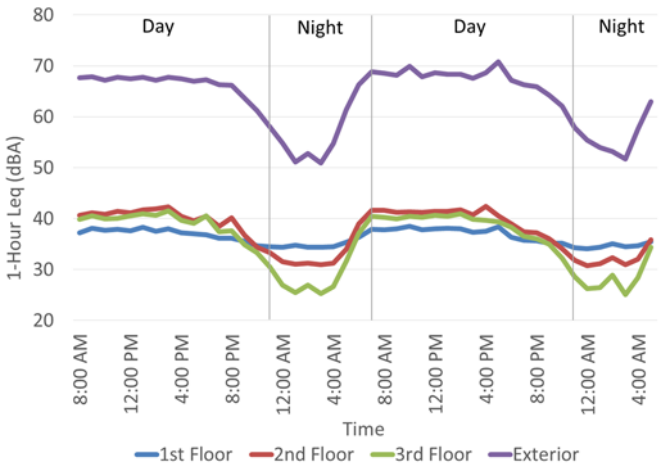


Figure 3: Measured Sound Levels, Hourly Leq, dBA)

Table 2: Measured Daytime $L_{EQ\ (16\ hour)}$ and Nighttime $L_{EQ\ (8\ hour)}$ Sound Levels

Monitor Location	Measured Sound Levels (Day/Night), dBA
Exterior	67 / 60
Inside 1st Floor Den	37 / 35
Inside 2nd Floor Living Room	40 / 34
Inside 3rd Floor Bedroom	40 / 31

5 Conclusion

These results confirm the accuracy of future sound level predictions using the ORNAMENT calculation method. They also confirm the applicability of the NRC calculation methods used to design the facades of residential buildings in terms of acoustical insulation and that they incorporate a reasonable degree of conservatism in terms of protecting the public and the interests of Municipalities and Home Builders alike.

Perhaps more interestingly, they remind us that assessment methodologies based on statistical distributions of annoyance in humans do not result in 100% satisfaction. You cannot make all the people happy all the time.

References

[1] NPC-300, "Environmental Noise Guideline Stationary and Transportation Sources – Approval and Planning"

[2] ORNAMENT "Ontario Road Noise Analysis Method for Environment and Transportation, Ontario Ministry of the Environment, October 1989

[3] National Research Council of Canada Building Research Note # 148 "Acoustic Insulation Factor: A Rating for the Insulation of Buildings against Outdoor Noise"

COMPARISON OF AIRCRAFT NOISE EXPOSURE FROM NEF CONTOURS AND FIELD MEASUREMENT RESULTS

Eric Gu, M.Eng, EIT* **and Mark Bliss, P.Eng+**
BKL Consultants Ltd., 308-1200 Lynn Valley Road, North Vancouver, BC V7J 2A2

1 Introduction

Noise due to by Vancouver International Airport (YVR) aircraft activities is of major concern for the public in the City of Richmond, British Columbia. Acoustical consultants commonly use the Noise Exposure Forecast (NEF) contours provided by Vancouver Airport Authority (YVRAA) to assess the impact of aircraft noise on new residential developments, which are regulated using the 24-hour equivalent sound level (L_{eq24}) metric. This is because it is quicker to rate a site's noise exposure by reviewing the NEF contours than by performing site noise measurements that are correlated with aircraft movements during that time.

The development of NEF contours is mandated by Transport Canada for planning purposes and are developed using the peak planning day of the forecast year, which, in this case, was 2015. The National Research Council of Canada (NRC) has previously estimated that the typical difference between the NEF and the corresponding 24-hour equivalent sound level is 32. [1]

BKL has performed 24-hour noise measurements in the City of Richmond and found that aircraft noise levels at those measurement locations tended to be significantly lower than would otherwise be assumed using the NRC estimate. Therefore, BKL has investigated the difference in aircraft noise exposure between the published NEF contours and noise measurement results.

2 Method

2.1 Selection of Noise Comparison Locations

YVR maintains a set of permanent Noise Monitoring Terminals locations (NMTs) for their long-term noise monitoring program. Data from these NMTs was made available to BKL by YVRAA for the purposes of this comparison. While aircraft noise is dominant at some of these locations, noise from local activities such as traffic is dominant at other locations. The first step of BKL's analysis focused on the selection of noise monitoring locations where aircraft noise is predominant in order to improve the accuracy of the comparison.

YVRAA publishes annual noise reports as part of their noise monitoring program. Two noise levels have been published in some of these reports: the equivalent total noise level and the equivalent aircraft noise level. The aircraft noise level is automatically calculated by the NMT system according to aircraft flyover (level and duration) characteristics.

BKL reviewed the past annual reports and selected the following best available locations as labelled in Figure 1.

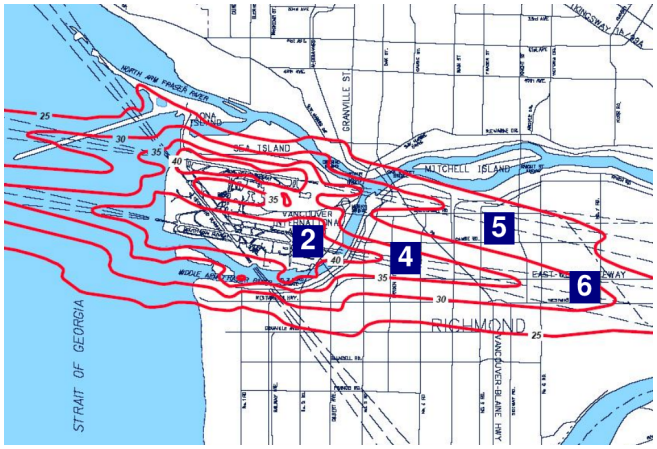


Figure 1: Selected NMT Site Locations where Aircraft Noise is Dominant

2.2 NMT Measurements

All of YVRAA's NMTs use Bruel & Kjaer Model 2250 Type 1 sound level meters. YVRAA provided BKL with daily aircraft noise monitoring data collected for the entire month of August 2015 at the NMT locations noted above, except at site 6 where data from August 2-5 was unavailable. The month of August was chosen since August is the busiest month of the year. BKL used the measured aircraft noise levels for the purposes of this comparison.

2.3 Predictions Using NEF Contours

NEF values were determined at each location using the YVR forecast 2015 NEF contours. The predicted aircraft noise level was estimated using the following equation:

$$L_{eq24,NEF} = NEF\ value + 32$$

3 Results

The summary of August 2015 measurement results at NMT sites 2, 4, 5 and 6 as well as the predicted $L_{eq24,NEF}$ values at these locations can be found in Table 1 below. The noise levels shown in Table 1 are the aircraft noise levels.

It is readily observed that the predicted aircraft noise levels using the NEF contours are higher than the measured values. The details of the differences can be found in Table 2.

* gu@bkl.ca
† bliss@bkl.ca

Table 1: Summary of August 2015 measurement results and predicted $L_{eq24,NEF}$ values at these measurement locations

NMT Site	August 2015 Measured Aircraft $L_{eq24,NMT}$		2015 NEF Contour Prediction	
	Average (dBA)	Maximum (dBA)	2015 NEF	$L_{eq24,NEF}$ (dBA)
2	61.4	68.2	> 40	> 72
4	57.4	62.8	35.0	67.0
5	54.8	57.4	27.7	59.7
6	53.3	56.4	32.2	64.2

Table 2: Summary of differences in August 2015 measurement results and predicted $L_{eq24,NEF}$ values at these measurement locations

NMT Site	Difference from Measured Average Value	Difference from Measured Maximum Value
2	>10.6	>3.8
4	9.6	4.2
5	4.9	2.3
6	10.9	7.8

The differences shown in Table 2 above are only for the average and maximum daily aircraft noise levels in the month of August 2015. Variations on a day-to-day basis can help visualize the range of differences. Figure 2 below shows the difference between measured daily aircraft noise levels and the NEF predicted levels at the NMT sites.

The error is defined as:

$$Error = L_{eq24,NEF} - L_{eq24,NMT}$$

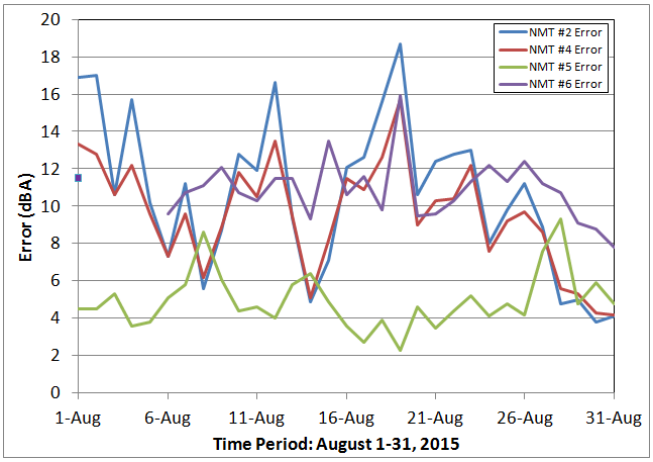


Figure 2: $L_{eq24,NEF}$ Error at NMT Locations During Peak Month

Data was unavailable from August 2 to August 5, 2015 at NMT site 6. From all of these four long-term NMT stations, aircraft noise predicted by NEF contours was most conservative at NMT site 5 and was least conservative at NMT site 2. The maximum error at NMT site 2 is 18.7 dBA while the minimum error at NMT site 5 is 2.3dBA. Therefore, the error of estimating aircraft noise levels by using NEF contours ranges from 2.3 dBA to 18.7 dBA during August 2015.

4 Discussion

NMT sites 2, 4, 5 and 6 are located in areas that have aircraft noise as a dominant noise source. The L_{eq24} noise levels predicted using the 2015 NEF contours plus 32 were more conservative in all cases, by at least 2.3 dBA.

One reason for the disparity is that the number of 2015 forecast aircraft movements was significantly higher than the number of actual movements. [2] Another is due to the accuracy of the software used to calculate NEF values. [1]

On one hand, a conservative estimate of aircraft L_{eq24} noise exposures would lead to better noise isolation to be installed at new residential developments that are affected by aircraft noise in the City of Richmond.

On the other hand, the conservative approach could increase construction costs significantly in some cases, resulting in higher housing costs. Returning to the least conservative case in BKL's analysis, the 2.3 dBA over-estimation with regards to aircraft noise levels can still be the difference between normal and thicker, laminated glazing: a significant cost increase.

5 Conclusion

Noise levels predicted using NEF contours tend to be significantly more conservative than 24-hour continuous site noise measurement results when assessing aircraft noise activities. When NEF predicted noise levels are used to assess aircraft noise for the design of new developments, it is likely that resulting improvement schemes are overly conservative. Therefore, actual site noise levels should be considered during the design of new developments. Professional acoustical consultants can establish a more realistic noise environment and recommend appropriate improvements to meet municipal requirements. This would require more initial effort but could result in significant overall development cost savings.

Acknowledgments

Special thanks are given to Mr. Mark Cheng at Vancouver Airport Authority for his kind support in providing the long-term noise measurement results. Without his help, this study would have been extremely difficult.

References

[1] Bradley, J.S. 1996. NEF Validation Study: (3) Final Report. NRC Contract Report to Transport Canada, A1506.6 (1996). Ottawa, National Research Council of Canada.

[2] Vancouver Airport Authority. 2016. 2015 Aeronautical Noise Management Report. Richmond, Vancouver Airport Authority.

MODULAR EXPANSION JOINT NOISE IN B.C.

Duane Marriner *[1] and Clair Wakefield †[1]
[1]RWDI AIR Inc./Wakefield Acoustics Ltd., #301-2250 Oak Bay Ave., Victoria, B.C., V8R 1G5

1 Introduction

Greater Vancouver is geographically divided into several municipalities by the Fraser River arms. Bridges have been constructed across the arms and estuaries that employ modular expansion joints (MEJs) capable of rotation and translation within the 1 m range or more. A key element of the basic MEJ design is an array of I-beams whose upper flanges provide the supporting roadway surface over the joint. The basic design however leads to tire/joint interaction noise that can be a concern to neighboring communities. This study defines the tire/joint interaction noise of an MEJ design through a measurement program.

The measurement program involved a new abutment and a new pier MEJ. Specifications of the MEJs, bridge deck, pavement type and the setting of the bridge/community are discussed in [1]. Noise control is presented in [2].

2 Method

2.1 Monitoring Locations and Instrumentation

Microphones were mounted on tripods at a height of 1.7 m above the sidewalk of the bridge deck at longitudinal offsets measured from the transverse centerline of an expansion joint. The offset is taken to be a positive distance in the direction of traffic flow and negative in the direction opposite the flow. For example, for measurements taken at a longitudinal offset of -15 m, the microphone would be 15 m *behind* the vehicles as they passed over the MEJ. A tripod-mounted microphone was also located 1.7 m above local ground level at a distant residential location of 585 m away at an elevation of 7 m above sea level.

Instrumentation included a Larson-Davis Model 820 Community Noise Analyzer (CNA) and Larson-Davis Model 2800 Real Time Analyzer (RTA).

The CNA was set to log the L_{maxF} and L_{peak} in units of dBA at one-second resolution. The Larson-Davis RTA was used in Linear Single (LS) mode to store the one-third octave spectrum of the true RMS sound pressure averaged over the measurement interval. In LS mode, the RTA could alternatively be set to store the maximum spectrum (mxSpec). The mxSpec is the one-third octave band spectrum with the highest energetic sum of all those sampled within the measurement interval.

3 Results

3.1 Subjective Observations

Subjective observations including effects of tire type, vehicle weight and speed on tire/joint emission levels as well as the effect of overall width of the joint are discussed in [1].

3.2 Short Range Average Maximum Noise Levels - General Traffic

The maximum levels (L_{maxF}) of tire/joint interactions were logged with a CNA at the abutment joint from vehicles chosen "on the fly" from the general traffic flow. Vehicles were selected that would result in an MEJ emission with a high signal-to-noise ratio.

The L_{maxF}s of the highest quality events were sorted into vehicle weight categories and averaged. The results are presented in Table 1. Reproducibility may be limited [3].

Table 1: Average Maximum Levels at a -15 m Offset from Abutment Joint - General Traffic, August 2009

Vehicle Type	No. of Events	Avg. L_{maxF} (dBA)	Std. Error (dBA)
Full-size Car	2	87.1	1.0
Heavy Pickup	12	86.6	1.3
Mid-size Car	16	84.2	1.1
Sports Utility	15	83.0	1.2
Compact Car	6	79.4	1.1

3.3 One-Third Octave Band Spectra

One-third octave band spectra of tire/joint interaction noise of various single vehicles selected from the general traffic are shown in Figure 1. The microphone was positioned at an offset of -15 m and the RTA was set to LS mode.

From Figure 1, in eight of eleven cases, the dominant frequency was found within the one-third octave band centered on 630 Hz. In three cases, the 800 Hz band was dominant.

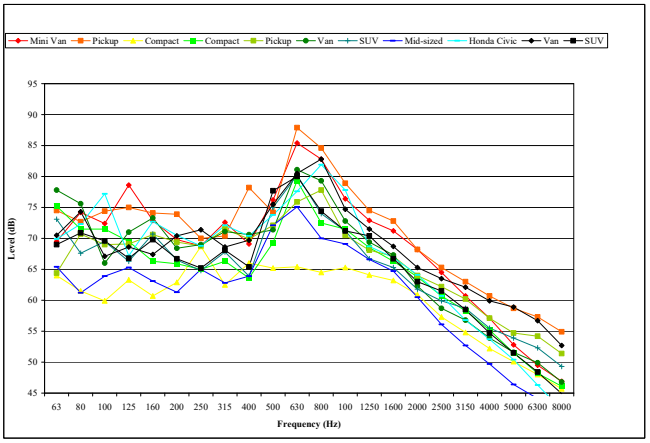


Figure 1: One-third Octave Band Spectra of Tire/Joint Interaction Noise with Tone in 630 Hz Band at Abutment Joint, August 2009

*duane.marriner@rwdi.com
† clair.wakefield@rwdi.com

These results were found to be insensitive to vehicle speed in the range 80 to 110 kph [1].

3.4 Long Range Average Maximum Noise Levels - General Traffic

Joint noise events from the general traffic were monitored at a distant residential location on the boundary of an exposed neighboring community. The monitoring station was located at a range of 585 m from the abutment joint and 1200 m from the pier joint along a line tangent to the direction of the traffic flow over the joint. The intervening ground consisted of tall natural grasses over which MEJ emissions propagated at an average height of 7 m. Sound travel times from abutment and pier joints to the microphone were approximately 1.6 and 3.6 seconds, respectively. Crosswind conditions prevailed along the propagation path since the winds were seen to follow the river.

Maximum levels of tire/joint interactions were identified using the recorded audio track. These short-term results are presented in Table 2.

Table 2: Average Noise Levels at Residential Location at a -585 m Offset from Abutment Joint - General Traffic, August 2009

Data Set	No. of MEJ Events	Avg. L_{maxF} (dBA)	Avg. L_{peak} (dBA)	Std. Error
1	30	58.1	70.1	0.6
2	30	57.0	69.6	0.6
3	30	56.6	68.4	0.7

In addition, maximum levels of joint events measured over 3-hours at the residence averaged 55.6 dBA against an ambient level of 52 dBA.

3.5 Directionality - Controlled Drive-bys

Motivation to investigate directionality was provided by numerous observations made at a variety of locations at two new bridge projects in B.C. that employ MEJs. In March 2010, the directionality of the tire/joint acoustical radiation field was more closely explored using a test vehicle at the pier joint. Noise levels were logged with the RTA using the mxSpec feature at longitudinal offsets of 7 m and 100 m from the transverse centerline of the pier MEJ. The results are presented in Table 3.

Table 3: Directionality of Tire/Joint Interaction Emissions

Microphone Longitudinal Offset(m)	Lane	Speed (kph)	mxSpec Level (dBA)	Difference
-7 m (behind) +7 m (ahead)	Middle	100	87.6 87.5	0.1
-100 m (behind) +100 m (ahead)	Middle	100	76.6 68.0	8.6

Column 5 of the table shows the difference in levels was negligible for an offset of 7 m on either side of the MEJ and this is believed to be the case up to a range of approximately 50 m. However, the difference for the 100 m offset indicated the MEJ emission level received 100 m behind the test vehicle was mxSpec 8.6 dBA higher than that received 100 m ahead of the vehicle. Subjective observations indicate this effect persists over longer distances and at the residential location at a range of 585 m, receding vehicles traversing the abutment MEJ were audible but approaching vehicles were not. MEJ emission directionality was also observed on a second new bridge project in B.C. employing a different type of MEJ.

4 Conclusion

Tire/joint interaction noise from the passage of general traffic over MEJs was found to be a complicated matter with numerous variables. Emissions from five vehicle classes varied as much as L_{maxF} 7.7 dBA. However, the tonal quality of MEJ emissions was quite consistent and the dominant frequency was found within the 630 Hz one-third octave band in eight of eleven cases. Moreover, this result was independent of vehicle speed; a result consistent with a resonance source mechanism [1]. Both subjective observations and measurements of MEJ emissions suggest factors leading to higher levels include higher vehicle speeds, tires that are more flexible and higher loading (kg/m^2) of the tire contact patch.

Joint noise imissions measured at a range of 585 m over largely soft ground averaged a clearly audible L_{maxF}(3-hour) 55.6 dBA against the ambient level of 52 dBA. The dominant 630 Hz tone of the tire/joint noise signature was audible at even lower levels that approached ambient. However, imissions from MEJ events would not be expected to contribute significantly to the L_{eq}(24-hour) since their durations were short on the order of 250 ms per event.

The directionality of tire/joint interaction noise was investigated. During controlled drive-bys it was found that the MEJ emission level received 100 m behind the test vehicle was L_{maxF} 8.6 dBA higher than that received 100 m ahead of the vehicle; an effect that persisted to the residential location at a range of 585 m. It was found that the joint emission level received 7 m behind the vehicle was approximately the same as the level 7 m ahead. More trials are necessary to determine the cause of the directionality property. The directionality property of the tire/joint radiation pattern may be of importance for noise mitigation purposes.

References

[1] D.E. Marriner, C.W. Wakefield, Modular Expansion Joint Noise in B.C. inter-noise 2011.

[2] C.W. Wakefield, D.E. Marriner, Analysis and control of bridge expansion joint croaking noise, Canadian Acoustics Vol. 39 No. 3 2011.

[3] K.A. Ravshanovich, H. Yamaguchi, Y. Matsumoto, N. Tomida, S. Uno Mechanism of noise generation from a modular expansion joint under vehicle passage. Engineering Structures 29 (2007).

EFFECTS OF TRAFFIC DENSITY ON MEASURED SOUND LEVELS – A CASE STUDY

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

1 Background

Consultants are frequently required to predict sound levels based on input data regarding road traffic. These assessments are used to design mitigation for new roadways or new noise-sensitive developments in the vicinity of existing roadways. It is typically assumed that the sound level is a function of (among other variables) the vehicular speed and the vehicle volume. As the volume of vehicles increases for a given speed, the sound level is assumed to increase.

2 Sound Level Relationships

2.1 Vehicle Volume and Sound Level

All models for transportation noise sources include a relationship between vehicle volume and predicted sound level. In Ontario, the mandated model is ORNAMENT[1] which uses the following equation :

$$L_{eq,i} = L_{0,i} + 10\log_{10} V - 10\log_{10} S + 10\log_{10} D_{ref} - 25 + other\ adjustments$$

Where $L_{eq,i}$ is the one-hour equivalent sound level for a specific vehicle type (automobiles, medium trucks, or heavy trucks), $L_{0,i}$ is the reference emission level for vehicle type i at a distance of 15 m, V is the hourly volume, S is the speed (in km/hr), and the "other adjustments" refers to terms accounting for distance, screening and angle of exposure, etc.

TNM (developed by the FHWA for use in the United States) uses the following relationship:

$$L_{eq,i} = 10\log_{10} E_{emis,i} + 10\log_{10} V - 10\log_{10} S - 13 + other\ adjustments$$

Where $E_{emis,i}$ is the reference emission energy of each vehicle type i.

In both cases, the dependence of sound level on volume is an adjustment equal to ten times the logarithm (base 10) of volume. The relationship is presented graphically in Figure 1.

2.2 Vehicle Speed and Sound Level

Similar to the above, models also assume/utilize a relationship between speed and predicted sound level. For both ORNAMENT and TNM, the vehicle speed factors into both the reference sound levels (which vary by vehicle type) as well as a direct speed adjustments shown in Equations (1) and (2). The relationships are presented graphically in Figure 2. In both cases, an arbitrary 1000 vehicles per hour has been used, split into 80% automobiles, 15% heavy trucks, and 5% medium trucks.

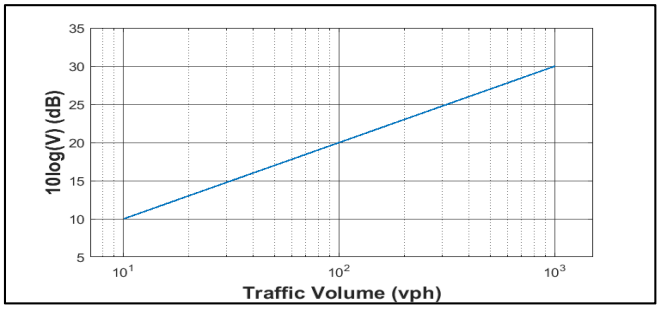


Figure 1: Relationship between Vehicle Volume and Sound Level

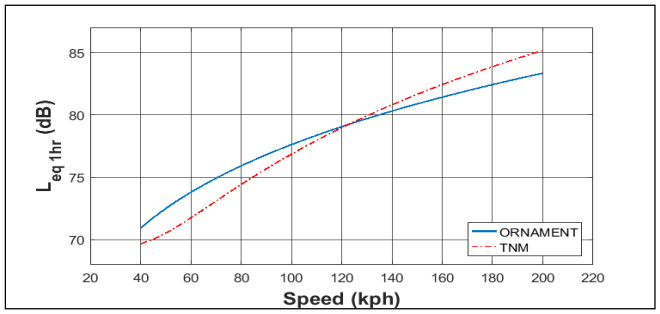


Figure 2: Relationship between Vehicle Speed and Sound Level

3 Case Study Measurements

3.1 Measurement Method

72-hour sound level measurements were done at 4 locations adjacent to the Don Valley Parkway in the City of Toronto. The measurement locations varied in setback distance relative to the highway as well as intervening topography. As the specific receptor sound level is not the focus of this study, the investigation will focus on one location which had a direct line of site to the highway (approximately 50 m from the roadway). The microphone was approximately 1.5m above grade. At the measurement location, the highway is divided with 3 lanes of traffic in each direction (northbound and southbound) and there are no access points in the vicinity of either set of lanes. Thus, flow should be essentially uninterrupted up to the critical vehicular density.

The sound level meters were set to measure 1 minute L_{eq} values for the full measurement duration. Road traffic counts were conducted concurrently with the sound level measurements.

It should be noted that the measurements were done during the spring season (prior to the end of the school year).

3.2 Results

Typically, sound levels resulting from road traffic on well travelled roadways follows a common pattern. See Figure 3. If it is assumed that the speed remains constant, then the measured sound level will increase with increasing vehicular volume. This results in two characteristic peaks – one during

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

the morning rush, and one during the evening rush – followed by a significant decrease during the nighttime period.

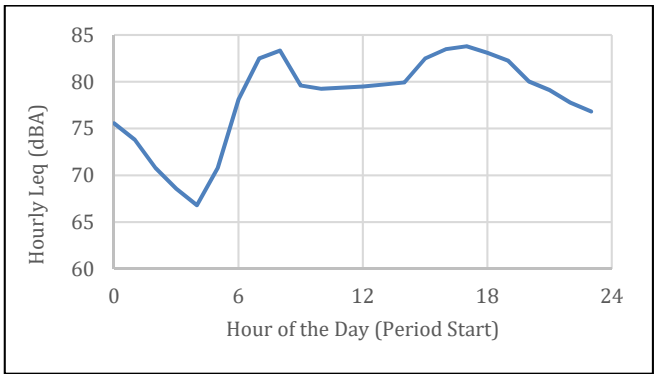


Figure 3: Typical Fluctuation of Sound Level vs Time of Day for a Well-Travelled Road

In this case study, there is a significant dip in measured sound level during the 0800 to 0900 hour (based on hourly Leq data). This dip was seen consistently in each of the three morning periods which were included in the measurements (see Figure 4). Road traffic volumes are also plotted in Figure 4.

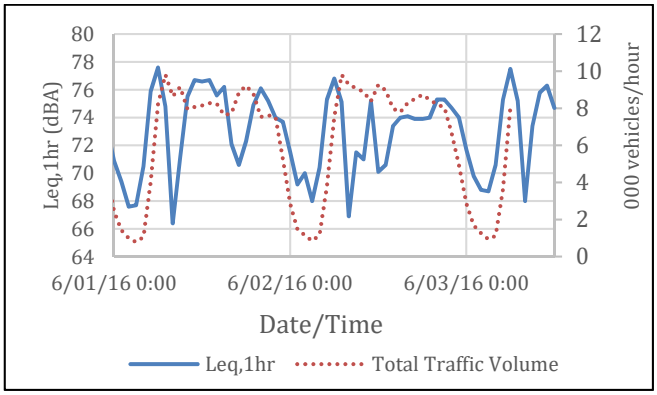


Figure 4: Measured Sound Level Time History

3.3 Discussion

It is clear from the measured data that although the sound level decreased significantly each day between 0800 and 0900 hours, a similar decrease in traffic volume was not observed. Thus, the decrease in measured sound level is not the result of a significant decrease in traffic flow. This is intuitive as the time period in question (0800 to 0900 hours) lies well within the typical AM Peak period ('rush hour').

Based on observations at the time of the measurements, the decrease in sound level was likely attributed to a significant decrease in vehicle speed resulting from an overall increase in vehicle density.

4 Relationship Between Vehicle Density, Flow, and Speed

The Traffic Engineering Handbook[2] provides the basic relationships between speed, density, and flow for uninterrupted traffic. Based on the traffic data presented above, specific interest is given to the relationship between flow (vehicles/hour) and speed (km/hour). See Figure 5.

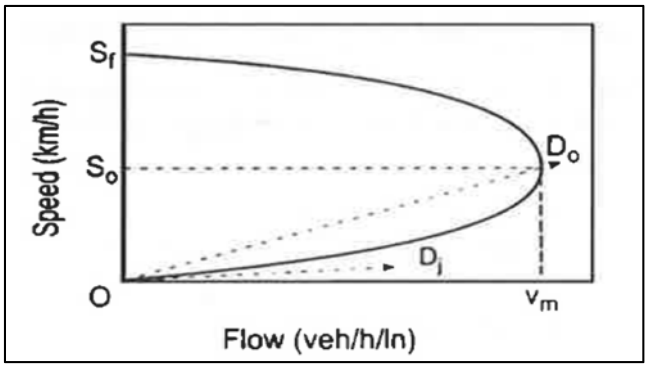


Figure 5: Conceptual Relationship Between Speed and Flow

For a roadway which operates in the vicinity of optimum density (D_0), large changes in speed can result in only small changes in flow rate (above or below speed S_0). This is consistent with the measured sound data which showed only small changes in flow (volume) but showed a remarkable decrease in measured sound level between 0800 and 0900 hours.

5 Effect on Sound Level for Increasing Vehicular Density

As noted above, the sharp decrease in measured sound level is not correlated to a significant change in traffic flow during the measurement period. Although the predicted sound level will decrease with the observed decrease in vehicle flow, the decrease in sound level far outpaces that of the volume decrease. Rather, it is related to a decrease in speed which results when the vehicular density exceeds the critical density for the roadway.

This decrease in speed produces a well understood decrease in emitted sound level for the vehicles on the source roadway.

6 Conclusions

It is clear from the above case study that the typical application of road traffic noise models (ie using static speed throughout the full time period) may not accurately predict hourly sound levels for a roadway which may exceed the critical density. Rather, when the density exceeds the critical density, a significant decrease in speed is possible without a corresponding decrease in flow (vehicles per hour). The speed reduction become the controlling variable in the sound emission level for the roadway and as such the emitted sound energy from the roadway decreases significantly.

References

[1] Ontario Ministry of the Environment. *Ontario Road Noise Analysis Method for Environment and Transportation (ORNAMENT)*, October 1989.

[2] J Pline. *Traffic Engineering Handbook*, 5th Ed., Institute of Traffic Engineers, 1999

CONSIDERATIONS REGARDING AN ACOUSTIC CRITERION FOR WIND TURBINE ACCEPTABILITY

William K.G. Palmer[1]
[1] TRI-LEA-EM, 76 Sideroad 33-34 Saugeen, RR 5, Paisley, ON, N0G 2N0, trileaem@bmts.com

1 Introduction

A common regulatory acceptance criterion for wind turbine installation in Canada is that sound pressure level does not exceed 40 dBA outside a home when the wind speed at 10 metres elevation does not exceed 4 metres per second. A clue to the ineffectiveness of this criterion can be seen from over 2700 complaints filed in Ontario with regulators by residents living in homes where acoustic conditions were predicted in approved models to comply with the current criterion. Residents noted the intrusiveness of an imposed sound higher in amplitude and different in quality than the pre-existing background. Residents reported disrupted sleep, and adverse health consequences. Fundamental premises of Environmental Protection Acts[1] (EPA) are that emissions of a contaminant such as noise should not cause an adverse effect including loss of enjoyment of normal use of property, or annoyance that lead to human health impacts.

2 Method

2.1 Listening to impacted individuals

The first step was to listen to the complaints. The author sat face-to-face with many individuals who told of adverse impacts since wind turbines were erected. Some had looked forward to the installation of wind turbines, and at first did not notice much impact. As time progressed, they noticed changes. Sleeping patterns were interrupted. They were often tired. Some members of their family were bothered by nausea, others by dizziness or vertigo. Those with pre-existing diabetes spoke of difficulty controlling blood sugar. They spoke of migraines and blood pressure changes. They recounted cognitive issues and feeling mentally fuzzy. While there were other issues and no universal problem, the individuals were sincere as they recounted how their lives had changed for the worse. They found that when they went away from the wind turbines, their condition improved, but on returning the adverse conditions resumed. While this paper does not pretend to give a medical diagnosis, or to identify a specific cause, the nature of the complaints were clues to search for changes in the environment that might explain what these people experienced.

2.2 Measuring background pre-turbine in service

Measurement of the background at homes distant from wind turbines, and before wind turbines were installed was conducted, in different seasons, and at different times of day. Often the rural environment background measurements challenged the 22 dBA noise floor of the Earthworks M30BX 9Hz to 30 kHz (± 1/-3 dB) microphone used or the 30 dBA noise floor of Level 2 sound level meters.

2.3 Monitoring outside homes post wind turbines

Measurements were conducted again at many of the same locations where pre-turbine monitoring had been conducted after wind turbines were erected and also both near to and distant from operating wind turbines, in a similar environment of weather, wind, and traffic.

2.4 Monitoring inside homes post turbine start-up

The reports of a number of individuals suggested that monitoring inside homes was also required. The narratives at first seemed to defy logic. Individuals described how:

- Because sleep was poor inside their homes, they tried sleeping in a tent outside, and found it was better.
- Some family members could not sleep in bedrooms and moved into basement rooms to try to get some rest.
- In a restless night, they might even try reversing their head to foot position in bed.

3 Results

3.1 Conditions outside: changes post turbines

Without turbines, during spring and summer insects and birds often caused sound backgrounds of 35 dBA or more detectable as audible higher frequency peaks in a FFT analysis of sound samples. But, at night and particularly in the fall and winter without leaves on trees or insects, the background was very low, and the call of coyotes several km from the measurement site might be clearly detected. Some test sites would show traffic noise during times such as “shift change” at nearly plants, but those sounds were intermittent and ended before bedtime. Other monitoring was conducted at sites near streams or the shore of Lake Huron, where wave action was present. Inspection of recordings taken showed a randomness of the wave pattern and a smooth rise and fall of the sound different from sharp, regular pattern of rise and fall near the wind turbines. Pre-turbines the rural background measured at night was often 10 to 15 dBA below the 40 dBA current acceptability criterion, and random events without a particular pattern predominated. A bird might call, but birds would then be quiet, particularly after dusk. Post wind turbine start up, several differences were readily identifiable. At homes near the wind turbines, while the dBA readings post turbine start up might be within the 40 dBA limit, the flat weighted sound pressure levels often increased by 10 to 15 dB. The low frequency component of the sound shown by a FFT analysis was 15 to 20 dB higher at frequencies below 500 Hz. The soundscape, or acoustic environment, was significantly altered. In particular the cyclic broadband sound variation in sequence with the blade/tower interaction

was readily identifiable. After wind turbines were in service there was no hearing of distant sounds such as coyotes, or farm livestock. The dBA numbers might not be much different. But the condition was very different.

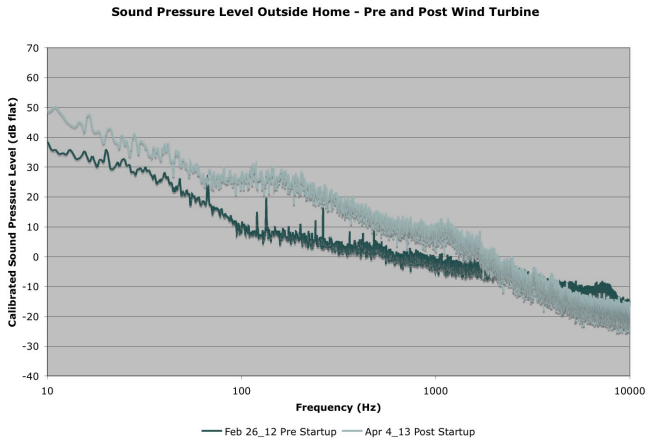


Figure 1: Level weighted sound pressure levels outside a home pre- and post-turbine startup (only one example of many)

3.2 Conditions inside homes differ from outside

The results of monitoring inside a typical Ontario farm with tall, square bedrooms, showed that the greatest sound pressure levels were in the corner of the room, and that the centre of the room had the lowest levels. Inside the home, the sound levels were greater at low frequencies, and more tonality was noted between about 10 Hz and 400 Hz than outdoors.

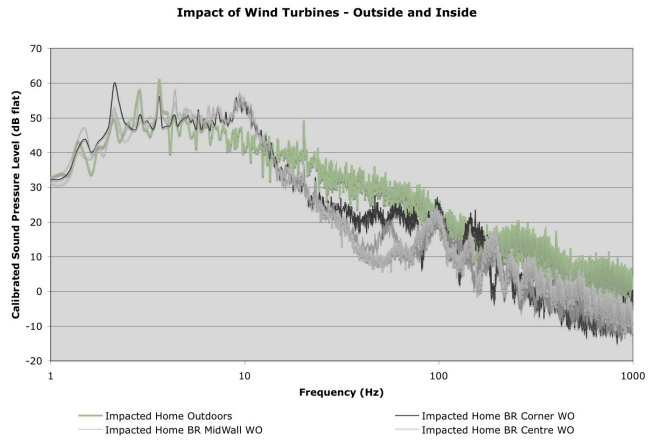


Figure 2: Level weighted sound pressure levels inside and outside home impacted by wind turbines (note cyclical blade pass tonality)

3.3 Acceptance criterion insensitive to change

The 40 dBA regulatory acceptance criterion is insensitive to the actual changes that take place, which increase the sound levels at frequencies below the values at which the dBA ratings are most sensitive. The acceptance criterion is insensitive to flag the change in the night time conditions, where the sound levels change by 10 to 15 dBA after wind turbines are erected – from perhaps 25 dBA to about 40 dBA. The acceptance criterion is unable to detect the changes inside homes, where sound levels actually can become higher inside than outdoors, and show more tonality. The acceptance criterion does not detect or act on the cyclical nature of the sound outside or inside impacted homes.

4 Discussion

The subject of amplitude modulation of wind turbine noise emissions (otherwise described as a cyclical noise rising and falling in magnitude) has been a principal focus of wind turbine noise international conferences in Glasgow (2015) and Denver (2013). Monitoring of the sound inside homes displays a different character than outside, showing pulses with peak to trough amplitudes exceeding 5 dB at frequencies that are within the audible range. A simple example shows that dBA weighting does not adequately reflect perception and annoyance. White noise at 40dBA has a very different perception than pink noise at 40 dBA.

5 Conclusion

The prevalence of complaints from residents impacted by wind turbines and the difference between pre- and post-wind turbine noise measurements and quality shows that the current regulatory acceptance criterion is ineffective at meeting the objectives of the EPA(s). Post-turbine changes should not cause annoyance that results in loss of enjoyment of normal use of property, or lead to adverse human health impacts. Changes in the intrusiveness of wind turbines compared to background are not sensed and are not being monitored. It has already been suggested that the cyclical nature of the sound from wind turbines is a contributing factor in complaints[2]. Others have suggested that monitoring conditions inside homes should be considered[3]. Revision of the regulatory acceptance criterion for wind turbines is called for to meet EPA objectives.

Acknowledgments

Many contributed to this work. Their assistance and inspiration is appreciated, but they are blameless for the presentation. The author acknowledges the patience of citizens living near wind turbines to permit acoustic monitoring outside and inside their homes. Early work by G.P. (Fritz) van den Berg alerted readers of annoyance from the sound of wind turbines. John E. Coulter raised concerns of cyclical noise as an annoyance factor before an Ontario Municipal Board Tribunal. Kerstin Persson Waye cemented the idea of cyclical noise annoyance. Comments of Brigitte Schulte-Fortkamp and Klaus Genuit guided understanding of soundscape and psychoacoustics. Discussions with Mathais Basner and Wolfgang Babisch aided in understanding the effect of noise on man.

References

[1] Environmental Protection Act, Ontario, R.S.O. 1990, c.E.19, 14

[2] J.E. Coulter. Testimony before Ontario Municipal Board 2007, PL060986.

[3] B. Howe, N. McCabe, I. Bonsma, Addressing low frequency sound and infrasound from wind turbines, JCAA (39) No.3 (2011)

COMMUNITY NOISE INVESTIGATION OF A "LOW LEVEL THRUMMING" ATTRIBUTED TO THE OPERATION OF A MINE

Andrew Williamson, P.Eng.[1]
RWDI AIR Inc./Wakefield Acoustics, 301-2250 Oak Bay Avenue, Victoria, BC, V8R 1G5

1 Introduction

In May 2016, RWDI/Wakefield Acoustics was asked by the operator of a mine to investigate noise complaints from a resident living 2.7 km from the mine. The mine produces concrete aggregate, construction materials and specialty blended sands. The major sources of noise at the mine during the time period of interest are conveyors and rock crushers. The resident reported experiencing the noise during the nighttime and early morning hours in their bedroom and described the noise as a "low level thrumming that increases in intensity when lying down in bed with my head on a pillow". The operator of the mine was surprised by these complaints as the mine equipment largely shuts down between 23:00 and 24:00. To investigate the noise complaints, RWDI/Wakefield Acoustics conducted continuous noise monitoring at the home of the resident, the mine and various locations in the community to establish the frequency spectra of mine noise emissions, observe the variations in mine noise levels, and determine if the mine frequency spectra and noise level variations correlated with the noise data collected in the community.

2 Methodology

2.1 Noise Monitoring

Noise monitoring was conducted at the mine and five other locations in the surrounding community. The key monitoring sites discussed in this paper are the mine (site 1), the home of the resident (site 2) and a "control site" at a nearby home on the same street as the resident (site 3). The sound level meter at the mine was placed in an elevated position with a clear line of sight to the noise sources (i.e. conveyors, crushers). As the resident had been experiencing the noise in the bedroom at the back of their house, the sound level meters at sites 2 and 3 were placed in the backyards.

The noise monitoring was carried out with Bruel & Kjaer 2250 and Larson Davis LxT sound level meters (SLM). Noise level data were logged in 15-minute intervals and the primary data collected were the A- and C-weighted L_{eq} and one-third octave band levels.

2.2 Noise Criteria

The World Health Organization (WHO) community noise thresholds[1] were used too assess the severity of mine noise at the resident's home (assuming mine noise was detectable). The WHO criteria are summarized in **Table 1**. The presence of low-frequency noise was assessed using the criteria recommended in the Health Canada (HC) 2011 Draft Noise Guidelines[2]. The HC document considers low-frequency noise to be present when the difference between C- and A-weighted levels is 10 dBA or more. HC also recommends that the energetic-sum of the sound levels in the 16, 31.5 and 63 Hz octave bands not exceed 70 dB to avoid rattling building structures.

Table 1: WHO Community Noise Thresholds

Environment	Health Effect	L_{eq} (dBA)
Outdoor Living Area	Serious annoyance, daytime and evening;	55
	Moderate annoyance, daytime and evening	50
Dwellings, indoors	Speech intelligibility and moderate annoyance, daytime and evening;	35
	Sleep disturbance, nighttime	30
Dwellings, outdoors	Sleep disturbance, window open (outdoor values)	45

3 Results

3.1 Average Noise Levels (A-weighted L_{eq})

Table 2 presents the A-weighted L_{eq} measured during time periods when the mine was fully operating, partially operating and shut down.

Table 2: L_{eq} for Different Periods of Mine Operation

Site	Approximate L_{eq} (dBA)		
	Fully Operating (18:00-00:30)	Partially Operating (00:45-2:30)	Shutdown (2:30-3:30)
1	61	45	39
2	40	25	35
3	39	27	37

From **Table 2**, it can be seen that noise levels at the residences (sites 2 and 3) did not exceed the WHO thresholds. While the L_{eq} at the residences were highest when the mine was fully operating, the higher noise levels were not necessarily due to mine operation. Rather, noise would generally be expected to be higher during the evening than during the early morning hours due to the presence of community noise sources such as road traffic, aircraft and marine vessels. When the field engineer visited the residences at approximately 23:00 mine noise was not observed to be audible relative to the background noise environment that was composed of natural sounds such as ocean waves breaking on the shore.

[1] Andrew.Williamson@RWDI.com

3.2 Low-Frequency Noise

Table 2 presents the C-weighted noise levels at the residence (sites 2 and 3) for different phases of mine operation and also presents the differences between the C-weighted and A-weighted levels.

Table 3: Low-Freq. Noise Levels for Different Periods of Mine Operation

Site	Mine Operation	L_{eq} (dBA)	L_{eq} (dBC)	Diff. (dB)
2	Full	40	55	15
	Partial	25	51	26
	Shutdown	35	46	11
3	Full	39	51	12
	Partial	27	45	18
	Shutdown	37	42	5

When the mine was operating (full and partial), the differences in the C- and A-weighted noise levels exceeded the HC 10 dB criterion at both sites. When the mine was shutdown, however, the criterion was only exceeded at site 2. The sum of the noise levels in the 16, 31.5 and 63 Hz one-third octave bands did not exceed the 70 dB rattle criterion at either residence.

3.3 Frequency Spectra

Figure 1 shows the one-third octave band frequency spectra measured at sites 1, 2 and 3 when the mine was partially operating.

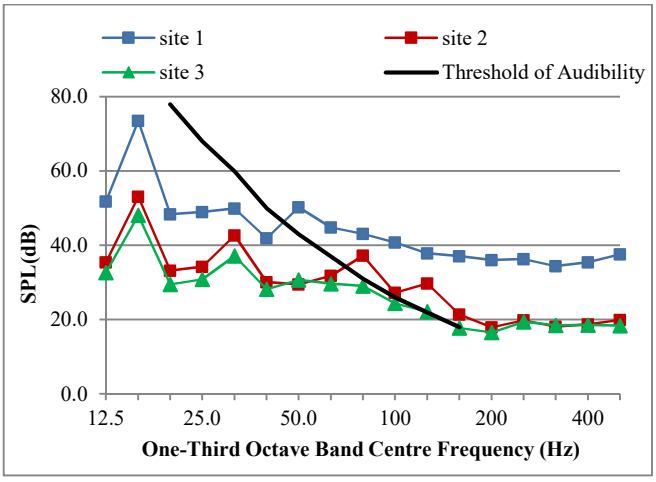


Figure 1: Frequency Spectra when Mine Partially Operating

In **Figure 1** it can be seen that at all three sites there is a prominent tone in the 16 Hz one-third octave band which, being most intense at the mine, was concluded to be due the operation of the mine. This tone, however, was both too low in frequency and sound pressure level to be audible by the average human ear. At site 2 there were also tones in the 80 and 125 Hz one-third octave bands which would be expected to be audible. As these tones were not present in the frequency spectra at the mine (site 1) it was concluded that they must be originating from a source other than the mine. In further support of this hypothesis, the tones were found to be present at site 2 even when the mine was shutdown. Interestingly, these tones were not present at the neighboring home (site 3). These results suggest that the tones are being emitted by a source(s) in close proximity to the resident's home. As these tones are above the threshold of audibility, it is possible that they are causing the resident's noise disturbance.

To further investigate the nature of these tones, time histories were created of the sound pressure levels in the 80 and 125 Hz one-third octave bands. The 125 Hz time history (**Figure 2**) was particularly interesting as it revealed a cyclical pattern at the resident's home (site 2) that is consistent with intermittently operating mechanical equipment (e.g. heat pump).

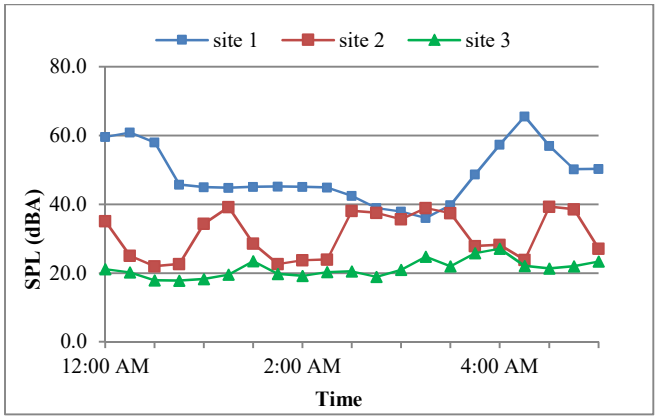


Figure 2: 125 Hz Time History

4 Conclusion

The results of the community noise investigation suggest that the noise being experienced by the resident is due to source(s) other than the Lehigh mine. Outdoor noise levels at the resident's home are within the thresholds of WHO community noise guidelines. While low-frequency noise from the mine was found to be present at the resident's home, the noise was largely at frequencies and sound pressure levels below the range of average human hearing.

The outdoor frequency spectra at the resident's home contains tones in the 80 and 125 Hz one-third octave bands that are at audible levels. It is suspected that these tones are what the resident is hearing. These tones were absent from the frequency spectra measured at the mine, and are present even when the mine is shutdown. As these tones were also absent from the frequency spectra measured at a neighboring residence (site 3), the source(s) of these tones are likely in close proximity to the resident's home.

References

1. World Health Organization. *Guidelines for Community Noise*. 1999
2. Health Canada. *Guidance for Evaluating Human Health Impacts in Environmental Assessment: Noise (Draft).* 2011


For Digital Recorders
Introducing
For USB A/D Systems
PHANTOM POWER
7052PH
Measurement Mic System
7052H Type 1.5™
Titanium Diaphragm
3Hz to >20 kHz
<20 dBA > 140 dBSPL
MK224 (<14 dBA to >134 dBSPL) Optional
4048 Preamp
Superior
IEC 1094 Type 1
Long-term Stability
Temperature and Humidity
Performance
Now in Stock
Phantom
to IEPE/ICP
Adaptor
Supplies 3-4 mA
Power
Accelerometers
Microphones
ICP1248
MATT™
MATT™
Family
Mic
Attenuator
Handle Higher Sound Pressure Levels
ACO Pacific, Inc., 2604 Read Ave., Belmont, CA 94002
Tel:(650) 595-8588 FAX: (650) 591-2891 E-Mail: sales@acopacific.com
Web Site: www.acopacific.com
ACOustics Begins with ACO
TM

ACOUSTICAL CONSULTING: NOISE CONTROL - CONSULTING : CONTRÔLE DU BRUIT

Selecting Suitable Noise Control For Mine Return Air Raise Systems
Andrew Dobson, William Gastmeier 48

Effectiveness Of Noise Reducing Asphalt Pavements
Scott Penton 50

Noise Mitigation Of Small Ptac Systems
Scott Penton, Lucas Arnold, Nick Walters 52

Predicting Exhaust Sound Power Levels Of General Purpose Boilers
Robert D Stevens, William J Gastmeier 54

Noise Control For High Plume Exhaust Fans: A Case Study
Peter VanDelden, Greg Downey, Matt Downey 56

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 58

SELECTING SUITABLE NOISE CONTROL FOR MINE RETURN AIR RAISE SYSTEMS

[1]Andrew Dobson, BSc and [2]William J. Gastmeier, MASc, PEng
HGC Engineering, 2000 Argentia Rd., Plaza 1, Suite 203, Mississauga, Ontario, Canada L5N 1P7

1 Introduction

The above ground fresh air and return air fans used to ventilate underground mines have the potential to be the greatest sources of noise at mines. These fans often have direct exposure to the outdoors via the air intake or exhaust duct openings, they are often situated closer to sensitive points of reception than other sound sources such as the supporting infrastructure surrounding a headframe, and they can therefore have a significant impact on the communities surrounding the mine. Accordingly, noise control measures, such as intake and outlet silencers, are often incorporated within the design of the ventilation system.

Recent measurements and observations conducted in response to a residential noise complaint suggest that moisture and particulate drawn through the return air ventilation systems can cause rapid clogging and degradation of the outlet silencers, if not properly accounted for in the design of the system. The implications of this are multifaceted, and include significant economic, environmental, and residential annoyance concerns. This paper will present measured sound level data collected as part of a complaint investigation and silencer maintenance program, outline elements which can be considered when designing a return air system to optimize the potential of the noise control equipment, and examine the feasibility of implementing nontraditional noise control methods.

2 Noise Complaint Investigation

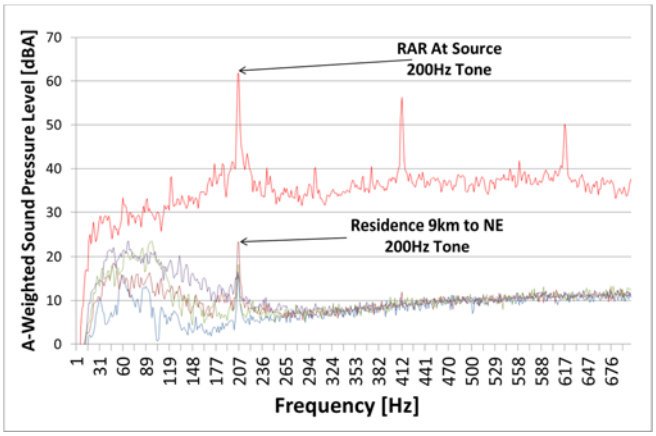
2.1 Residential Measurements

An initial series of sound level measurements were conducted in response to a noise complaint from a resident situated approximately 9 kilometres to the northeast of a mine in northern Ontario. In general, natural sounds dominated the acoustical environment, however a characteristic tonal frequency signature was evident during lulls in the background sound. Accordingly, narrow-band (Fast Fourier Transform, or "FFT") sound levels were measured as a means of discerning the tonal frequency, which resulted in a clear peak at 200 Hz.

It was suspected that the fans associated with the main return air raise ventilation system ("RAR") were responsible for the audible tone at the residence. To confirm this, additional FFT sound levels were measured near to the RAR to quantify the acoustic signature unique to the fan sound. The audible tone associated with ventilation fans are a function of the blade pass frequency ("BPF"), which in this case, was used to verify the source of the sound at the residence. The measurements were conducted using a Norsonic Nor140 precision sound level meter, which was within its annual laboratory calibration period. Correct calibration was field-verified.

The results of this comparison indicated that the acoustic signature of the primary tone measured and observed at the residence matched the acoustic signature of the RAR. Figure 1, below, shows the A-weighted narrow-band sound levels measured at the residence, and near to the RAR.

Figure 1: Narrowband Frequency Analysis



2.2 RAR Silencer Cleaning

Inspections of the RAR silencers suggested that the perforations in the baffle casing had become significantly clogged, due to the abundance of moisture and particulate being drawn through, and discharged from the silencers. Figure 2, below, shows a picture of a clogged silencer.

Figure 2: Clogged Perforations in Baffle Casing



[1]adobson@hgcengineering.com
[2]bgastmeier@hgcengineering.com

As a temporary means of minimizing the sound of the RAR, attempts were made to clean the silencers using water and a pressure washer. Because the silencers were not designed to facilitate routine cleanings, the perforated casing was not removed from the silencers, thus the degree of damage incurred by the baffle fill was not known.

Measurements of the RAR outlets were conducted prior to and after the cleaning of the outlet silencers, on three occasions, to quantify the benefit of the cleaning activities. Measurements were conducted in third-octave bands using a Brüel & Kjær Hand-held Analyzer Type 2270, equipped with Sound Intensity software BZ-7233, a Brüel & Kjær model 3654 Sound Intensity Probe and a pair of phase-matched model 4197 microphones. All instrumentation was calibrated correctly before and after the measurements. Sound levels were measured at a distance of 50 metres, at various angles from the outlets, and converted to the overall A-weighted sound power levels listed in Table 1, below.

Table 1: Sound Power Levels [dB re. 10^{-12} Watt] Silencer Outlet Before and After Cleaning

Silencer Condition	**dBA**
March 4, 2015 - Before Cleaning	128
April 7, 2015 - After Cleaning	119
Sound Level Reduction	**9**
November 3, 2015 Before Cleaning	124
November 6, 2015 After Cleaning	116
Sound Level Reduction	**8**
April 11, 2016 – Before Cleaning	124
April 14, 2016 – After Cleaning	118
Sound Level Reduction	**6**

The results outlined in Table 1 indicate that the cleaning of the RAR outlet silencers in April of 2015, November of 2015, and April of 2016 provided sound level reductions of 9 dBA, 8 dBA, and 6 dBA, respectively. The decreasing improvement in silencer performance afforded by the successive cleanings suggested that the absorptive fill within the acoustic baffles of the silencers was damaged and continuing to degrade with time, and that future cleanings would not mitigate the future audibility and potential for disturbance at the residence of the complainant.

In addition, when the system was designed, the sound power levels specified by the manufacturer for the silenced and unsilenced RAR outlets were approximately 97 dBA and 120 dBA, respectively, under a similar operating speed (750 RPM). Therefore, it was evident that the sound level reduction of the silencers, after becoming clogged, was negligible, and that the fans were louder than specified.

3 Alternative Considerations & Discussion

Considering the results outlined above, it was determined that the noise control equipment on the RAR system would need to be retrofitted or replaced.

To address the issue, the most straightforward method will be to install new rectangular dissipative silencers, to avoid the reconstruction of the outlet ducting. Silencers which have easily accessible, removable and/or replaceable acoustic baffling should be considered, to allow for cost effective cleaning and/or replacement in the future.

Because the reason for the degradation and clogging of the silencers is the moisture and particulate being drawn through the RAR system from underground, the addition of a moisture dropout chamber/trench will be beneficial toward preserving the internal silencer media. An added benefit of this will also be minimizing the release of water contaminated from the underground mining operations, into the surrounding environment.

Consideration can also be given to investigating reactive silencers. These can be tuned to attenuate sounds at specific frequencies, such as the BPF (and harmonics thereof) of the RAR fans. Further, as demonstrated in Figure 1, the acoustic energy measured and observed at the residential location is concentrated at the BPF of the fans; by specifically targeting the BPF, rather than a more typical broadband approach, the efficiency of the silencer could be maximized. Conversely, this approach is not advisable for ventilation systems which utilize variable speed fans, when the BPF (and associated acoustic signature) will also be variable.

4 Conclusion

The noise complaint investigation outlined above has indicated that incorporating typical, or “off-the-shelf” noise control measures on equipment which require extra considerations can result in noise control measures which need significant maintenance, and do not work as intended.

Specifically, it has been determined that mine return air fans have the potential to draw moisture and particulate from underground, and this can degrade a dissipative silencer to a state of being nearly entirely ineffective. The consequence of this has been the unforeseen annoyance of neighbouring residents, the need for costly routine maintenance, and the need to replace the ventilation equipment and/or noise control equipment. Moreover, this has the potential to contribute to sound levels which exceed the regulatory sound level limits enforced by a governing body.

Therefore, it is prudent that RAR systems are fully understood during the design phase of a mine, and the appropriate noise control measures are incorporated.

EFFECTIVENESS OF NOISE REDUCING ASPHALT PAVEMENTS

R. L. Scott Penton, P.Eng.[1]
[1] Novus Environmental Inc., 150 Research Lane, Suite 105, Guelph, Ontario.

1 Abstract

As vehicles pass over a roadway surface, noise is created due to the interaction between the tires and the pavement surface. For most vehicles travelling at speeds in excess of 60 km/h, the majority of the noise emitted by the vehicle comes from the tires. Noise reducing asphalts operate by affecting the noise generation mechanisms, and through providing an acoustically absorptive surface for sound waves travelling outward from the vehicle. On-board sound intensity measurements were used to evaluate reductions in tire noise for several pavement types, as part of testing for the Ontario Ministry of Transportation. Measurements were conducted over several years to quantify the longevity of any measured noise reduction. Comparisons are made versus other published test results.

2 Introduction

Novus Environmental Inc. (Novus) was retained by the Ontario Ministry of Transportation (MTO) from 2010 through 2014 to undertake On-Board Sound Intensity (OBSI) noise measurements of sound levels from noise reducing asphalts on the Hamilton-bound section of Highway 405, between Station 12+000 and 14+500. These are quiet pavement test sections, paved in fall 2009, where MTO tried several asphalt mix types intended to reduce tire/pavement noise.

The purpose of the measurement program was to evaluate the noise generation of five test pavement sections, through OBSI measurements, conducted using the method outlined in the then-current AASHTO TP-76-12 test method.

3 Test Sections

Five (5) 500-m long test sections, including a normal pavement control section, were tested.

Test Section	Pavement Mix Type and Thickness
A	30mm OGFC, over 50mm HL4 OBC
B	30mm OGFC, over 50mm Superpave 19.0
C	30mm RAC-O, over 50mm Superpave 19.0
D	30mm SMA 9.5, over 50mm Superpave 19.0
E	40mm Superpave 12.5 FC2, over 50mm Superpave 19.0 – CONTROL

Table 1: Pavement Test Sections

The noise reducing asphalts tested included two open-graded friction course (OGFC) types, an open graded rubberized asphalt concrete (RAC-O), and an open graded stone mastic asphalt (SMA). Open graded "popcorn" type pavements have particular size fractions of the aggregate sifted out prior to mixing. The resulting pavement has air voids and gaps between relatively coarse stones.



Figure 1: Open graded pavement (OGFC) versus Typical Control pavement (Superpave HMA)

Noise reducing asphalts operate by affecting the noise generation mechanisms, and potentially through providing an acoustically absorptive surface for sound waves travelling outward from the vehicle.

4 On-Board Sound Intensity Measurements

The OBSI method allows for noise emissions from the tire / road to be measured directly, and allows for various pavement types to be compared. Phase-matched microphone pairs in a "side by side" configuration are attached to the wheel hub at fixed distances and locations from the tire on a given vehicle. The probes are located to capture noise generated at the beginning (leading edge) and end (trailing edge) of the "contact patch". A standard reference tire is used.

The orientation of the probes allows for the "sound intensity" (sound power level flux per unit area) originating from the tires to be directly measured, eliminating noise from extraneous sources, such as wind noise.

In this assessment, the procedures outlined in the American Association of State Highway and Transportation Officials (AASHTO) Standard TP-7-12, Standard Method of Test for Measurement of Tire/ Pavement Noise Using the On-Board Sound Intensity (OBSI) Method were used. Additional guidance was obtained from the National Cooperative Highway Research Program Report 630.

The test sections were paved in 2009. Measurements were conducted in 2009 (by another firm), 2010, 2012, and 2013 (all by Novus), to study the noise reductions due to the asphalt types, and the longevity of any reductions identified. In addition, the 2013 measurements were conducted before and after a vacuum truck cleaned the pavement, to see what effect that would have on the results. Measurements were conducted at 80 km/h and 100 km/h, representing typical highway posted speed limits.

[1] scottp@novusenv.com



Figure 2: AASHTO-complaint OSBI Test Rig Used in Measurements

5 Results

A review of the measurement results shown in **Table 2** indicates the following trends:

- Overall average sound intensity levels measured became progressively louder for the first 3 years after installation.
- The 4th year following installation did not significantly affect sound intensity levels.
- Different pavements deteriorate at different rates.
- Cleaning did not restore the asphalts to original intensity levels.

A comparison of the OBSI results versus the results from the Noise Intensity Testing in Europe (NITE) Studies I and II has been conducted. The results are shown in **Figure 3** below.

Test Section	Sound Intensity (dBA) Relative to Control (Section E)				
	2009	2010	2012	2013	2013 Cleaned
A	-4.5	-3.7	-1.7	-2.6	-3.4
B	-1.9	-1.1	-0.3	-1.6	-2.4
C	-3.8	-3.2	-1	-0.9	-1.4
D	-2.8	-0.5	1.7	0.9	1.0
E	-	-	-	-	-

Table 2: Comparison of test results (100 km/h) versus control pavement (Section E) over time.

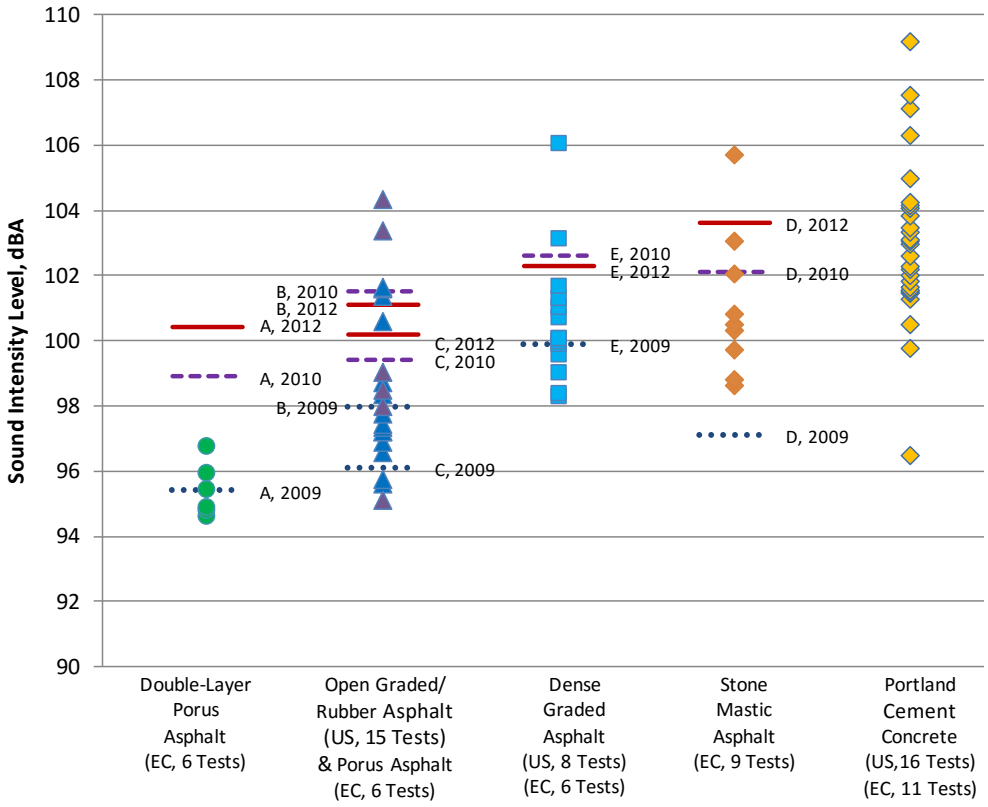
The measured values for sound intensity, sorted by type of pavement, generally fall into the NITE measured ranges.

6 Conclusions

A review of the measurement results indicates that noise reducing asphalts provide only modest reductions in noise level (< 3dB); that their effectiveness quickly decreases with time; and that cleaning provides only a marginal improvement.

Acknowledgments

Thank you to the Ontario Ministry of Transportation Materials Engineering Research Office (MERO).



Notes:

- OBSI results for light vehicles, in dBA. Results are shown for five general pavement types.
- Dots are results from Noise Intensity Testing in Europe (NITE) Studies I and II. NITE values are compilation/ comparison of California, Arizona, and European noise measurement results at 60 mph (97 km/h)
- Lines are results from Highway 405 testing for 2009, 2010, and 2012, at 100 km/h. Highway 405 Test Sections are:
  - Test Section A: 30mm OFC, over 50mm HL4 OFC
  - Test Section B: 30mm OFC, over 50mm Superpave 19.0
  - Test Section C: 30mm RAC-O, over 50mm Superpave 19.0
  - Test Section D: 30mm SMA 9.5, over 50mm Superpave 19.0
  - Test Section E: 40mm SP 12.5 FC2, over 50mm Superpave 19.0

Figure 3: Comparison of OSBI Measurement Results with NITE Studies I & II Results

NOISE MITIGATION OF SMALL PTAC SYSTEMS

R. L. Scott Penton, P.Eng. [*1], **Lucas Arnold, P.Eng.** [†1], and **Nick Walters, MASc., EIT** [‡1]
[1] Novus Environmental Inc., 150 Research Lane, Suite 105, Guelph, Ontario.

1 Abstract

In order to meet the facade noise level requirements of an Ontario municipality, mitigation of externally generated noise from already-installed packaged terminal air conditioning (PTACs) systems on several residential towers was required. Several noise mitigation methods were developed which could be retrofitted into the existing units. These mitigation methods and the design process are discussed.

2 PTAC Systems

PTAC systems are used to provide heated or cooled air in a number of applications, from residential apartments and condominiums, to hotels and motels.

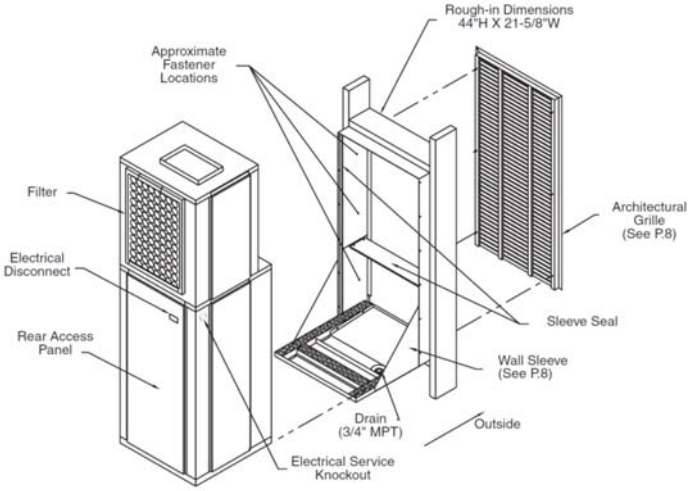


Figure 1: PTAC system general assembly [1]

Heating for the unit is provided either by electrical heat or by hot water from central boilers. Cooling is provided by an internal air conditioning unit consisting of a compressor, heat exchanger, fan and condenser.

The condenser and compressor are cooled using outside air which is drawn in through and exhausted out of the unit through a louvre on the outside of the building. Thus, noise from the unit can radiate into the outdoor environment, and can affect other units in the same building, or off-site receptors/ residences.

There are a number of noise guidelines for these types of "through the wall" air conditioning units. The Regional Municipality of Waterloo has adopted Ontario Ministry of the Environment and Climate Change (MOECC) Publication NPC-300 guideline limits as applying to the cumulative noise of these units, as well as other noise sources on-site (e.g., make-up air units, exhaust fans, etc.). As a result, for many buildings/ developments, the applicable cumulative noise guideline limits are 45 dBA at night, and 50 dBA during the daytime (L_{eq} (1hr) sound exposures, in dBA).

As a result, for a number of Waterloo development projects, noise mitigation was required, to address noise from PTAC units in existing buildings which were being converted to condominium ownership, and for new construction, where the PTAC systems had already been selected and purchased.

Novus was retained to conduct environmental noise impact assessments for several of these residential developments. This paper discusses the mitigation measures which were developed as a result.

3 PTAC Systems

Noise assessments conducted for the various residential developments under consideration, indicated that noise reductions up to 10 dBA were required. No manufacturer's noise control measures were available.

An examination of the PTAC systems indicated the fan and compressor provided the vast majority of noise emitted to the outdoor environment as shown in **Figure 2**:

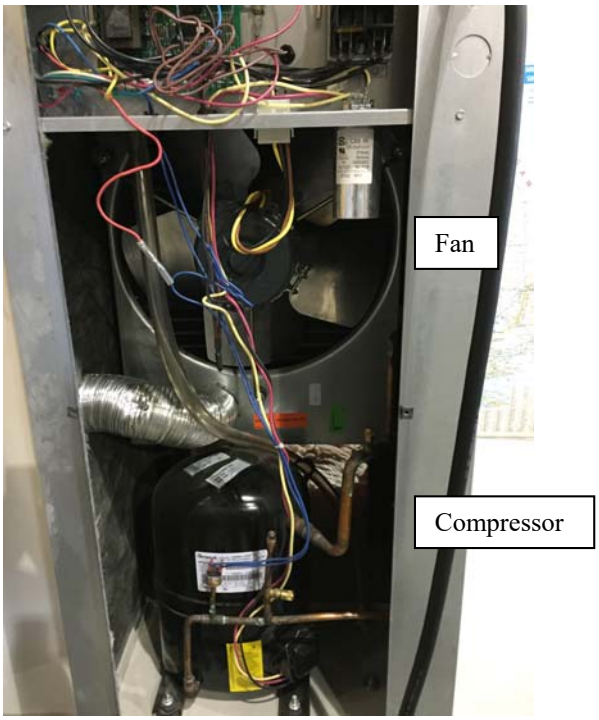


Figure 1: Noise sources in PTAC system

[*] scottp@novusenv.com
[†] lucasa@novusenv.com
[‡] nickw@novusenv.com

Noise is emitted to the environment through an intake air louvre at the bottom of the unit (opposite the compressor), and through an exhaust air louvre at the top of the unit (opposite the propeller fan and condenser).

A primary concern was that the mitigation measures should be able to be installed from inside the building, if possible.

4 Insert Louvres

Parklane Mechanical Acoustics [2] was retained to develop small acoustic louvres which could be inset into the installation sleeve for the unit.

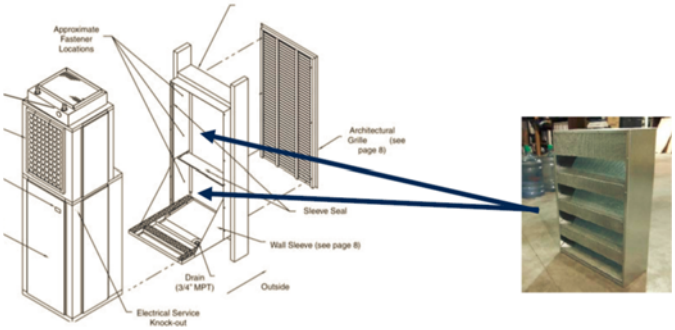


Figure 3: Installation Location for Insert Louvres

These insert louvres were used in conjunction with a compressor noise blanket. Option A placed a louvre in the condenser exhaust air outlet area only. Option B placed louvres in both the exhaust air and intake air openings. The following sound level reductions were achieved:

Option	Octave Band Dynamic Insertion Loss (dB)						
	125	250	500	1000	2000	4000	8000
A	4	4	7	7	6	5	6
B	5	5	8	11	11	14	13

Table 1: Insertion Losses for Acoustic Louvre Options

While the insert louvre solution works, it proved difficult in practice to ensure that the louvres were installed properly. In addition, the solution is costly (~ $ 1,000 per PTAC unit, for Option B). For several residential developments which required lower sound level reductions, other solutions were investigated.

5 Replacement Fans and Compressor Blankets.

Noise from the compressor can be easily reduced through the use of a compressor blanket.

The propeller fan supplied with the unit is shown in **Figure 4**. The supplied fan is an inexpensive, low-efficiency type which can create significant noise. Novus worked with Multi-Wing America [3] to identify a replacement low-noise airfoil/ scroll blade fan with the same flow rates and static pressure requirements.

Example fans were purchased and tested along with compressor noise blankets in a semi-anechoic test chamber at Total HVAC Solutions in Waterloo.



Figure 4: Original and Final Replacement Fan for PTAC Units

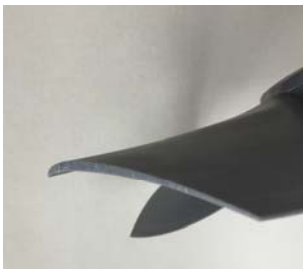


Figure 5: Profile of Replacement Fan Blade

The following sound level reductions were achieved:

Option	Octave Band Dynamic Insertion Loss (dB)						
	125	250	500	1000	2000	4000	8000
C	4	6	5	5	4	9	9

Table 2: Insertion Losses for Fan Replacement Option

The fan replacement solution does not provide the same high level of insertion loss as the Insert Louvre Option B, but is much cheaper (~ $ 200) and easier to install. This solution was optimal for a number of the residential developments under consideration.

Acknowledgments

Thanks to Randy Montag at Total Engineered HVAC Solutions Inc., Greg Downey at Parklane Mechanical Acoustics and David Cisan at Multi-Wing America fans.

References

[1] www.firstco.com/getattachment/Products/Multi-Family-Residential-Products/Single-Package-Vertical-Units/SPXR-X-(cooling-with-electric-heat)-High-Efficienc/spxa-x1215-(1).pdf
[2] www.parklanemechanical.com/
[3] www.multi-wing.net/

PREDICTING EXHAUST SOUND POWER LEVELS OF GENERAL PURPOSE BOILERS

Robert D. Stevens[1] and William J. Gastmeier[2]
HGC Engineering, 2000 Argentia Road, Plaza 1, Suite 203, Mississauga, Ontario L5N 1P7

1 Introduction

Presented herein is a new prediction equation derived empirically from sound power measurements of various boiler exhausts, conducted by the authors between 2003 and 2015, with thermal ratings ranging from about 500 kW to 50 MW. The prediction applies to general purpose boilers, for comfort heating or industrial processes; large-scale power generating boilers and those with induced draft fans or special resonant, "pulse" combustors are not considered here. The new equation agrees well with the measured data, suggesting that it is a suitable estimate when measured data are unavailable.

2 Existing Method from Reference Texts

There are standard reference texts available [1, 2], which purport to provide prediction equations to estimate the sound level for boilers, with thermal ratings in the range of 40 to 1500 kW:

$$L_W = 95 + 4 \cdot \log(bhp), \qquad (1)$$

where L_W is the un-weighted "overall" sound power level in decibels, *bhp* is the thermal rating, in "boiler horsepower" (equivalent to 0.746 times the thermal rating in kW), and "overall" refers to the single-number, spectrally-summed sound power level. To obtain the component sound power levels in octave bands, and the overall A-weighted sound power level, the texts offer a set of adjustment factors, which are to be subtracted from the overall unweighted sound power level, as listed in Table 1, below.

Table 1: Published Octave Band Adjustments to be Subtracted from the Overall Unweighted Level [1, 2]

Frequency [Hz]	Adjustment Factor	Frequency [Hz]	Adjustment Factor
31	6	1k	15
63	6	2k	18
125	7	4k	21
250	9	8k	24
500	12	Overall, A	9

Reference [3] provides similar results, in tabulated form, rather than an equation, and only in terms of overall A-weighted spectral sum sound levels.

In using the predicted sound levels from these three reference texts over time, the authors have noticed that the results generally differed significantly from measured sound power levels of actual boiler exhausts. Toward investigating this apparent discrepancy, the authors have compiled their available measurement results for various boilers gathered between 2003 and 2015, and compared those to the predictions from Equation 1. The measured sound power levels were obtained using sound intensity instrumentation, generally following the methods of ISO Standard 9614-2 [4]. That comparison is shown graphically in Figure 1.

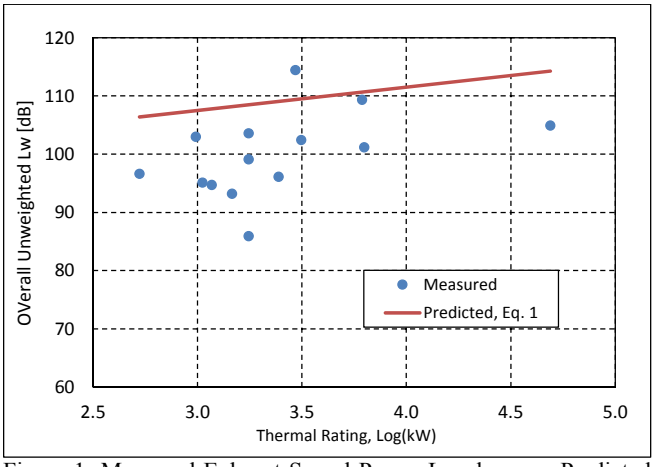


Figure 1: Measured Exhaust Sound Power Level versus Predicted Using Equation 1

The predictions tend to overestimate the sound levels, relative to measured results. The discrepancy between the predictions and measurements ranges in magnitude from 2 to 22 dB. If we define an ensemble-average discrepancy as the logarithmic mean of the magnitude of difference in decibels between the measurements and prediction, the average discrepancy in overall unweighted sound power levels is 14 dB. The average discrepancy on an octave-band basis (comparing the results obtained using the adjustment factors in Table 1 relative to measured values) is 39 dB.

None of references [1] through [3] contains any primary research in regard to predicting the sound from boilers, but instead, each cites earlier works. References [1] and [2] cite a seminal but difficult-to-obtain 1978 publication [5] from the Edison Institute, authored by Bolt Beranek and Newman ("BBN"). Reference [3] cites a 1981 publication by BBN [6]. Both of the BBN texts present the results of original work in deriving an empirical prediction from measured data.

What is apparent explicitly from [5 and 6] is that the prediction is not intended for the sound power emitted from the boiler exhaust outlet, but from the housing of the boiler. References [1] through [3] are unclear in this regard. In fact, reference [3] specifically purports to be a guide for estimating outdoor sound levels in the far field from mechanical equipment. Since general purpose boilers are typically located indoors, such that the exhaust outlet is the only source of sound to the outdoors, it is not surprising that references [1] through [3] could be misinterpreted to apply to the exhaust sound levels. There is a clear need for an equation to predict the sound power level of boiler exhausts.

1 rstevens@hgcengineering.com
2 bgastmeier@hgcengineering.com

3 A New Empirical Prediction

As is common for many types of mechanical equipment, a general trend was evident in the authors' measured data between the logarithm of the power rating (thermal input, in this case) and acoustic power output. A linear regression of this relationship yielded equation 2, below.

$$L_W = 81 + 5.6 \cdot \log(kW), \qquad (2)$$

Figure 2, shows a plot of the measured sound power level versus the logarithm of the thermal input for the fourteen boilers in the data set, along with the predicted sound power level, using Equation 2.

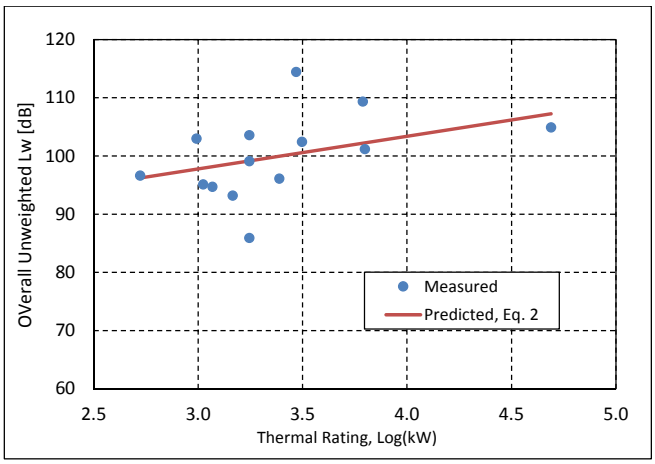


Figure 2: Measured Exhaust Sound Power Level versus Predicted Using Equation 2

The average error between the predicted and measured overall unweighted sound power levels shown in Figure 2 is 7 dB. This average error, while not negligible, certainly represents better correlation than would be achieved using the (albeit inapplicable) Equation 1, with an average error of 14 dB.

Having derived an equation to estimate the overall sound power level of a boiler knowing its thermal rating, it is further possible to derive a spectrum of adjustment factors, to apportion the overall level into its component octave band levels, and the corresponding A-weighted overall level, as was done in [5 and 6]. The results are presented in Table 2. Figure 3 shows the octave band levels used to derive the weightings in Table 2.

Table 2: Octave Band Adjustments for Boiler Exhausts Derived from Measured Data of 14 Boilers

Frequency [Hz]	Adjustment Factor	Frequency [Hz]	Adjustment Factor
31	4	1k	29
63	5	2k	37
125	8	4k	40
250	10	8k	54
500	22	Overall, A	16

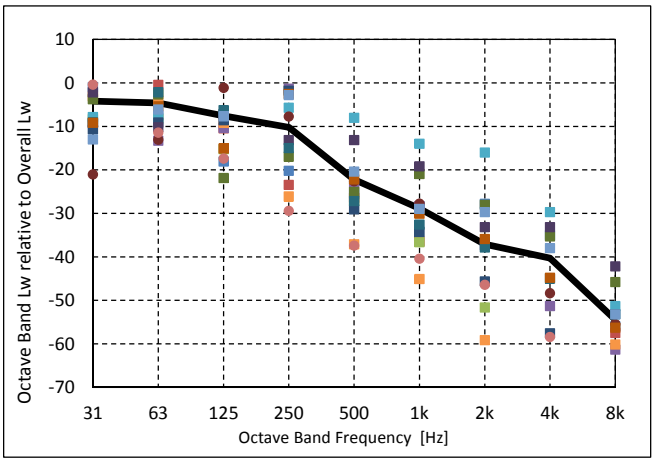


Figure 3: Octave Band Adjustments, Measured and Best-Fit Values

Using Equation 2 and the adjustments in Table 2, the predicted octave band sound levels exhibit an average error of just 11 dB, which is considerably better correlation than would be achieved using the (albeit inapplicable) Equation 1 and Table 1 (with average octave band error 39 dB).

Coincidentally, the data set of 14 boilers measured by the authors is almost the same size as the data set of 13 boilers in the seminal paper by BBN [5], although the current data set spans a slightly greater range of thermal power ratings: 500 kW to 50 MW here, versus 40 kW to 1.5 MW in [5].

4 Conclusion

Equation 2 and the weighting factors in Table 2, provide a reasonable estimate of sound levels from boiler exhausts, when measurements or manufacturer's data are not available.

Acknowledgments

The authors wish to thank W. Dale Stevens, Assistant Professor of Psychology at York University, Toronto, for assistance in obtaining reference [5] via inter-library loan.

References

[1] David A. Bies and Colin H. Hansen, *Engineering Noise Control Theory and Practice*, (2nd Ed.) E & FN Spon, 1996. p. 459-60.

[2] Malcom J. Crocker (Editor), *Encyclopedia of Acoustics*. John Wiley & Sons, Inc. New York, 1997. p. 1049-50.

[3] Ontario Ministry of the Environment Energy (now Ministry of the Environment and Climate Change), "Noise Red Flag Tables." 1997. p. 11.

[4] ISO Standard 9614-2, "Acoustics – Determination of sound power levels of noise sources using sound intensity – Part 2: Measurement by scanning." International Organization for Standardization, 1996.

[5] Bolt Beranek and Newman Inc. *Electric Power Plant Environmental Noise Guide – 2nd Edition*. Edison Electric Institute. Washington, DC, 1st Ed. 1978, 2nd Ed., 1984.

[6] Bolt Beranek and Newman Inc. *Noise Control for Buildings and Manufacturing Plants*. New York. 1981.

NOISE CONTROL FOR HIGH PLUME EXHAUST FANS: A CASE STUDY

Peter VanDelden[*1], Greg Downey[†2], and Matt Downey[§1]
[1] RWDI, 600 Southgate Drive, Guelph, Ontario, Canada N1G 4P6
[2] Parklane Mechanical Acoustics, 1180 Speers Road, Oakville, Ontario, Canada L6L 2X4

1 Introduction

High plume exhaust fans, also known as air induction exhaust fans, are commonly used for exhausting fume hoods. By their nature this type of fan has above-average sound level. Close proximity of these fans to sensitive points of reception can result in requirements for large sound reduction.

2 Background

The intent of air induction exhaust fans is to get contaminated exhaust away from areas that are sensitive to air quality. This includes fresh air intakes that are located on or near the rooftop, where air tends to recirculate. Historically, contaminated exhaust has been removed by the use of stacks that reach high enough that the dispersion is away from the building. More recently, specialty fan systems that have shorter stacks and specialized exhaust nozzles with high exit velocities have been used with the same intent as tall stacks.

The concept of keeping exhaust stacks out of sight appeals to the architectural aesthetic demands on institutions. Tall stacks rising from the roofs to exhaust research or diagnostic testing fume hoods conflict with the intended public image. However, in practice institutions bring the public into closer proximity with air emission sources.

Similarly, institutions have noise sources that are in closer proximity to sensitive areas. Universities, colleges and hospitals are located close to noise sensitive residential areas. Institutions themselves can also be noise sensitive, or have noise sensitive spaces. The proximity of sources to sensitive areas results in more frequent application of abatement, and larger abatement measures to achieve the required criteria.

In this case study the equipment sound levels required more abatement than what was available from the manufacturer in order to achieve the Ontario Ministry of Environment and Climate Change criteria for the facility. A custom-built solution was required to mitigate sound from this complex source. This case, however, required reductions that were more modest that are required elsewhere because the points of reception were 100 m or more away.

[*] peter.vandelden@rwdi.com
[†] greg@parklanemechanical.com
[§] matt@parklanemechanical.com

3 Source Description

The air induction fan in this case is a multi-component system. Exhaust from a number of fume hoods is ducted together, passes through a heat recovery system associated with an air handling unit, and then enters the fan system. The exhaust flow enters the base of the system, which is a plenum. The fans that draw from the plenum achieve a constant airflow, despite variable flow from the fume hoods, by taking additional air into the plenum through bypass dampers. In this case three fans are connected to the plenum. The fans discharge vertically through a special tapered nozzle whose tip is enclosed in a wind band. The nozzle has a high exit velocity. The nozzle and wind band are together intended to induce additional diluting airflow to the exhaust air stream. A diagram of the system prior to the installation of noise abatement is provided below.

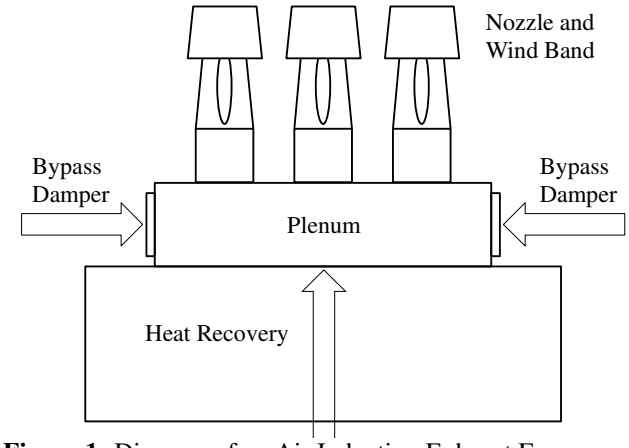


Figure 1: Diagram of an Air Induction Exhaust Fan

The system in this case study uses axial fans. Several other manufacturers use centrifugal fans. Despite the variability in design, the systems have similar key sources of sound emission to the outdoor environment: the nozzle outlet, the bypass louvre(s), and radiation from the casing. Inherent to the design is the high flow velocity at the exit and the resultant higher sound levels.

Sound level data provided by the fan manufacturer was supplemented by measurements of the system components, as installed. Manufacturer data and measurements provided nozzle outlet and bypass louvre sound levels, but were not able to isolate casing-radiated sound.

4 Criteria

Motivation for implementing sound level reductions was primarily driven by the regulator. This installation is in Ontario, where the Ministry of Environment and Climate Change specifies criteria which must be met to receive an operating approval. It is the cumulative sound level from all sources at the facility which is required to meet the applicable criteria at nearby sensitive locations. Because numerous sources at the facility contribute to the cumulative sound level, the individual contribution of each source must be significantly below the criteria. Each individual emission point was therefore limited to a contribution between 30 and 35 dBA.

5 Analysis

Modelling of the sound level contributions at points of reception was done in the Cadna/A software package. The influence of buildings between the sources and points of reception was thus considered.

Minimum sound level reductions of approximately 15 dB were required on the nozzle outlets, and 10 dB on the bypass louvres. This level of reduction is towards the upper limit of what could be achieved by acoustic barriers. The negative influence of barriers on the air dispersion as well as the structural loading from such large elevated surfaces further discouraged the use of barriers.

The casing radiated sound was expected to become a secondary source once sound from the nozzle outlets was reduced. The casing was therefore also identified for mitigation.

6 Mitigation Measures

The specified sound level reductions exceeded the reductions that could be provided by standard abatement equipment supplied by the fan manufacturer. A firm focussed on design, construction and installation of custom noise abatement solutions was retained to complete the project. The package of equipment provided by Parklane Mechanical Acoustics included outlet silencers, bypass silencers, and an enclosure to address the casing noise.

To ensure that the function of the nozzles and wind bands would be unchanged, the decision was made to install the outlet silencers between the fans and nozzles. This placement meant that the silencers would address sound generated by the motor and fan but not any sound caused by the nozzle and wind band. Achieving an overall reduction of approximately 15 dB would therefore require the motor noise to be mitigated more in compensation for the nozzle source that could not be addressed. In-situ measurements were not able to separate motor noise from that of the nozzle. Silencer dynamic insertion losses were therefore planned to be significantly more than the requested 15 dB reduction.

At the same time, the silencers' internal construction was not allowed to degrade the performance of the fans or the function of the nozzles and wind bands. They were therefore designed to ensure that flow to the nozzles was uninterrupted and that there was only a very small pressure drop across the silencers.

The additional weight introduced by the mitigation measures was supported independently of the fan and plenum structure. A photo of the installation is provided in Figure 2.



Figure 2: Noise Abatement Measures Installed on an Air Induction Exhaust Fan System

7 Verification

Sound levels with the noise abatement equipment in place were verified once the system was back in operation. Casing radiated noise from the fans was inaudible. Sound levels at the nozzle were 15 dB quieter, as specified. The bypass louvre sound was reduced by more than the minimum, with the silencers achieving 15 dB.

In addition to the achievement of regulatory compliance requirements, these mitigation measures improved the acoustic environment for pedestrians in the area.

8 Conclusions

An air induction fan, also known as a high plume exhaust fan, is a more complex type of noise source that frequently needs mitigation when installed at locations that are close to sensitive locations. In this case a mitigation strategy was developed to provide more modest reductions of 15 dB without the use of acoustic barriers. In-situ measurements verified that the design developed for this application achieved the required reductions.

Laboratory Impact And Airborne Sound Attenuation In Cross-Laminated Timber Construction Using Resiliently Mounted Ceilings And Poured Concrete Toppings.

Wilson Byrick

Cross laminated timber (CLT) construction is a wood wall and floor-ceiling construction method that enables the architect to design tall buildings made of renewable materials. The assemblies are constructed by laminating multiple layers of wood sheets together under pressure to create quasi-monolithic wood slabs. In 2014 we began laboratory experimentation on a 175mm CLT floor assembly, evaluating the efficacy of various floor build ups when tested according to ASTM E90 and ASTM E492. These test procedures evaluate sound transmission loss and impact insulation respectively. These floor build-up results were presented at the proceedings of Internoise 2015. Next steps were outlined in said paper and testing was completed in fall 2015. Data from the follow-up test series are presented here including quantifying the effect of various resiliently mounted ceiling assemblies with and without floor finish materials. The performance of poured concrete toppings of various densities and thicknesses was also evaluated.

ACOUSTICS
NOISE
VIBRATION

HGC ENGINEERING

> Noise & Vibration Control in Land-use Planning
> Noise & Vibration Studies: Residential and Commercial
> Building Acoustics, Noise & Vibration Control
> Land-use Compatibility Assessments
> Third-party Review of Peer Reports
> Expert Witness Testimony

905-826-4546
answers@hgcengineering.com
www.hgcengineering.com

ACOUSTICAL MEASUREMENTS - MESURES ACOUSTIQUES

Time Domain Source Localization Technique Based On Generalized Cross Correlation And Generalized Mean
Thomas Padois, Olivier Doutres, Franck Sgard, Alain Berry 60

Colt C8a2 Carbine Impulsive Noise Auditory Hazard Assessment Through Testing In A Reverberant Environment
Sebastian Ghinet, Andrew Price, Anant Grewal, Yong Chen, Viresh Wickramasinghe 62

Development Of The Second Generation Nrc Acoustic Spectrum Control System For High Intensity Noise Testing
Anant Grewal, Yong (Eric) Chen, Shahrukh Alavi, Viresh Wickramasinghe, Brent Lawrie 64

Identifying Localized Noise Sources During Industrial Vehicle Passbys
Jordan Michael Reniak 66

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 68

TIME DOMAIN SOURCE LOCALIZATION TECHNIQUE BASED ON GENERALIZED CROSS CORRELATION AND GENERALIZED MEAN

Thomas Padois [*1], **Olivier Doutres**[†1], **Franck Sgard**[‡2], **and Alain Berry** [§3]
[1]Département d'ingénierie mécanique, École de Technologie Supérieure, Montréal, Québec, H3C 1K3, Canada
[2]Institut de Recherche Robert-Sauvé en Santé et en Sécurité du Travail, Montréal, Québec, H3A 3C2, Canada
[3]Groupe d'Acoustique de l'Université de Sherbrooke, Département d'ingénierie mécanique, Université de Sherbrooke, Sherbrooke, Québec, J1K 2R1, Canada

1 Introduction

Microphone arrays are an efficient tool for localizing acoustic sources [1–3]. The acoustic field is recorded by a set of distributed microphones. Then the microphone signals are processed in order to build a noise source map which highlights the source positions. The standard technique is the Delay And Sum (DAS) beamforming which can be performed in the frequency domain via the cross-spectral matrix [1, 2] or in the time domain via the cross correlation of the microphone signals [3]. In the frequency domain, DAS beamforming is known to be less efficient at low frequencies, therefore several techniques based on inverse methods have been developed to improve its performances [1, 3]. In this study, DAS beamforming is implemented with the generalized cross correlation in the time domain. The arithmetic, geometric or harmonic means of the spatial likelihood functions are used similarly to Padois *et al.* [4]. The influence of the number of microphones is investigated with numerical data in the case of a standard circular microphone array.

2 Source localization method

Generalized cross correlation of the microphone signals

The cross correlation $\mathrm{R}_{x_m x_n}(\tau)$ is a useful function to estimate the time delay between two microphone signals x_m and x_n. Commonly in microphone array processing, the cross correlation function between two signals is obtained by the inverse Fast Fourier Transform of the cross-spectrum $C_{x_m x_n}$ which allows prefiltering operation

$$\mathrm{R}_{x_m x_n}(\tau) = \sum_{k=0}^{N_f-1} W(k) C_{x_m x_n}(k) \exp\left(j2\pi \frac{k}{Nf}\tau\right), \quad (1)$$

where $W(k)$ is the prefilter and k is the frequency index. The technique used to prefilter is the PHAse Transform (PHAT) which allows removing the magnitude of the cross-spectrum in order to only retain the phase information. This technique is named the Generalized Cross-Correlation (GCC) or GCC-PHAT when the PHAT prefiltering operation is used (in the following the PHAT prefilter is used but the term PHAT is omitted for simplicity).

*Thomas.Padois@etsmtl.ca
†olivier.doutres@etsmtl.ca
‡franck.sgard@irsst.qc.ca
§alain.berry@usherbrooke.ca

Spatial Likelihood Function of a microphone pair

The peak value of the cross correlation provides the time delay between the microphone signals. This time delay corresponds to a set of potential source positions. If the cross correlation function is interpolated over spatial locations $\mathbf{r}_l$ where the source is searched, the resulting map is typically a hyperbola in two dimensions, this map is called Spatial Likelihood Function (SLF). However, the source position cannot be accurately detected with a single hyperbola (i.e a single microphone pair). If three microphones are used, three hyperbolas can be defined. If the SLFs are summed, the maximum value corresponds to the intersection of the hyperbolas and provides the source position. Commonly, the microphone array output signal $y^{AM}(\mathbf{r}_l)$ is obtained by computing the Arithmetic Mean (AM) of the SLFs determined for M_p microphone pairs as

$$y^{AM}(\mathbf{r}_l) = \frac{1}{M_p} \sum_{p=1}^{M_p} \mathrm{R}_p(\tau_{\mathbf{r}_l}), \quad (2)$$

where R_p is the cross correlation function of microphone pair p and $\tau_{\mathbf{r}_l} = \Delta t_{ml} - \Delta t_{nl}$ is the time lag for the spatial source location l.

Generalized mean of the SLFs

In the case of the GCC and M_p microphone pairs, the geometric mean (GM) and harmonic mean (HM) [4] are given by

$$y^{GM}(\mathbf{r}_l) = \left(\prod_{p=1}^{M_p} \left| \mathrm{R}_p(\tau_{\mathbf{r}_l}) \right| \right)^{\frac{1}{M_p}}. \quad (3)$$

$$y^{HM}(\mathbf{r}_l) = M_p \left(\sum_{p=1}^{M_p} \frac{1}{\left| \mathrm{R}_p(\tau_{\mathbf{r}_l}) \right|} \right)^{-1}. \quad (4)$$

where $|\cdot|$ denotes the absolute value. The influence of the number of microphones, therefore the number of microphone pairs, is investigated with numerical data in the following section.

3 Influence of the number of microphones on the noise source map

Numerical set-up: case of a standard circular microphone array in front of an acoustic point source

An acoustic point source, arbitrary located at $(x = -0.25, y = 0, z = 0.25)$ m, generates a white noise signal.

A standard circular array with a radius of 20 cm is set in front of the source 1 m away. The center of the circular array is located at $(x = 0, y = 1, z = 0)$ m. Therefore, the distance source-array is 1 m and the source is outside the inner circle of the array. This kind of array is commercially used and the geometry is convenient for investigating the decreasing of number of microphones. The time delay between each microphone pair signal is computed via Eq.1. The noise source maps are obtained by using the AM (Eq.2), GM (Eq.3) and HM (Eq.4). The scan zone is a square with side equal to 1 m and discretized with 2601 points (51×51). The only difference between the noise source maps is the combination of the SLFs.

Noise source maps: case of a 20-microphone circular array

The noise source maps provided by the three different means are shown in Fig 1. With the AM, the source position is correctly detected. However, the number of side lobes is very large due to the number of microphones. Moreover, the side lobes level is high which may prevent the localization of a source with a weaker level. When the GM and HM are used, all the side lobes are removed and the main lobe is narrower. The best noise source map is provided by the HM with a sharp main lobe.

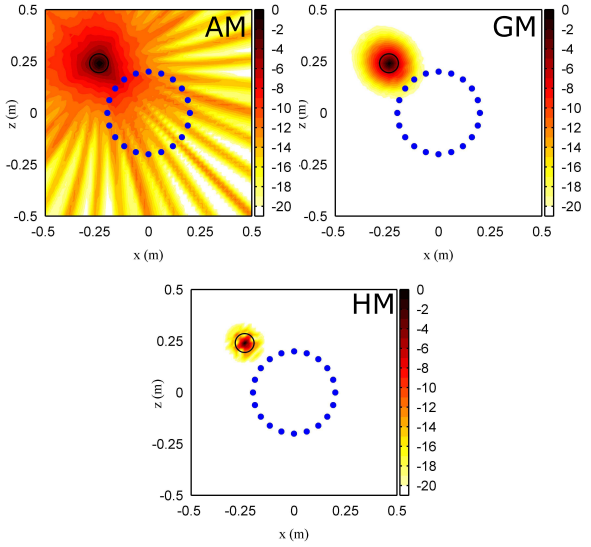


Figure 1: Noise source maps obtained with the AM, GM, and HM in the case of a 20-microphone circular array. The black circles is the source position. The blue dots are the microphone positions. The colorbar is in dB.

Noise source maps: case of a 5-microphone circular array

Now, the number of microphones is reduced to 5. The noise source maps provided by the three different means are shown in Fig 2. With the AM, the number of side lobes is smaller due to the decreasing of microphone pairs. However, the side lobes level is higher because there is less destructive interferences. With the GM and HM, the noise source maps are similar to Fig 1 without side lobes and with a sharp main lobe.

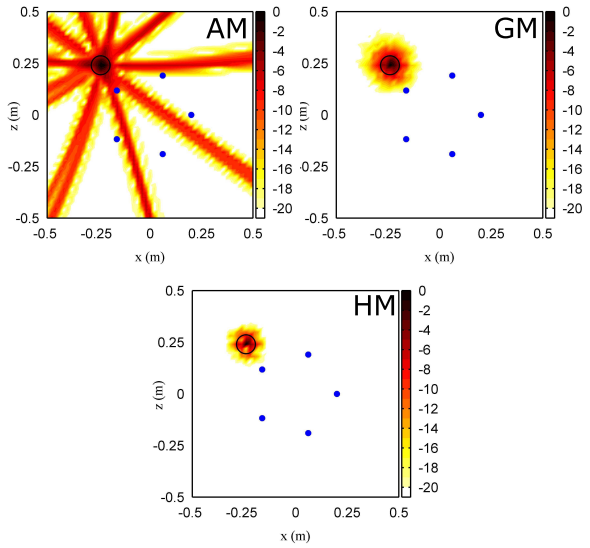


Figure 2: Noise source maps obtained with the AM, GM, and HM in the case of a 5-microphone circular array. The black circles is the source position. The blue dots are the microphone positions. The colorbar is in dB.

4 Conclusions

This paper deals with a source localization technique based on the generalized cross correlation in the time domain. Commonly, the Arithmetic Mean (AM) is used to average the spatial likelihood functions. In this study, the AM has been replaced by the Geometrical and Harmonic Mean (GM and HM). Two numerical cases involving a large or a small number of microphones were investigated. In each case, the GM and HM outperform the noise source map provided by the AM by removing the side lobes and narrowing the main lobe. Moreover, the GM and HM are less sensitive to the decrease of number of microphone as compared to the AM.

References

[1] T. Padois, P-A. Gauthier and A. Berry. *Inverse problem with beamforming regularization matrix applied to sound source localization in closed wind-tunnel using microphone array*, Journal of Sound and Vibration, Vol.**330(25)**, pages 6858-6868, 2014.

[2] T. Padois and A. Berry. *Orthogonal matching pursuit applied to the deconvolution approach for the mapping of acoustic sources inverse problem*, J. Acoust. Soc. Am., Vol.**138(6)**, pages 3678-3685, 2015.

[3] T. Padois, F. Sgard, O. Doutres and A. Berry. *Comparison of acoustic source localization methods in time domain using sparsity constraints*, Proceedings of Internoise, 9-12 August,San Francisco, CA, USA, pages 1-10, 2015.

[4] T. Padois, O. Doutres, F. Sgard, and A. Berry. *On the use of geometric and harmonic means with the generalized cross-correlation in the time domain to improve noise source maps*, J. Acoust. Soc. Am, Vol.**140(1)**, pages EL56-EL61, 2016.

COLT C8A2 CARBINE IMPULSIVE NOISE AUDITORY HAZARD ASSESSMENT THROUGH TESTING IN A REVERBERANT ENVIRONMENT

Sebastian Ghinet*, Andrew Price, Anant Grewal, Yong Chen, and Viresh Wickramasinghe
NRC-Aerospace, Flight Research Laboratory.

1 Introduction

Exposure to high levels of gunfire impulsive noise is potentially hazardous to human hearing and may lead to permanent auditory as well as non-auditory damage in the unprotected ear. A client approached NRC Aerospace to address a concern for potential hearing damage from discharging of firearms within enclosed spaces, characterized as reverberant acoustic environments.

In order to address this concern, a reconfigurable room was built by NRC on a shooting range and this test setup was used to measure the impulsive noise time trace waveforms. The measured waveforms were analyzed in accordance with the updated standard procedure MIL-STD-1474E (revision 15 April 2015) [1] using the algorithm suggested by the standard: Auditory Hazard Assessment Algorithm for Humans (AHAAH) [2]. This algorithm enabled the assessment of the auditory hazard risk to which personnel would be exposed during a Colt C8A2 carbine discharge. Three different geometry and floor area configurations were considered for the room. Moreover two different window configurations where considered, namely: a) armored window with a narrow slit gun port opening surrounded by ballistic glass and b) wide open large window.

The paper presents the results and conclusions of the data analysis of the testing campaign aimed at evaluating the noise exposure and assessing the auditory hazard of personnel using the Colt C8A2 carbine.

2 Measurement procedure

The test procedure developed for this study took into consideration several technical challenges related to the high complexity and specific technical requirements.

Most of the impulsive noise measurements presented in literature are conducted in free field conditions such as a shooting range, which provide ideal conditions. Depending on the nature of the study in the literature, a part of the measurements was conducted using a firearm fixture while other studies were conducted with a human subject operating the firearm [3].

The present work differentiates from any previous studies because the major part of the complexity of the problem arose from the large number of shock wave reflections involved in this measurement procedure. In the present work the emphasis was on determining the importance of the shock wave reflections and influence of reverberation behavior of the partially enclosed room on the auditory hazard assessment.

In the present setup the spherical shock wave centered on the carbine mussel (situated outside the control post) was expected to impinge the ballistic glass of the window and penetrate through the 2-inch wide gun port slit opening. The diffracted component, through the slit opening, of the shock wave was expected to propagate towards the shooter's body. One part of the wave was expected to impact the body and reflect back towards the window. The shock wave was expected to experience a number of reflections between the shooter's body (head, arms and torso in shooting representative position) and the window/wall in front of the shooter. A second part of the shock wave was expected to propagate inside the room and experience the reverberation behavior of the room.

In order to provide a realistic testing environment and address the challenging test specifications, NRC proposed to design and build a reconfigurable reverberant room on the premises of a shooting range. Impulsive noise measurements were performed on the shooting range of the Eastern Ontario Shooting Club (EOSC) in Cheney, Ontario.

2.1 Configurable room

A configurable room as shown in Figure 1, having the footprint and floor area of representative small (green), medium (red) and large (blue) size control posts was built on one of the long range EOSC shooting ranges. The small size room had a hexagonal shape and a floor area of $8m^2$. The medium size room had a polygonal 6 sided shape and a floor area of $13.5m^2$. The large size room had an octagonal shape and a floor area of $18m^2$. The three room configurations and the wall dimensions are shown in Figure 1 (a=1.93m; aa=1.93m; b=2m; c=2.44m; d=2.22m). All walls of the rooms were 2.44m in height.

2.2 Instrumentation

The impulse noise waveform record duration was 0.5s. The data acquisition system was set to record 50msec of data prior to the trigger. The sampling rate of the recorded signal throughout the present measurement campaign was 1 M-Samples/s.

The data acquisition system was composed of a National Instruments (NI) front-end (NI PXIe-8840 Core i5-4400E 2.7GHz, Dual Core, Win 7 (64)-bit) with a high sampling rate data acquisition card PXIe-6124 Simultaneous Sampling Multifunction DAQ which provided 4 simultaneously sampled analog inputs, at up to 4 MS/s per channel, 16 bits of resolution, and a PXI Express interface for dedicated bandwidth from module to controller. The NI hardware was used in conjunction with an in-house developed LabVIEW computer program application with microphone calibration capability as well as trigger control. The microphone used in the present measurement campaign

*Sebastian.Ghinet@nrc-cnrc.gc.ca

was a 1/8-inch GRAS Type 46DP with a frequency range of 6.5Hz – 140KHz, a dynamic range of 46dB(A) – 191dB and a sensitivity of 0.85mV/Pa.

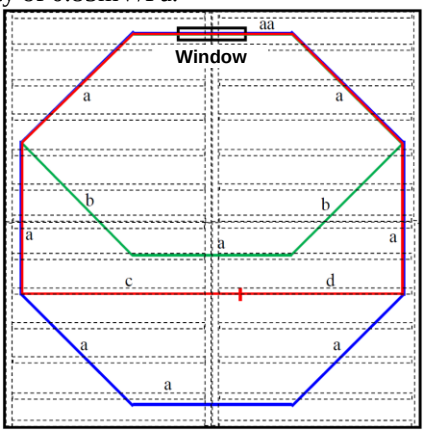


Figure 1. The three configurations of the room: Green (hexagon); Red (polygon, 6 sides); Blue (octagon),

3 Results

3.1 Free-field

Measurement configurations conducted in a free field condition were performed to facilitate comparisons and highlight specific behaviours governing the results measured within the room. Those measurements were conducted with the microphone placed at the following locations: 2m from the carbine at the height and azimuth position of the muzzle - the peak impulsive noise level Lpk was 162.8dBP; 0.5m from the carbine at the height and azimuth position of the front sight of the carbine (0.6m from shooter's right ear) - Lpk was 173dBP; 0.5m from the carbine at the height and azimuth position at the half length of the carbine (0.38m from shooter's right ear) - Lpk was 168.5dBP; 0.5m from the carbine at the height and azimuth position of the shooter's right ear - Lpk was 163dBP; 0.1m from the shooter's left ear - Lpk was 160dBP and 0.1m laterally from the shooter's right ear - Lpk was 159dBP.

An example of free field time-domain waveform measured at a lateral position 0.5m from the carbine axis and 0.6m from shooter's ear (front sight of the carbine) is shown in Figure 2.

3.2 Reverberant Environment

For each of the three configurations of the room, the impulsive noise wave pressure was measured at a distance of 0.1m from the right ear and left ear of a shooter. As an example, the time-domain waveform measured at 0.1m from the right ear of the shooter in room Configuration A is shown in Figure 3.

Using the AHAAH algorithm, the estimated number of allowed occasional exposures (single shots), within a 24-hour sliding window period, without wearing a hearing protection, was found to be very limited e.g. maximum of 3 (three) rounds for two smaller control posts configurations A and B and 4 (four) rounds for the largest control post configuration C. Exceeding these limits may result in noise induced hearing loss (NIHL) as per the MIL-STD-1474E standard.

4 Conclusion

The auditory hazard risk for personnel exposed to gunfire noise as a result of discharging a Colt C8A2 carbine from within a reverberant environment of a room was assessed in this study. The measured waveforms were analyzed in accordance with the updated standard procedure MIL-STD-1474E using the algorithm suggested by the standard: Auditory Hazard Assessment Algorithm for Humans (AHAAH). The data analysis provided the estimated number of allowed occasional (no more than once per week) impulsive noise exposures, within any 24-hour sliding window period.

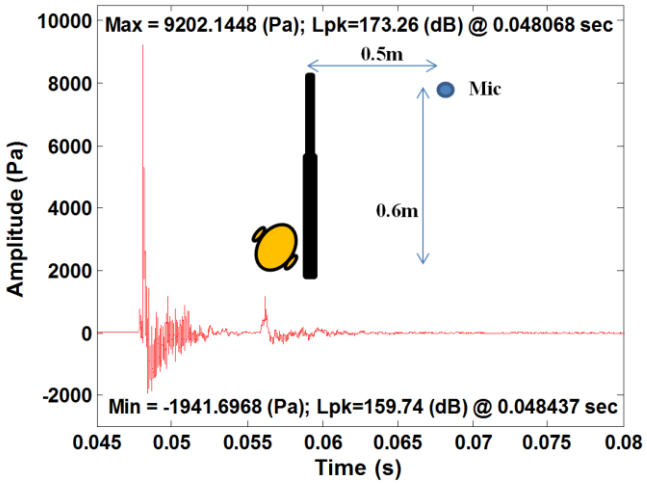


Figure 2. Free field time-domain waveform measured at a lateral position 0.5m from the carbine axis and 0.6m from shooter's ear

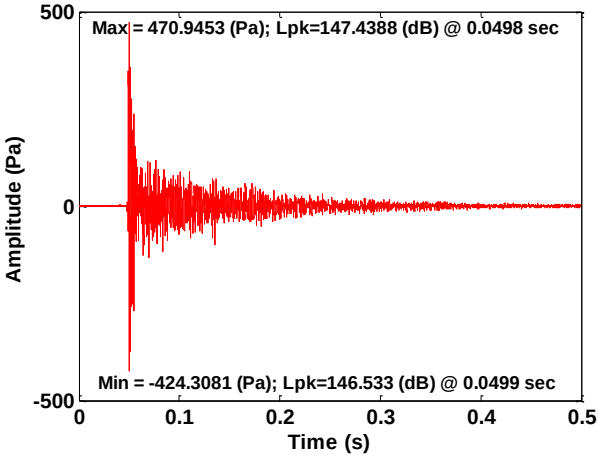


Figure 3. Time-domain waveform measured at 0.1m from the right ear of the shooter in Control Post Configuration A.

References

[1] MIL-STD-1474E, "Department of Defense Design Criteria Standard Noise Limits".

[2] G. Richard Price, "The Auditory Hazard Assessment Algorithm for Humans (AHAAH): Hazard Evaluation of Intense Sounds", July 2011, ARL-TR-5587.

[3] P. Rasmusen, G. Flamme, M. Stewart, D. Meinke, J. Lankford, "Measuring Recreational Firearm Noise", Sound & Vibration, August 2009, pp14-18.

DEVELOPMENT OF THE SECOND GENERATION NRC ACOUSTIC SPECTRUM CONTROL SYSTEM FOR HIGH INTENSITY NOISE TESTING

Anant Grewal[*1], Yong (Eric) Chen[†1], Shahrukh Alavi[‡2], Viresh Wickramasinghe[§1], and Brent Lawrie[¶3]
[1]Research Officer
[2]Flight Test Instrumentation Engineer
[3]Senior Technical Officer
Flight Research Laboratory, National Research Council Canada, 1200 Montreal Road, Ottawa, Canada, K1A-0R6.

1 Introduction

Since 1982, NRC has conducted high-intensity diffuse field noise testing using a reverberant chamber that measures 6.9 x 9.75 x 8m. This is the only facility in Canada capable of supporting full-size satellite and large spacecraft component testing to ensure that they withstand exposure to the intense acoustic launch environment (see Figure 1). Within this chamber, testing at overall SPLs of greater than 150dB with accurate spectrum shaping is routinely performed.

Until 1993, the target acoustic spectrum for test articles was achieved through the real-time manual adjustment of ⅓-octave spectrum shapers. Over time, with ever tighter target tolerances, the ability of human operators to achieve the target spectrum over the duration of tests became severely challenged. In response, NRC developed an in-house closed-loop system for spectrum control. The key components of the first generation controller were a B&K-2131 spectrum analyzer, a Norsonic-731 noise-shaper, and a PC executing the control algorithm. Data flow between the analyzer, computer and shaper was achieved using the IEEE-488 interface (see Figure 2). The system worked extremely well, providing accurate and reliable performance, and was used in the testing of major satellites (RADARSAT-1 and -2, and CASSIOPE amongst others).

In order to eliminate the risk due to the obsolescence of ageing hardware and the reliance on an unsupported operating system and development environment, a plan was developed to migrate the control algorithm to a current generation of hardware and software. After a review of available hardware and software, the National Instruments PXI and Labview-RT development system was chosen for the new spectrum control platform. A control system that provided the baseline functionality of the first generation controller with additional features and performance was developed and benchmarked against the original system as well as two commercial acoustic control systems.

2 Spacecraft Diffuse Field Testing

Diffuse field acoustic testing in a reverberant chamber is the principal method used to qualify satellites, as well as their components against the intense acoustic environment present during launch. This approach relies on an accurate re-creation of the launch noise levels in the chamber. Specifications are typically provided in ⅓-octave levels between the 25 Hz and 10 kHz band, and a typical tests lasts between 1 and 2 mins.



Figure 1: The RADARSAT-2 Satellite in the NRC Chamber.

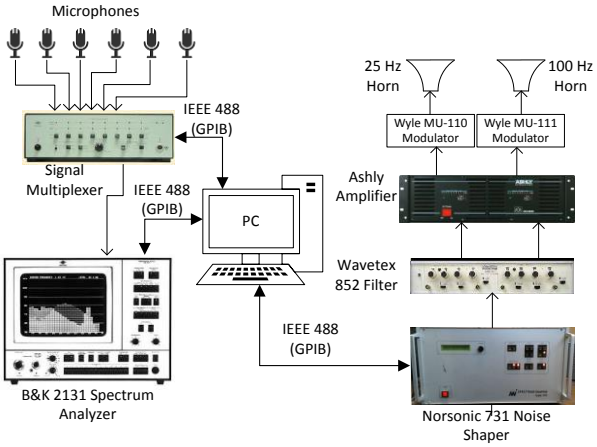


Figure 2: Schematic of the First Generation NRC Acoustic Spectrum Control System.

3 First Generation NRC Control System

The first generation system architecture and algorithm were developed by NRC staff and the code development was contracted out. In the NRC chamber, noise generation was provided using Wyle WAS-3000 (MU-110/111) airstream modulators for frequencies lower than approximately 1600 Hz. Additional broadband noise between approximately 1 and 10 kHz was generated using a pair of opposed gas jets.

Time-domain multiplexing of 6 microphones positioned

* anant.grewal@nrc-cnrc.gc.ca
† eric.chen@nrc-cnrc.gc.ca
‡ shahrukh.alavi@nrc-cnrc.gc.ca
§ viresh.wickramasinghe@nrc-cnrc.gc.ca
¶ brent.lawrie@nrc-cnrc.gc.ca

around the test article was employed to obtain an averaged noise signal. A B&K analyzer continuously performed ⅓-octave analysis using exponential averaging. Every 0.2 seconds, the current spectrum measurement was transferred to the PC via IEEE-488. In addition to executing the overall supervisory level of control, the PC implemented an integral control loop for each ⅓-octave band between 25 Hz and 1600 Hz to determine the required level of ⅓-octave band attenuation that were communicated to the shaper via the IEEE-488 interface. The output of the shaper was passed through a Low Pass, High Pass Filter network configured using a programmable Wavetek 852 filter. The signals were amplified using a dual-channel Ashly audio amplifier and applied to a pair of Wyle modulator coupled to horns.

4 Second Generation NRC Control System

When the decision was made to replace the first generation controller, a number of options were considered including two commercial controllers as well as a array of in-house development options. Ultimately, the in-house option using the Labview RT development environment and National Instruments hardware option was chosen. Many features of the 1st generation system were not available on the commercial systems, nor were a number of enhancements and improvements that resulted from over 25 years of operational experience. The Labview RT environment relies on an intuitive and powerful graphical programming language that permits rapid prototyping and development with excellent real-time performance. The signal analysis, control algorithm, noise generation and cross-over network functions were all implemented within an NI PXI chassis that contained a Pentium i7 processor module along with 24-bit analog input and output modules. The main features are given in Table 1. An early version of the controller was used to perform closed loop control of acoustic modulators from TEAM Corporation in support of a project to design the NASA Glenn Research Center's Reverberant Acoustic Test Facility [1].

5 Results

The 2nd generation control system performed very well during a comprehensive validation and commissioning phase and is currently in use for all tests. During commissioning, comparisons were made between the old and new control systems for single and dual horn cases. The errors between the target and measured levels for all bands for the new controller were comparable to those obtained with the old controller in all cases tested. Furthermore, the transition from pre-test to full-test levels exhibited negligible overshoot, and remained steady throughout the test duration. An example of the overall SPL time history for a test is shown in Figure 3. One of the new features of the new controller is the ability of the operator to toggle between automatic and manual control for individual 1/3-octave bands. Another is the ability to specify the controller bandwidth to match the frequency response of the modulator. For example, the new controller can be used to control the Wyle WAS-5000 modulator, which has a frequency range of 25 Hz to 5 kHz. This would not have been possible with the old controller.

Table 1: Salient Features of Acoustic Control Systems

1st Generation NRC System	2nd Generation NRC System	Commercial Systems
External crossover required for multi-horn control.	Integ S/W multi-horn control via crossover or direct mapping.	External crossover required for multi-horn control.
Obsolete Hardware and Software Platform	Proven track record of closed-loop control. H/W & S/W supplier is well-entrenched.	Dependent on continual support from control system vendor
"Slam" start for quick convergence	"Slam" start for quick convergence	No "slam" start capability
Fixed control range (25 Hz to 1.6 kHz)	Arbitrary control range (25 Hz-10 kHz)	Arbitrary control range (25 Hz-10 kHz)
Time-domain multiplexing for multi-microphone averaging.	Various multi-mic averaging strategies (min, max, spectral ave, time multiplex)	Unknown

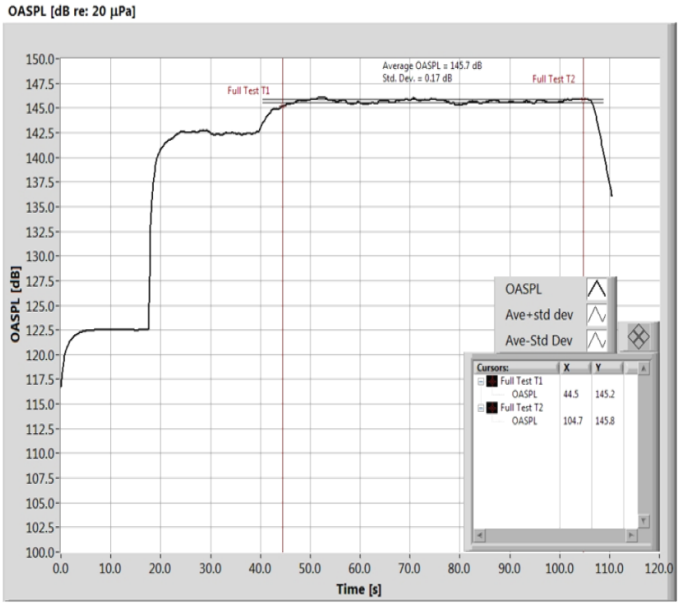


Figure 3: Time history of the OASPL for a typical test.

6 Conclusions

The original acoustic spectrum control system developed in the 1990s for the NRC reverberant chamber has been upgraded using National Instruments real-time hardware. A control algorithm, encompassing all the features of the old controller along with important enhancements, has been implemented in the Labview RT environment. The performance of the new controller is similar to the old one in terms of accuracy and transient behaviour, and is currently being used for testing of client space hardware.

References

[1] Grewal, A., Ramakrishnan, R., Hughes, W.O., Woyski, B., Elfstrom, G., Mech, C., and Chen, Y., "High Intensity Noise Generation for Extremely Large Reverberant Room Test Applications," Paper 119, IMAC XXIX A Conference and Exposition on Structural Dynamics, 31 Jan – 3 Feb 2011, Jacksonville, Florida.

IDENTIFYING LOCALIZED NOISE SOURCES DURING INDUSTRIAL VEHICLE PASS-BYS

Jordan Reniak [‡1]
[1]BKL Consultants Ltd., Vancouver, British Columbia, Canada

1 Introduction

Certain sections of roads will invoke certain noises from industrial transport vehicles. In our study, we were specifically looking to determine how the localized noise sources on an industrial transport vehicle contribute to roadside pass-by noise and to locate the source of trailer clanging noise from unloaded trailers passing over uneven road sections. Using the Norsonic Nor848 acoustic camera, we used representative snapshots of roadside truck pass-bys to identify the location and contribution of localized noise sources contribute to roadside pass-by noise. From our measurements, we found that tire noise made up 81% of the roadside pass-by noise during coasting, and 26% of noise during acceleration. Engine noise contributed to 66% of roadside pass-by noise during acceleration and 13% of roadside pass-by noise during coasting pass-bys. We also found out that exhaust noise contributed between 6-8% of the total roadside pass-by noise. Using the acoustic camera, we were also able to determine that engine noise received from the accelerating trucks comes directly from a reflection off the roadway directly beneath the engine housing. This means that the majority of the noise received at the roadside from truck pass-bys comes from ground level noise sources (engine and tire noise) which, could be an important consideration when determining appropriate noise mitigation.

2 Method

2.1 Location

The studied roadway section included both even and uneven lengths of roadway. We used the even section of roadway to determine the composition of the roadside pass-by noise from noise sources such as engine, tires, exhaust etc. We studied the uneven length of roadway to locate and identify the source of the impulsive trailer clanging noise. Both roadway sections had a 50 km/hr speed limit.

2.2 Equipment

To determine and quantify the noise sources from industrial transport vehicles, we used a Norsonic Nor848 acoustic camera (the acoustic camera). The acoustic camera is a one-metre diameter solid disc that is embedded with 256 microphones and is paired with software developed by Norsonic to process the data from the microphone array. The software coupled with the acoustic camera gives the user the capability to view and capture live noise events, and to review and study logged events.

‡ reniak@bkl.ca

3 Measurement

We recorded several different scenarios, which included accelerating and coasting trucks on even (flat) pavement, and loaded and unloaded trucks on uneven pavement. Recording these scenarios enabled us to determine how the loading of an industrial truck affected the noise emission of the vehicle during typical roadside pass-bys.

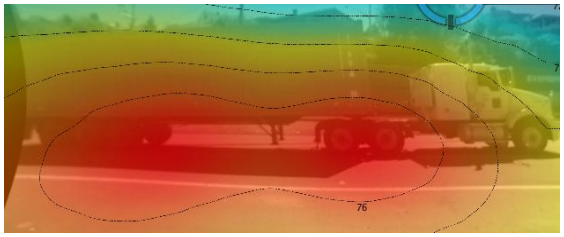


Figure 1: Showing a coasting roadside pass-by

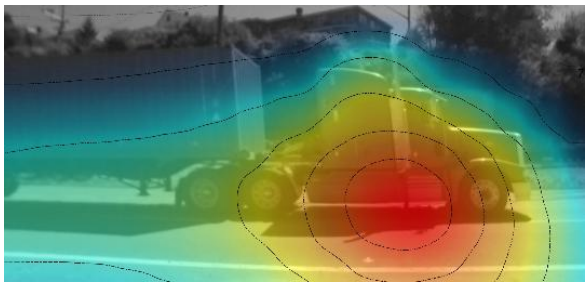


Figure 2: Accelerating truck pass-by showing engine noise

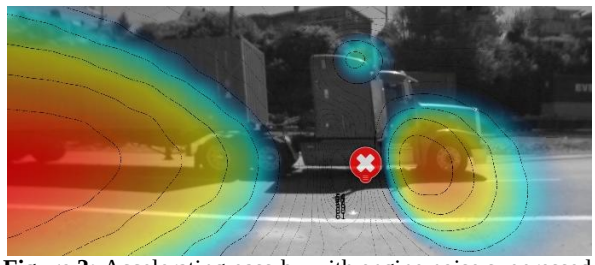


Figure 3: Accelerating pass-by with engine noise suppressed

3.1 Calculation

The Norsonic software paired to the acoustic camera provides the user with the measured sound pressure level, as well as the ability to determine the sound pressure level received from localized sources within the cameras field of view. Using the snapshots captured from the acoustic camera, we were able to approximate how the different noise sources contributed to the vehicle pass-by noise. Our approximations were based on a captured snapshot of what we felt was representative of the scenario that we were

interested in. For example, if we were looking for how the noise is distributed during the acceleration of a truck to maintain speed, we would record several pass-bys and replay the recording until we found a pass-by that had the sound characteristics of an accelerating truck. Figure 1 shows a snapshot of a coasting truck pass-by, and Figure 2 shows a moderately accelerating truck pas-by. The types of pass-bys that we were interested in were pass-bys that included trailer clanging (from uneven road surfaces), accelerating, and coasting. To quantify and locate non-dominant noise sources during vehicle pass-bys, we used the suppress feature shown in Figure 3. We then used the energy levels from each source to determine the respective proportion of the total received sound level

4 Results

4.1 Engine noise

We observed that the engine noise from an accelerating industrial truck pass-by came primarily from ground reflections as shown in Figure 2. The engine noise from the truck pass-by constituted approximately 66% of the total noise generated by a moderately accelerating truck pass-by.

4.2 Exhaust Noise

To quantify this noise source, we needed to use the suppress feature in the acoustic camera software to suppress the more significant noise sources so that an approximate sound pressure level could be assigned to the exhaust noise. We found that exhaust noise typically accounted for around 6% of the total noise during a coasting pass-by and 8% for an accelerating pass-by.

4.3 Tire Noise

Tire noise is a primary contributor to the immision of roadside pass-by noise. As shown in Figure 1, tire noise is easily identifiable and was a significant contributor to the overall noise from the truck pass-by. Tire noise contributed approximately 26% if all truck noise during an acceleration, and 81% if truck noise during coasting.

4.4 Trailer Clanging

Trailer clanging is a highly impulsive sound that primarily comes from the moving components of the unloaded trailer suspension and the worn landing gear. The landing gear was easy to isolate because as the truck passed over the uneven section of roadway, the landing gear clanging would become the most dominant noise source as shown in Figure 5. The rear suspension, located at the back of the trailer, would also become the most dominant noise source as the rear set of wheels would pass over the uneven road section. The rear suspension clanging was only identifiable as trailer body noise because the source was not easily identifiable as it was hidden behind the rear trailer tires. Figure 5 shows what the rear suspension clanging noise looked like when identified by the acoustic camera.

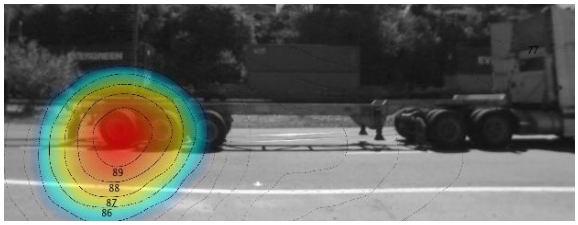


Figure 4: Rear suspension clanging over uneven road

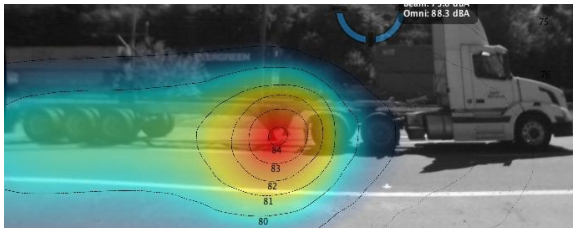


Figure 5: Trailer landing gear clanging over uneven roadway

4.5 Noise Distribution

We determined that noise from accelerating truck pass-bys is 66% engine noise, 26% tire noise, and 8% exhaust noise. We also determined that coasting truck pass-by noise is comprised of 13% engine noise, 81% tire noise, and 6% exhaust noise.

5 Discussion

When investigating noise sources to determine what mitigation strategies would be most effective, understanding how the noise from industrial vehicles is emitted can greatly aid the design and effectiveness of the noise mitigation.

As an example, if noise mitigation is considered for an uphill road section, one would expect the vehicles on one side of the road to be coasting, and vehicles on the other side to be accelerating. Knowing that the majority of noise from accelerating and coasting transport vehicles is generated at ground level, one could deduce that the majority of roadside pass-by noise could be mitigated by the installation of a low roadside barrier. It would also be important to know a barrier of this type would not be able to mitigate higher elevation noise sources like exhaust noise. A barrier intended to mitigate exhaust noise would likely have to be quite tall and may not be feasible or cost effective. Especially since exhaust noise only makes up 6-8% of the roadside pass-by noise.

6 Conclusion

Engine, tire, and exhaust noise are the primary contributors to industrial vehicle pass-by noise. Knowing that engine and tire noise both emanate from ground level, and are the two largest contributors to roadside pass-by noise, designers of roadside noise mitigation will be able to effectively design barriers to mitigate the majority of roadside truck pass-by noise.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Acoustical Effects Of Wind Screens On Infrasound Measurements

Todd Busch, Pearlie Yung, Jeremy Thorbahn, John Swallow

Swallow Acoustic Consultants Ltd (SACL) conducted a two-week long measurement program to determine audible sound and infrasound levels at five homes in the Municipality of Kincardine, Ontario. The results of these acoustical measurements, conducted both outside and inside of each home, are discussed; along with comparisons to meteorological data obtained at one of the residences. In the absence of technically substantive information to the contrary, a standard 60 mm diameter open-cell foam wind screen was utilized for both outdoor and indoor measurements. Wind speeds ranged from 0 to 17 m/s with 24-hr daily averages as high as 12 m/s. During the course of the data analysis, of unweighted, A-weighted, and G-weighted results, SACL noted that high wind speeds appeared to have compromised some of the data collection. A methodology for filtering good/bad data is discussed. Furthermore, SACL undertook a wide ranging literature review of the technical and regulatory documentation pertaining to the use of wind screens. After this review, the lack of technical justification for the use of a wind screen has become apparent, despite their conventional utilization.

ACOUSTICAL STANDARDS - NORMES ACOUSTIQUES

Standards For Modelling Road Traffic Noise In Western Canada
Forest Borch, Mark Bliss 70

Csa Z107.6 Audiometric Testing For Use In Hearing Loss Prevention Programs: A New Title For A New Edition
Sasha Brown, Christian Giguère, Michael Sharpe 72

Csa Z1007: A New Management Standard From The Canadian Standards Association For The Management Of Hearing Loss Prevention Programs
Jeffrey Michael Goldberg, T Kelsall, A Behar, P Malek 74

Csa Z107.56 And Safe Use Of Music Players
Tim Kelsall 76

Changing Standards For Building Acoustics Regulation
J. David Quirt 78

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 80

STANDARDS FOR MODELLING ROAD TRAFFIC NOISE IN WESTERN CANADA

F. Borch[*]**, EIT and M. Bliss, P.Eng.**[†]
BKL Consultants Ltd., #308 – 1200 Lynn Valley Road, North Vancouver, BC V7J 2A2

1 Introduction

A comparison was made between three road noise prediction standards: U.S. Department of Transportation Traffic Noise Model 3.0 (TNM), [1] and French standards NMPB 1996 [2] and NMPB 2008, [3][4] based on their ability to accurately predict the acoustical environment near Greater Vancouver highways. The comparison was made using post-construction noise monitoring performed by BKL staff and noise models that were developed by BKL staff for two large highway infrastructure projects near Vancouver, BC: the Port Mann Highway 1 Improvement (PMH1) Project and the Golden Ears Bridge (GEB) Project.

Both projects involved the construction of a multi lane bridge crossing the Fraser River, and the upgrading of highways and roadways on either side. BKL's scope on each project included extensive traffic noise modelling to predict post-construction conditions, and post-construction noise monitoring to confirm the accuracy of the predictions.

2 Method

2.1 Model Development

For the GEB project, a traffic noise model was developed using CadnaA® which included ground contours at 0.5 metre intervals, buildings, barriers, and highway traffic with 2-9% heavy trucks and posted highway speeds ranging from 60-100 km/h.

In a similar manner, a CadnaA® model was developed for the PMH1 project which included ground contours at varying intervals that were less than 0.5 metres, buildings, barriers, and highway traffic with 3-8.3% heavy trucks and highway speeds ranging from 70-100 km/h.

2.2 Measurement Procedure

Post-construction noise monitoring was performed at 15 and 80 locations along the extents of the GEB and PMH1 projects, respectively.

All monitoring was performed using sound level meters conforming to Type 1 specifications of American National Standard ANSI S1.4-1983 [5]. All measurements were taken in general accordance with the ISO 1996-2:2007 [6] and were clearly dominated by highway traffic noise.

2.3 Model Settings

Each CadnaA® model was developed using 3-D electronic files and drawings provided by the project highway engineers. Both models used baseline noise monitoring results for initial calibration. The post-project noise environment was predicted using highway traffic data (total volumes, vehicle classification and speeds) measured during the post-construction noise monitoring periods. The settings in Table 1 and Table 2 were applied or assumed for all predictions.

Table 1: General input parameters for CadnaA® road traffic noise modelling

Parameter	**Value**
General	
Max. Error (dB)	0.00
Max. Search Radius (m)	750-1000
Min. Dist Scr to Rcvr (m)	0.00
Digital Terrain Model	
Model of Terrain	Triangulation
Reflections	
Max. Order of Reflections	1-2
Projected Line Sources	On
Global Ground Absorption	1.0
Road Surface Ground Absorption	0.0

Table 2: Prediction standard specific input parameters for CadnaA® road traffic noise modelling

NMPB 1996 Specific Parameters	**Value**
Spectral Shape	2003/613/EC
Meteorology	Day: 50, Night: 100
Calculate Outer Lanes Separately	Enabled
NMPB 2008 Specific Parameters	**Value**
Calculate Outer Lanes Separately	Enabled
Meteorology	Day: 50, Night: 100
Calc Atalus	Enabled
Use Ch in Calculation of Abar	Enabled
TNM 3.0 Specific Parameters	**Value**
Truck Type	Heavy
Correct Bug in Fresnel-zone-calculation	Enabled

[*] borch@bkl.ca
[†] bliss@bkl.ca

2.4 Site Selection

Seven sites were selected to compare against the model predictions. Sites were selected where: the dominant noise sources were clearly apparent, the barrier attenuation was easy to define (e.g., no property line fences or other difficult-to-model objects), the measurement duration was at least 24 hours in duration, and where road traffic data specific to the site location was available. In addition, sites were excluded where there was the potential for significant attenuation from foliage due to the difficulty in accurately modelling excess attenuation due to heavy foliage.

All seven receivers had a 180 degree exposure to the highway, with setback distances ranging from 20-180 metres. Receivers were positioned at heights ranging from 1.5-2.8 metres above the ground, and the terrain between the highway and receiver was soft ground in all cases. One of the sites was on structure and four others had noise walls. Sites with modelled noise walls are shown with an asterisk in Table 3.

3 Results

Table 3 shows the average difference and the standard deviation of the differences for each prediction standard, along with the individual differences between predicted and monitored sound levels at each site.

Table 3: Calculated difference between predicted and measured sound levels (dBA) for three road traffic noise modelling standards

Site No. and Project	NMPB 1996	NMPB 2008	TNM 3.0
1 – GEB	-1.7	-3.9	-5.8
2 - GEB[1]	-1.9	-0.5	-2.9
3 - GEB[2]	-1.7	-0.6	0.6
4 - GEB	-1.0	-1.2	-4.5
1 – PMH1[1]	-1.6	-2.3	-6.6
2 – PMH1[1]	-0.8	-2.3	-3.6
3 – PMH1[1]	-0.8	-2.3	-3.8
Average Difference	-1.4	-1.9	-3.8
Standard Deviation of Difference	0.4	1.3	2.6

1– With 3 metre height noise wall, 2 – On structure with parapet

4 Discussion

All noise prediction standards under-predicted when compared with the sound levels that were measured. The road traffic noise model NMPB 1996 gave the most accurate results, with a maximum difference from the measured results of -1.9 dBA. Both NMPB models consistently under predicted for all sites. The TNM model was not consistent in its predictions, ranging from over predicting by 0.6 dBA to under predicting by -6.6 dBA. This is reflected in the large standard deviation of difference of 2.6 dBA.

The same set of calculations were repeated using the two NMPB standards, but changing the ground absorption setting for the road surfaces to non-reflective. The revised results were an average difference of -2.0 dBA and a standard deviation of differences of 0.5 dBA for NMPB 1996, and an average difference of -3.0 dBA and a standard deviation of differences of 1.3 dBA for NMPB 2008.

5 Conclusions

The accuracy of traffic noise modelling standards NMPB 1996, NMPB 2008 and TNM 3.0 were compared using post-construction measurements for two large highway infrastructure projects near Vancouver, BC, and the noise models developed for each. While all standards under-predicted traffic noise levels on average, the French standard NMPB 1996 yielded the most accurate results, with an average deviation of -1.4 dBA and a standard deviation of the differences of 0.4 dBA. The accuracy of the results was reduced if the road surfaces were not modelled as sound-reflective.

Although the TNM model was developed for North America, where traffic is more typical of that in BC as compared to traffic in France, the TNM model had poor results in comparison to the French road standards, for sites with and without noise walls.

References

[1] TNM-FHWA Federal Highway Administration Model, (http://www.trafficnoisemodel.org) TNM Version 2.5, McTransCenter University of Florida, 2088 Northeast Waldo Road, Gainesville, Fl 32609, http://mctrans.ce.ufl.edu

[2] NMPB-Routes-96. 1997. “Methode de calcul incluant les effets meteorologiques, version experimentale, Bruit des infrastructures routieres.” Lyon, Centre d’etudes sur les reseaux, les transports, l’urbanisme et les constructions publiques. Service d’etudes techniques des routes et autoroutes - Laboratoire central des ponts et chaussees - Centre scientifique et technique de batiment.

[3] NMPB-Routes-2008. 2009a. “Guide méthodologique, Prévision du bruit routier, Volume 1: Calcul des émissions sonores dues au trafic routier.” Référence Sétra: 0924-1. SETRA (Service d'études sur les transports, les routes et leurs aménagements).

[4] NMPB-Routes-2008. 2009b. “Methodological guide, Road noise prediction, volume 2: NMPB 2008 - Noise propagation computation including meteorological effects.” Référence: LRS 2008-76-069. SETRA (Service d'études sur les transports, les routes et leurs aménagements).

[5] American National Standards Institute (ANSI). 1983. “Specification for Sound Level Meters”. Reference No. ANSI / ASA S1.4-1983 (R2001). New York, Acoustical Society of America (ASA).

[6] International Organisation for Standardization (ISO). 1996. “Acoustics - Attenuation of Sound During Propagation Outdoors - Part 2: General Method of Calculation”. Reference No. ISO 9613-2:1996. Geneva, International Organisation for Standardization.

CSA Z107.6 AUDIOMETRIC TESTING FOR USE IN HEARING LOSS PREVENTION PROGRAMS: A NEW TITLE FOR A NEW EDITION

Sasha Brown *[1], Christian Giguère †[2] and Michael Sharpe ‡[3]
[1]WorkSafeBC, Vancouver, British Columbia, Canada
[2]Audiology & SLP Program, University of Ottawa, Ottawa, Ontario, Canada
[3]Workplace Hearing Assessments & Training, Dartmouth, Nova Scotia, Canada

1 Introduction

The Canadian Standards Association's (CSA) Occupational Hearing Conservation Technical Committee was formed in 2010, and created several subcommittees with different focuses. Among the subcommittees, one was tasked with reviewing and revising CSA Z107.4 *Pure Tone Air Conduction Audiometers for Hearing Conservation and for Screening*, and CSA Z107.6 *Pure Tone Air Conduction Audiometry for Hearing Conservation*, which were written in 1986 and 1990 respectively. It was quickly recognized that the two standards contained both redundant and complementary information. In 2012, CSA Z107.4 was withdrawn with the understanding that the new edition of CSA Z107.6 would contain the relevant information. Concurrent to this, a new standard, CSA Z1007 *Hearing Loss Prevention Program Management* was being drafted by a different subcommittee, and efforts were made to create complementary documents intended for different audiences. While Z1007 would be aimed at employers and their Hearing Loss Prevention Program Administrators (HLPPAs), and covers all aspects of creating and managing an organization's HLPP, CSA Z107.6 would be directed to individuals, organizations and service providers responsible for conducting audiometric tests as part of a Hearing Loss Prevention Program (HLPP).

This new edition of the CSA Z107.6 Standard will be completed in 2016, and like its predecessor, focuses on quantifying hearing loss as a means of early detection of possible damage from hazardous noise exposure. The 2016 edition most notably includes guidance to the technician on classifying test results, advising the tested individual about their test results and providing reports to the Hearing Loss Prevention Program Administrator for use within an HLPP. Also added were mobile test facility requirements, and expanded sections on audiometric test equipment and audiometric technician training requirements.

*sasha.brown@worksafebc.com
†cgiguere@uottawa.ca
‡info@hearingconservation.ca

2 New Sections

2.1 Explaining test results

Significant threshold shift

Canada has never had a single method of determining whether a change in hearing is indicative of early signs of noise induced hearing loss. CSA Z107.6 tackled this issue and recognized that there are many different strategies to determine if hearing has possibly deteriorated due to exposure to noise, both nationally and internationally. Many jurisdictions within Canada have developed their own unique methods and definitions of hearing loss and change in hearing. In British Columbia for example, "Early Warning Change" is defined as $\geq$ 15dB deterioration at either 3 or 4kHz in either ear. In jurisdictions without defined methods of interpretation, companies or service providers tend to develop their own methods or follow OSHA's recommendations.

The Z107.6 Standard now provides a recommendation with two methods of determining whether a change in hearing consistent with hazardous noise exposure has occurred. Specifically, it recommends the following for the calculation of Significant Threshold Shift (STS):

- an average shift of the thresholds at 2000, 3000 and 4000 Hz of $\geq$ 10dBHL (30dBHL or more combined) in either ear, or
- a shift of $\geq$ 15dBHL at 3000 or 4000 Hz in either ear.

The new edition of the Standard also provides a recommendation for determining whether an individual's hearing has possibly been affected by noise, even when there is no previous test with which to compare it, either because it is a 'base-line' test or because previous records are unavailable. In such cases the recommendation is that, if the difference between a threshold at 3000 Hz, 4000 Hz, or 6000 Hz and the adjacent lower frequency is $\geq$ 15dBHL, a note should be made indicating possible contribution of noise.

Explaining results to the tested individual

While, previous editions of CSA Z107.6 did not require that the technician counsel the tested individual, the 2016 edition requires that technicians explain the implications of hearing loss or shifts in hearing, and ensure that each individual, even those without a hearing loss or shift in hearing, is advised on the importance of protecting their hearing and appropriate use of Hearing Protection Devices.

Reporting Results

CSA Z1007, the Management standard, provides guidance to the HLPPA on what actions to take when hearing loss consistent with noise exposure has occurred. CSA Z107.6 in turn provides guidance to the audiometric technician on what to report to the HLPPA.

2.2 Mobile facility requirements

Many employers or organizations where hazardous noise might be present prefer to have an audiometric testing facility visit their site, rather than either purchasing the equipment to have "in-house" testing, or spend time commuting to and from an audiometric test site. Recognizing that mobile testing facilities are common, but also recognizing that there are unique issues posed by their transient nature, the new Z107.6 includes requirements specific to mobile testing facilities. Specifically, to meet the requirements set out in the standard, ambient noise levels are to be verified daily, and whenever the facility is relocated. It is also recommended that noise levels be continuously monitored, and that when ambient noise levels are exceeded testing is paused or stopped until allowable levels are achieved again. Relocation of the mobile facility might be necessary in order to obtain the required ambient noise levels.

3 Expanded sections

3.1 Audiometric Testing Equipment

The technical characteristics of audiometers for use in hearing loss prevention programs, which were previously specified in a separate document Z107.4, have now been integrated into the revised Z107.6 Standard. Given that audiometers are designed and fabricated for the global market, it was also felt that specifying detailed characteristics and requirements for the Canadian market was not warranted and harmonization with the international standards was a better approach.

The minimum requirement is a Type 4 audiometer as per ANSI/ASA S3.6 with recommended additional provisions for an expanded frequency range (500-8000 Hz) and hearing level range (-10 to 100 dB HL). Furthermore, CSA Z107.6 now specifies a wider range of earphones, either supra-aural or insert types, that can be used for audiometric testing. Manual or automatic threshold search methods may still be used as they were in the previous edition.

The new edition also provides an expanded section on calibration requirements. These are comprised of periodic annual electroacoustic calibration to be carried out by a trained service technician as well as routine equipment checks to be carried out by the audiometric technician by means of a biologic or bio-acoustic simulator approach. Reporting and recording requirements are now specified for the issuing of the certificate of calibration and routine equipment checks.

3.2 Training Requirements for the Technician

The revised edition of CSA Z107.6 has additional training components due to the expanded role of the audiometric technician. The Standard now requires training on how to classify test results, provide appropriate feedback to individuals and HLPPAs, as well as the criteria for referral to a health care professional. In addition, technicians are to be trained in the use of hearing protection devices and their selection and rating methods (as outlined in CSA Z94.2). Like the original edition, it is suggested that this training requires a minimum of 20 hours. Furthermore, a "refresher" course must also be completed by technicians at least every 5 years.

Only someone who has taken all components of the training course meets the requirements to be labeled an "audiometric technician" within the Standard, but the new edition recognizes that for some organizations, roles can be divided between people. One person will conduct the hearing tests, for example, and another will provide the advice and recommendations. In these cases, the Standard requires that all elements of the training must still be covered by the different people.

4 Conclusion

The new edition of Z107.6 is a comprehensive revision of the first edition of the standard and is aimed at covering all aspects of audiometric testing for use in Hearing Loss Prevention Programs, and is directed to organizations and service providers responsible for conducting or administering audiometric testing for HLPPs. It now includes the characteristics of the test environment and equipment, training requirements for technicians, detailed pre-testing and testing procedures, interpretation of test results and advising, and record keeping and reporting requirements.

It is hoped that the second edition will foster a more structured and uniform approach to audiometric testing and contribute to the success of HLPPs across the country.

Acknowledgements

The members of the subcommittee (SC3) who revised Z107.6, were selected from a broad range of practitioners, government and academia and were as follows: Alberto Behar (Ryerson University), Sasha Brown (WorkSafeBC), John Franks (LytleSound), Christian Giguère (University of Ottawa), Tim Kelsall (HATCH), Pierre Lamontagne (Department of National Defence), Jim Leblanc (NS Dept of Labour and Advanced Education), and Michael Sharpe (WHAT Ltd.). The authors of the present paper gratefully acknowledge the contributions made by the members of SC3.

References

ANSI S3.6-2010 American National Standard- Specification for Audiometers

CSA Z94.2-14 Hearing protection devices — Performance, selection, care, and use

CSA Z1007-16 Hearing loss prevention program management

CSA Z1007:16 A NEW MANAGEMENT STANDARD FROM THE CANADIAN STANDARDS ASSOCIATION FOR THE MANAGEMENT OF HEARING LOSS PREVENTION PROGRAMS

Jeffrey M. Goldberg[*], Chair of Subcommittee 5, S304 Noise and Vibration Technical Committee
T. Kelsall[†] and A. Behar[‡], Co-Chairs of S304 Noise and Vibration Technical Committee
P. Malek[^], Project Manager Canadian Standards Association -

A new 1000 series standard developed for the Canadian Standards Association's series of Noise Management and Hearing Loss Prevention group of standards. Produced under the auspices of CSA's S304 Noise and Vibration Technical Committee

1 Introduction

12 years ago (2004), the Canadian Standard Association's Acoustics and Noise Control Committee met at an annual conference of the Canadian Acoustics Association. A paper was presented, at that meeting, on the value of companies implementing a "hearing conservation program". Several of the Committee members were inspired by this presentation and asked to discuss this subject at the Technical Committee (TC) meeting. From that meeting, the Committee suggested that there ought to be a National Standard on occupational hearing conservation. This idea was subsequently put on the back burner, as this subject was not within the scope of the Acoustics and Noise Control Committee.

A couple of years later (2006), the Ontario Ministry of Labour, as part of a review of their noise control and hearing protection regulations, came forward with a recommendation that "hearing conservation" be recognized as an essential part of OHS programs. However there was insufficient support from the other Provinces and the Federal Government (as well as other potential advocates). As a result, CSA could not get this project past the proposal stage.

Moving forward another three years (2009), as a result of a reorganization of a couple of the CSA Standards Program areas it was that we decided to combine the remainder of our Acoustics and Noise Control TC with our Hearing Protection TC – forming the new TC on Occupational Hearing Conservation. This new TC was given a broader mandate, which included occupational hearing loss prevention**i**.

There was still need for National support for such a standard (having failed to get this essential support in 2006). With an increased value proposition, we presented the proposal for development of this management system standard to our Advisory Council (comprised of OHS regulatory authority representatives from all 14 jurisdictions across Canada). Through this Advisory Council, funding was sought from CAALL-OHS (the OHS Committee of the Canadian Association of Administrators of Labour Legislation). A year later (2010), funding was offered for the development of this new standard. CSA decided to designate this new standard "CSA Z1007" as it was to be part of the Z1000 series and it was related to the Z107 noise control and audiology standards.

With this support in place, CSA took the proposed new project back to the Technical Committee. They put together a development plan and struck a new sub-committee (SC5) to draft the standard. I was named as the sub-committee's first Chairperson. Work began on Z1007 in May of 2011. Today I'll present the results!

2 Method

2.1 The Canadian Regulatory Framework

Its present condition

Canada has a fragmented set of norms for Hearing Conservation owing the 14 jurisdictions that establish standards for these programs. CSA Z1007, written with the support of regulators in various parts of Canada, seeks to establish norms, which can be used to manage these programs as hearing loss prevention programs. The difference between loss prevention and conservation is a proactive stance seeking to remove the causes of noise induced hearing loss before permanent hearing damage takes place.

2.2 Sources of Information

The committee reviewed standards from New Zealand and Australia1, the National Institute of Occupational Safety and Health (NIOSH) in the United States2 and the European Union. As well, we reviewed the 14 different standards that are currently used in Canada.

3 Results

3.1 What does the standard include

The standard covers all aspects of hearing loss prevention from detection of potential toxic noise through to proactive measures to determine if the program is being effective. The standard is compatible with and reflective of CSA's other standards in this area

[*] Chairman, S304/SC5 subcommittee, Jeffrey@protectear.com
[†] Hatch, tkelsall@hatch.com
[‡] Ryerson University, albehar31@gmail.com
[^] CSA Group Rexdale, ON, Priya.malik@csagroup.org

1 1269.1-2005_R2016 Occupational Noise Management

2 Preventing Occupational Hearing Loss – A Practical Guide, DHHS(NIOSH) Pub. No. 96-110; A Practical Guide to Effective Hearing Conservation: Programs in the Workplace, DHHS(NIOSH) Pub. No. 90-120; Criteria for a Recommended Standard – Occupational Noise Exposure, DHHS(NIOSH) Pub. No. 98-126.

3.2 How Does the Standard Enable Hearing Loss Prevention

The Standard addresses

- The fragmented nature hearing loss prevention in Canada, how the standard seeks to address that,
- What proactive measures can and should be taken, and how to determine if the standard is being applied effectively.
- There are several informative annexes addressing parts of the program in greater detail and Best Practices for those who wish to excel in this area.

4 Conclusion

The standard was published in March of 2016, and is seeking adoption from the regulators that enabled its creation. It is the first such Standard in North America. "CSA Z1007" is one of the leading standards in our Z1000 series of OHS management standards. In the future, we hope to improve Z1007 further through the development of a 2nd edition.

Remerciements/Acknowledgments

The authors of the present paper gratefully acknowledge the contributions made by the members of SC5. We completed this new standard in three and one half years. That is a significantly small amount of time for a new document. That couldn't have been achieved without members of the subcommittee consenting to many conference calls between our semi-annual face-to-face meetings. The members of the subcommittee (SC5) who created Z1007, were selected from a broad range of practitioners, government, corporate, and academia and were as follows: A. Behar (Ryerson University), E. Berger (3M Corporation, Indianapolis IN), S. Bly (Health Canada), B. Borst (3M Canada, London ON), J.M. Cousineau (Ontario Ministry of Labour, Toronto ON) S. Brown (WorkSafeBC, Richmond BC) I. Bhunnoo (Workplace Safety and Prevention Services ON) J. Franks (LytleSound), A. Fedee (Via Rail, Toronto, ON), Tim Kelsall (HATCH), S. Lake (Consulting Acoustical Engineer, Colorado Springs, CO), P.Malek. (CSA Group, Rexdale, ON.), M. Russo (Infrastructure Safety Assoc. of Ontario, Toronto ON), T. Schulz (Honeywell Safety Products, College Station TX.), D.Shanahan (CSA Group Mississauga, ON), K. Turner (Protec Hearing Inc., Winnipeg MB), L. Wells (3M Corporation, St. Paul MN) The members of S304/Subcommittee 5 wish to acknowledge the participation of those members of the subcommittee from the United States that worked on this project.

Just prior to that time, CSA established a new series of standards on OHS management systems (the CSA Z1000 series). CSA's Strategic Steering Committee was asked to review the new Z1000 standards and suggest other subjects that should be given special focus in under the broad subject of OHS Management. From that review, a couple of our Strategic Steering Committee members suggested that a standard on "Management of Hearing Conservation Programs" be added to this series. CSA staff considered this suggestion and came to the conclusion that the creation of such a management system standard could serve to tie together the diverse standards assigned to the Hearing Conservation TC. We presented this idea to the TC and they accepted it (in principle). The final title of the standard changed to Hearing Loss Prevention Program Management was chosen during the standard's development.

CSA Z107.56 AND SAFE USE OF MUSIC PLAYERS

Tim Kelsall CIH, ROH, INCE Bd.Cert.
Hatch Noise and Vibration Group, Mississauga, Ontario
tkelsall@hatch.ca

Résumé

Il y a eu beaucoup d'inquiétude au sujet des jeunes (et plus) personnes écoutant les lecteurs de musique personnels dans le cadre de leur vie quotidienne et la façon de les protéger contre la perte d'audition. CSA Z107.56 comprend une section sur l'estimation de l'exposition au bruit dans les casques qui met cette question en perspective. Basé sur des recherches indiquant que la plupart des gens régler le volume de la musique et de la parole à environ 15 dB au-dessus de la température ambiante actuelle la norme fournit une estimation de leur exposition au bruit.
En supposant, comme la norme fait, que les écouteurs typiques offrent peu de réduction du bruit ambiant, la plupart des gens ne seront pas exposées au-dessus de 85 dBA à condition que le bruit de fond est inférieur à 70 dBA. Dans la pratique, cela signifie que la surexposition devrait être rare sauf dans les rues très fréquentées ou dans des véhicules plus forts, tels que les voitures de sport, le transport à grande vitesse et des avions.
Cela limite les domaines de préoccupation considérablement et devrait aider à cibler les moyens utiles pour contrôler la surexposition.
Mots-clés: Musique, exposition au bruit, les normes de bruit des casques de CSA, le bruit des transports, le bruit de la circulation

Abstract

There has been much concern about young (and older) people listening to personal music players as part of their daily life and how to protect them from hearing loss. CSA Z107.56 includes a section on estimating noise exposure under headsets which puts this issue in perspective. Based on research indicating that most people set the volume of music and speech at about 15 dB above the existing ambient the standard provides an estimate of their noise exposure.
Asssuming, as the standard does, that typical earbuds provide little reduction to the ambient noise, most people will not be exposed above 85 dBA provided the noise background is below 70 dBA. In practice, this means that overexposure should be rare except on very busy streets or in louder vehicles, such as sports cars, high speed transit and aircraft.
This narrows the areas of concern considerably and should help in targeting useful ways to control overexposure.
Keywords: Music, noise exposure, headsets CSA noise standards, transportation noise, traffic noise,

The 2013 update of CSA Standard Z107.56 {1} contained a new section entitled "Workers exposed to music players, radios, or other sound reproduction devices". It takes advantage of a new Estimation Method proposed by David Quirt and based on research by Christian Giguere et al summarised in Reference [2].

The standard takes advantage of the behaviour of most listeners who adjust the sound level of music to be about 15 dB above the background noise they are exposed to. Such behaviour is unlikely to be seen only among workers and provides a method for identifying when exposure to music, reproduced speech, etc. is likely to be hazardous outside work without the need for any instrumentation. Given the widespread use of personal music players this has been a public concern for some time.

In practice, assuming the music is from a speaker or from a headset, earbud, etc. with minimal inherent noise reduction, which is a conservative assumption, it can be expected that the user will adjust the sound level about 15 dBA above the ambient noise they are exposed to and provided the music plays more than say 80% of the time their noise exposure will be 15 dB above the ambient noise in their location. Put another way, if we take the 85 dBA limit used in most countries as the industrial noise limit as indicative of the point at which there is a concern, then provided the ambient sound level was below 70 dBA there should not be a concern unless the person were exposed more than 40h a week. There is certainly evidence that some people can suffer hearing loss from long term exposure below 85 dBA in the workplace. Listening to music more than 40h a week does occur but is not that widespread, certainly not while being exposed above 70 dBA. In addition, even poorly designed headsets will provide some attenuation, so 70 dBA appears to be a reasonable cutoff.

Most residences would be well below 65 dBA except when music or TV was playing or a householder was vacuuming. Unless a person was listening to earbuds while watching TV or vacuuming for a very long time, there should be little concern.

Outdoors, most streets except major arterials would be below 70 dBA Leq on sidewalks, so the main concern would be people listening to a music player while walking close to a major arterial. This is already considered a hazard

from a safety standpoint and the extra concern should reinforce the need to avoid such behaviour.

Using gas powered tools such as lawnmowers would also clearly be a noise concern, with or without music players, although the players would exacerbate the problem considerably. Again it would be unlikely that many would be exposed to gas powered tools for many hours a week outside the workplace. Encouraging the use of hearing protection and discouraging music players while operating such equipment would be worthwhile.

Recreational activities can also have high sound levels and headset use should be discouraged with those activities.

Offices would generally be fine for using earbuds, but industrial areas and loud stores with music could be a concern. In any case, workplace noise exposure is already regulated and controlled, although work is needed to spread the information needed to do this properly.

The other major venue for public noise exposure would be transportation.

Neitzel et al measured sound levels on public transit in New York City [3], getting results from 71 to 80 dBA, the latter in the subway. Exposure time would generally be limited but headset use in transit vehicles should be discouraged.

The interior noise in cars has been tabulated in Reference [4]. Up to 100 km/h only 7 of the 130 cars tested exceed 70 dBA. At 120 km/h 22 of the 130 cars exceeded 70 dBA, mostly high powered sports cars. With windows open 79 dBA can be reached as low as 50 km/h [5]. It can be concluded that there is little hazard to hearing in playing music in cars except for high powered cars at very high speed or in cars with open windows.

Reference [4] (which admittedly is old) also indicates that most seats on passenger jet aircraft are above 80 dBA. Using a headset to listen to music or a movie would exceed the daily allowed noise dose of 85 dBA in less than 48 minutes in an ambient of 80 dBA. This is clearly one reason why noise cancelling headsets are so popular on aircraft and provided they drop the sound level by 10 – 15 dBA they should be protective, while also improving the sound quality.

This leads to a discussion of what can be done.

- Do not listen to a music player in noisy environments such as:
    1. When other music or shows are being played.
    2. While walking near busy major arterial roads. It is both unsafe and can cause hearing loss.
    3. During noisy recreational activities, like clubbing and power boating or in venues already playing music or TV,
- Use noise cancelling headphones or headsets while listening to music or recordings in high speed powerful cars or in any car with the windows open.
- Use noise cancelling headphones or headsets with passive hearing protection providing at least 10-15 dB noise reduction on passenger jet aircraft.

This list is probably not complete but is sufficient to give most lay people sufficient guidance to avoid situations where listening to music, radio, TV, etc. would be likely to be hazardous to their hearing over the long term. It also indicates that most uses of personal music players should not adversely affect hearing unless unusual volume levels are selected.

---

1. CSA Standard Z107.56-13 Measurement of "Noise Exposure
2. Christian Giguere et *al Direct and indirect methods for the measurement of occupational sound exposure from communication headsets NCEJ Sept 2012*
3. Richard Neitzel, MS, Robyn R. M. Gershon, DrPH, MHS, Marina Zeltser, BS, Allison Canton, BA, and Muhammad Akram, PhD Noise Levels Associated With New York City's Mass Transit Systems Am J Public Healthv.99(8); Aug 2009
4. http://www.auto-decibel-db.com/index_kmh.html
5. Passenger noise environments of enclosed transportation systems By United States. Office of Noise Abatement and Control

CHANGING STANDARDS FOR BUILDING ACOUSTICS REGULATION

J. David Quirt [1]
1949 Mulberry Crescent, Ottawa K1J8J8, Canada

1 Introduction

A new approach to the control of sound transmission between adjacent units in multi-family residential buildings is among the changes in the 2015 edition of the National Building Code of Canada (NBCC). Provincial codes are expected to follow essentially the same approach. The design objective has changed from a minimum direct STC for the wall or floor/ceiling assembly separating adjacent units to a requirement for the Apparent Sound Transmission Class (ASTC) that includes both direct and flanking sound transmission. The required design procedure uses data from conventional laboratory measurements (ASTM E90) of direct transmission through wall or floor/ceiling assemblies combined with measurements of flanking transmission (conforming to ISO 10848) as inputs for a calculation procedure based on ISO Standard 15712 Part 1 (1995).

Meanwhile, ISO committee TC43/SC2 is revising the ISO calculation standard. The new ISO 12354 series of standards will use the latest version of the European EN 12354 standards thus aligning with European practice, and the ISO 15712 series will be withdrawn. This change was balloted in early 2016, and should become final by the end of this year. In the longer term this will provide better support for Canadian regulations, but in the short term this leaves the new provisions in the Canadian building code pointing to an old standard that will no longer be available to practitioners.

This paper focuses on explaining the technical intent and form of the new Building Code requirements, and how these will be supported better by the ISO 12354-1 (2016) version of the ISO calculation standard.

2 The Calculation Method

ISO 15712-1 and the new Code requirements approach predicting the sound transmission from the same basic concept – combining the sound power transmitted directly through the separating assembly with the flanking sound power transmitted via first-order flanking paths at the four edges of the separating assembly.

To discuss this, it is useful to introduce the convention used in ISO 15712-1 and the new ISO 12354-1 for labelling the transmission paths, as illustrated in Figure 1. Consider transmission from a source room at the left to the receiving room beside it. Each transmission path involves one surface in the source room (denoted by a capital letter) and one in the receive room (lower case). **D**irect transmission through the separating wall is path **Dd**. For each edge of the separating assembly there are three 1st-order flanking paths, each involving a surface in the source room and one in the receiving room, that connect at this edge: **Ff** from **f**lanking surface **F** to **f**lanking surface **f**, **Df** from **d**irect surface **D** to **f**lanking surface **f,** and **Fd** from **f**lanking surface **F** to **d**irect surface **d** in the receiving room.

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

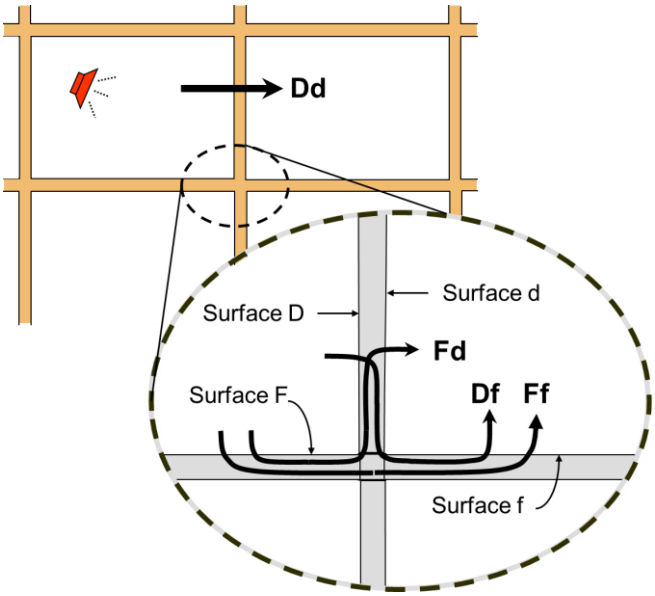


Figure 1: Labelling convention used in ISO 15712-1 (and its successor ISO 12354-1) for direct and flanking transmission paths

In Canada, the acoustic testing of building elements (walls, floors, etc.) is normally done according to ASTM E90 and the Code requirements are given in terms of STC or ASTC ratings determined from the 1/3-octave test data, following the procedure in ASTM E413. Merging this ASTM context familiar to Canada's building industry and regulators with the ISO procedures now being added to Canadian building codes, requires new terminology, so "direct transmission loss" and "flanking transmission loss" have been introduced. These provide consistency with ASTM terminology, but match the function of the direct sound reduction index and flanking sound reduction index, as defined in ISO 15712-1 and the new ISO 12354-1.

Section 4 of ISO 15712-1 (and of its successor ISO 12354-1) specifies a process to calculate apparent sound transmission by combining the sound power transmitted via the direct path and the twelve first-order flanking paths (as illustrated in Figure 1).

1 jdq.acoustics@bell.net

Equation 14 of ISO 15712-1 is recast here in Eq. 1 with the paths labelled and grouped consistent with the naming in Fig.1, assuming rectangular room geometry and neglecting paths due to leaks, ducts, crawlspaces, etc., which should be controlled by normal good practice. The Apparent Sound Transmission Loss (ATL) between two rooms is the decibel expression of the sum of sound power due to Direct Sound Transmission Loss (TL_{Dd}) through the separating wall or floor element and the sound power due to Flanking Sound Transmission Loss contributions (TL_{Ff}, TL_{Fd}, and TL_{Df}) of the three flanking paths for every junction at the edges of the separating element:

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

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

- ***For heavy homogenous types of construction*** such as concrete floors or concrete block walls, the NBCC design procedure determines the flanking sound transmission loss following either the Detailed or Simplified calculation procedures of ISO 15712-1. For input data, these calculations use sound transmission loss data (for the base wall and floor assemblies and for linings) measured according to ASTM E90. Changing to the new ISO 12354-1 will not change this approach.
- ***For lightweight steel- or wood-framed assemblies***, the NBCC design procedure substitutes the use of experimental flanking data (treating flanking sound reduction index determined using ISO 10848 as Flanking Transmission Loss) for values calculated following the procedures of ISO 15712-1. Either a detailed calculation using 1/3-octave-band data in Eq. 1 or a simplified procedure using the corresponding single-number ratings is permitted.

In both cases, the calculation of system performance combines the sound power due to direct and flanking sound transmission in the same way, using Eq. 1.

3 Discussion

Because the ISO 15712-1 standard used in the noise control requirements in the 2015 NBCC was based on an old (circa 1990) version of EN12354-1, it made no reference to the newer measurement procedures of the ISO 10848 standards, which are essential for accurate characterization of the lightweight constructions used in many Canadian multifamily residential buildings. Specific provisions for calculating sound transmission in lightweight constructions were written into the NBCC requirements to get around this weakness of ISO 15712-1.

The new ISO 12354-1 standard includes references to ISO 10848 measurement procedures, clearly identifies situations where the standard procedures don't apply, and strongly recommends use of ISO 10848 data for lightweight constructions. Thus the new ISO 12354-1 standard is fully compatible with the intent of the NBCC provisions.

A second major advantage of the new ISO 12354 series is that Part 2 provides both detailed and simplified procedures for calculating transmission of impact noise in buildings of various types, including those with lightweight framed constructions. This provides a good foundation for adding impact noise control to the requirements in future editions of the NBCC.

4 Conclusion

The National Building Code and the standards referenced in the Code do not constitute a clear and readily-followed guide to the intended design approach for noise control in buildings. For additional advice, the NBCC refers users to the NRC publication RR-331, "Guide to Calculating Airborne Sound Transmission in Buildings". Edition 2 of this guide [1] was published in 2016, and further editions are expected as content is refined and approved by the industry steering committee (including extensive editorial changes to reference the new ISO 12354-1 after the change of standards is completed). Guide RR-331 presents extended descriptions of the calculation process (both simplified and detailed methods) for specific types of construction, and includes numerous benchmark examples of the calculations. It is supported by a set of other NRC reports, each for a specific type of constructions (such as concrete block walls, or wood-framed wall and floor assemblies), that include the test data needed as inputs for the calculations.

5 Remerciements/Acknowledgments

The author wishes to acknowledge continuing support from Standards Council Canada for participation in development of improved standards for building acoustics.

References

[1] B. Zeitler, D. Quirt, C. Hoeller, J. Mahn, S. Schoenwald, I. Sabourin, "RR-331: Guide to Calculating Airborne Sound Transmission in Buildings, 2nd Edition, NRC Canada, 2016.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Update On Communication Headset Noise Measurements In The Workplace

Flora G. Nassrallah, Nicolas N. Ellaham, Christian Giguere, Hilmi R. Dajani

Increased use of communication headsets found in various workplaces raises concerns regarding potentially hazardous sound exposure levels. Current national and international standards specify a wide range of simple and specialized methods for the measurement of sound exposure under communication headsets. The ISO 11904 describes two methods for the measurement of noise from sources close to the ears: the Microphone in a Real Ear and the acoustic manikin techniques. Some national standards also specify the use of general-purpose artificial ears. Finally, standard CSA Z107.56-13 describes an indirect calculation method as a simpler alternative requiring only basic noise measurement equipment and calculation steps. However, to date, quantitative data comparing the degree of agreement between the different measurement methods or their relative performance are lacking, and it is not known if occupational health and safety or hearing loss prevention stakeholders have the necessary training and equipment to integrate them in their daily practice. A three-step study including a survey questionnaire and a series of laboratory experiments was conducted to address the several knowledge gaps on the topic. This research provided new knowledge to guide selection of the most suitable methods for the assessment of communication headset exposure taking into account expertise, access to equipment, and field logistic constraints. The paper will summarize research findings and discuss implications for future revisions of standard CSA Z107.56. [This project was funded by a research grant provided by the Workplace Safety and Insurance Board of Ontario]

ARCHITECTURAL ACOUSTICS - ACOUSTIQUE ARCHITECTURALE

Acoustics In Alberta Schools - Overview And Challenges
Philippe Moquin, Ellen Buchan 82

Sound Field Diffuseness In Oblique-Shaped Reverberation Rooms With Different Test Configurations
Md Mehadi Hasan, Murray Hodgson 84

Scale-Model Investigation Of The Effect And Optimal Design Of Sound Absorbers In An Open-Plan Office
Md Amin Mahmud, Murray Hodgson 86

Objective Acoustical Quality In Healthcare Office Facilities
Murray Hodgson 88

Subjective Acoustical Quality In Healthcare Office Facilities
Murray Hodgson 90

Optimizing Ceiling Systems And Lightweight Plenum Barriers To Achieve Ceiling Attenuation Class (Cac) Ratings Of 40, 45 And 50
Gary S. Madaras, Andrew E. Heuer 92

The Dynamic Response Of Composite Floating Floors In Buildings
Max De Castro Magalhães 94

Exterior Noise Ingress In Greater Vancouver Residential Buildings: Prediction Versus Field Measurements
Gary Mak, Mark Bliss 96

Acoustical Design For 21st Century Learning
Paul Marks 98

Noise Control Issues In A Steel-Framed Institutional Building
Clair Wakefield 100

Acoustical Properties Of Homogeneous Planar Surfaces In Room Acoustics: A Theoretical Review
Behrooz Yousefzadeh, Murray Hodgson 102

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 104

ACOUSTICS IN ALBERTA SCHOOLS - OVERVIEW AND CHALLENGES

Philippe Moquin *[1], Ellen Buchan [†2]
[1]Alberta Infrastructure, Technical Services Branch, 3rd Flr. 6950 - 113street, Edmonton, Alberta, T5H 5V7

1 Introduction

The province of Alberta is presently embarked on a major modernization and new school building program of about 230 schools. Alberta Infrastructure has developed acoustical requirements and it is our mandate to implement these. The difficulty is that schools are the property of school boards and we cannot impose our requirements.
To achieve success, we need to ensure that our requirements are reasonable and also be able to answer questions on certain issues.

2 Alberta Infrastructure requirements

2.1 Comparison to ANSI/ASA S12.60-2010[1]

The "Standards and Guidelines for School Facilities"[2] was first published in 2000 as a result of years of involvement in various teaching facilities starting with a band room study in 1977. For the most part, the requirements are the same as S12.60. HVAC noise is specified as RC and not dBA as used in S12.60. This is typically the way a mechanical engineer designs for noise.
S12.60 specifies the STC for toilets and bathing rooms which we do not specify and also the coverage uniformity for Classroom audio distribution systems which is not included in our requirements.

2.2 Requirements not mentioned in S12.60

Gymnasia are required to have a reverberation time less than 2s and the HVAC noise RC 35.
Music rooms require a range of reverberation between 0.7s an 0.8s and well as door seals
Music practise room are required to have an NRC $\geq$0.8 on the walls and ceiling as well as door seals.
Corridors and lunchrooms require ceilings with NRC $\geq$0.55.
Student gathering areas and computer lab require ceilings with NRC $\geq$0.7
Drama theatres require advice from an acoustical consultant.

3 Challenges

3.1 Classroom Audio Distribution Systems

These were a very intensely studied area in early 2000. There were claims of significant academic improvement. However, in recent studies Dockrell [3] explains how difficult it is to obtain meaningful data and concludes "*academic attainments showed no benefits from the use of sound-field systems*". If these are to provide benefit they must be installed in classrooms that meet acoustical requirements. The question remains as to the benefit they provide in such a classroom.

3.2 Gymnasium reverberation

As we require the reverberation in the gymnasium to be less than 2 seconds, any difficulty in measuring this becomes a contractual issue. This is how the issue reported by Packer [4] came to Infrastructure's attention.
Since then, no further investigation has occurred but the use of plywood "diffusers" for testing is accepted by Alberta Infrastructure.
The challenge is still to explain the physics of the observed phenomenon and then, using this data, make the changes in the measurement standards bodies or redefine how to calculate reverberation time in such a space.

3.3 Movable partitions

The trend we are seeing is a rather indiscriminate use of movable partitions. The architects point to the STC values of the products (typically 54) and feel justified in using these. The more troubling trend is that of using mall style sliding glass partitions as these have a lab STC 42 rating or less.
There are unfortunately little published NIC values of installed movable partitions [5] but the manufacturers expect 8-12dB degradation from STC. This means that NIC of 46 is the best one can expect.
Few architects follow ASTM E557-12 which is very stringent. For example, the top track deflection must be less than l/1156 - much stiffer than the usual l/360.
A concern is also how these partitions perform over time. How does the NIC change over years of service?
The challenge is to educate the owners and architects as to how difficult it is to obtain acoustic separation with these and discourage their use.

3.4 Trade teaching areas

The existing don't address rooms where trades are taught in acoustically challenging environments. Ideally they should represent a typical working environment but this may not be acoustically reasonable teaching environment.

Cosmetology

The challenge of these spaces is that they typically emulate a high end salon. The floors are typically sheet goods and the walls must be easily washable and the ceiling is typically drywall. Because of the number of students in the space and the noise of the equipment it becomes difficult to communicate beyond a meter. Convincing the designer to use absorptive ceiling tiles or baffles has become easier with

* philippe.moquin@gov.ab.ca
† ellen.buchan@gov.ab.ca

some newer micro-perforated metal look products and felted polyester products.

Culinary Arts

These are effectively large commercial kitchens. The flooring is hard and the walls typically a highly washable plastic finish. Ceiling tiles are typically vinyl faced drywall. The ventilation system is typically loud.
The only area that can easily be modified is the ceiling tile. Previously the only alternative was mylar faced ceiling tiles. Recently, newer FDA approved products, with good acoustic absorption have been introduced.
Abating the exhaust fan noise remains a challenge.

Construction trades & Automotive Shops

These include a variety of workshops: woodworking, fabrication, auto body, and automotive shops. They can be treated together because they require hard flooring, generate significant dust and have high ceilings. The wall construction tends to be massive for fire control. The activities are also very noisy.
The primary surface that can receive acoustical treatment is the ceiling. Convincing the designers and owners of this expense is somewhat difficult as the argument is given that it is not typically provided in the workplace.
The noise of the exhaust air or dust collection systems is rarely addressed.

Modular classrooms

The intent of Alberta's modular classrooms is that each structure is capable of being moved to accommodate varying demands from shifting demographics in the community. These structures connect to other modular classrooms and/or the core school. They must be designed to be able to be moved over three times during a thirty-year period and meet all the "standard" acoustical requirements. Over the last decade the province has introduced teachers and students to the new High Performance Modular Classroom, with approximately 80 m2 floor space, each with a dedicated HVAC system.
The most significant acoustical challenge for the builders is achieving RC 30 (N) background noise criterion. .
The acoustical environments in a particular builder's modular classroom prototypes has been accessed in 2005, 2007, 2009, 2010 and 2012. Background noise levels were found to be RC 48 (N), RC 32(R), RC 31, RC 20(R)/RC 26(MF) and RC 27(R)/RC 31(MF), respectively. These levels are non-compliant due to some of the following deficiencies:
1. Furnaces were oversized requiring a major balancing damper restriction to provide the required flow rate resulting in significant turbulence noise at mid and high frequencies.
2. Mechanical room doors opened into the classrooms compromising the mechanical room's envelope as was a recessed electrical panel in the mechanical room wall.
3. Exhaust fans were found to be another dominant noise source at low frequencies as were roof mounted exhaust fans not properly isolated from the structure.
4. A major path for low frequency fan energy resulted in fan vibration being transferred to the mechanical room floor, through the monolithic floor assembly, and then radiated out into the classroom.
5. No flex acoustic duct was installed between the balancing dampers and each diffuser or lined duct.

Based on this experience we relaxed the RC 30(N) spectrum requirement and introduced a new 1/3 octave band criteria that allows low frequency noise to be a bit higher for Modular Classrooms but is not objectionable even though ASHRAE would classify it with an (R) rating; this low frequency noise component does not interfere with speech.
In the 2015 assessment of a modular classroom prototype, the builder only had to implement some minor acoustical renovations by additional structural isolation of the mechanical room ductwork but the background noise was found to be compliant to the new criteria.

4 Conclusion

To achieve schools that provide good acoustics we have a task of educating the public and school administrators. Ideally we would like to show that there is a correlation between acoustics and student outcome.
However, we still have some pure acoustic work to be done:
1. Is sound reinforcement in a classroom necessary in a school with good acoustics?
2. How do we explain the difference in reverberation measurement in an empty gym and one with some diffusion?
3. Will movable walls ever provide hoped for isolation?
4. Can we define and agree on acoustic requirements for trades' education areas?

Acknowledgements

We would like to acknowledge the contributions of our predecessors at Alberta Infrastructure: Kelly Kruger, Erwin Rebke and Martin Fayers.

References

[1] American National Standards Institute, "ANSI S12.60-2010: Acoustical performance criteria, design requirements, and guidelines for schools," New York.

[2] Alberta Infrastructure, (2007). Standards and Guidelines for School Facilities.

[3] J. E. Dockrell and B. Shield, The Impact of Sound-Field Systems on Learning and Attention in Elementary School Classrooms *Journal of Speech, Language, and Hearing Research* 55, 1163–1176, 2012

[4] K. Packer, C. Faszer, Reverberation measurement and prediction in gymnasia with non-uniformly distributed absorption; The importance of diffusion *Canadian Acoustics / Acoustique canadienne* Vol. 39 No. 4 (2011) 47-53

[5] A.D. Lightstone Practical Aspects of operable wall isolation, *Canadian Acoustics / Acoustique Canadienne* vol.22 no.3(1994) 31

SOUND-FIELD DIFFUSENESS IN OBLIQUE-SHAPED REVERBERATION ROOMS WITH DIFFERENT TEST CONFIGURATIONS

Md Mehadi Hasan [†1] **and Murray R. Hodgson** [‡1]
[1]Department of Mechanical Engineering, University of British Columbia, Vancouver, BC, Canada.

1 INTRODUCTION

The accuracy of the reverberation-room methods, especially at low frequencies, has a long controversial history, which can be attributed to insufficient sound-field diffuseness [1, 2]. To find a solution to the problem, numerous studies have been carried out over the years. One of the recent studies [3] suggested that an oblique-shaped reverberation room having an ISO-prescribed volume of 150 m^3 [4] with the regular dimensional orientation (shortest vertical dimension) and a test configuration prescribed by standards would provide better sound-field diffuseness and, hence, better measurement accuracy than other reverberation rooms of different shapes and volumes. Based on this finding, the same study was extended to other test configurations, and their effect on sound-field diffuseness is addressed in this paper. A number of parameters like curvature (κ) of temporal energy-decay curves, spatial uniformity of the reverberant sound field (SPL), prediction accuracy of reverberation times (RT), prediction accuracy of absorption coefficients, etc. have been used as descriptors to quantify the degree of sound-field diffuseness. Results obtained with the help of a numerical finite-element-based modal approach are discussed. Finally, an optimal test configuration is recommended that would provide better field diffuseness.

2 THE REVERBERATION ROOMS

For a given oblique-shaped reverberation room of volume 150 m^3 with regular dimensional orientation, six different test configurations were considered here. The configurations considered are: Room #1 – with no diffusers (empty) (Fig. 1a); Room #2 – with diffusers (Fig. 1b); Room #3 – with absorbent corner treatments (Fig. 1c); Room #4 – with both diffusers and absorbent corner treatments (Fig. 1d); Room #5 – with Quadratic Residue Diffusers (QRD) (Fig. 1e); and Room #6 – with both diffusers and QRD (Fig. 1f). Predictions were performed with 1 source and 5 receivers in nine third-octave bands ranging from 50 to 315 Hz. Positioning of the sources and receivers was performed based on the standard's prescriptions [4].

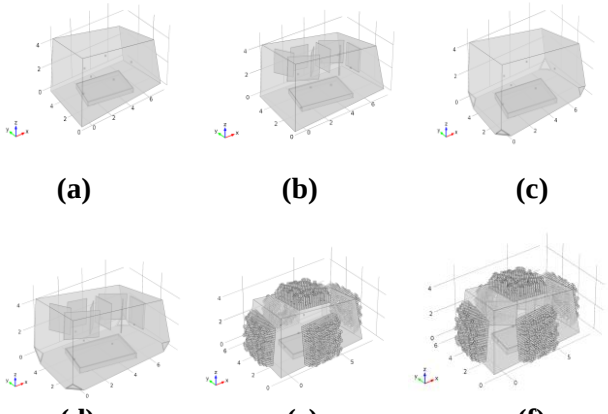


Figure 1: Different test configurations.

3 RESULTS

The curvatures of the temporal-decay curves in third-octave bands for the different test configurations are presented in Figure 2. It is noticeable that the curvatures are smaller than 10 for the room equipped with diffusers and absorbent corner treatments (Room #4) from the 125-Hz third-octave band. The curvatures are a bit higher for the rooms equipped with QRDs.

The SPL deviations predicted in third-octave bands for the different test configurations are presented in Figure 3. It is obvious that the deviations are smaller than the ISO-prescribed limit of 1.5 dB from the 160-Hz third-octave band for all of the test configurations except those equipped with QRDs. Overall, it is also noticeable that Room #4 (with diffusers and absorbent corner treatments) yields a smaller deviation from the 125-Hz band than the other test configurations.

The RT-prediction errors in third-octaves for the different test configurations are presented in Figure 4. It is obvious that the errors are smaller than 10% from the 125-Hz third-octave band for Room #4. However, the same is not true for the rooms equipped with QRDs; rather they show greater errors than the other test configurations.

The predicted absorption coefficients of a sample absorber in third-octave bands for the different test configurations are presented in Figure 5. It is noticeable that the predicted values from the 100-Hz band are closer to the

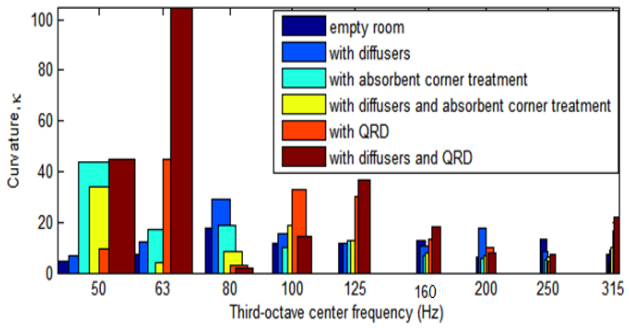


Figure 2: Curvatures of temporal-decay curves in third-octave bands for different test configurations.

† mehadi@alumni.ubc.ca
‡ murray.hodgson@ubc.ca

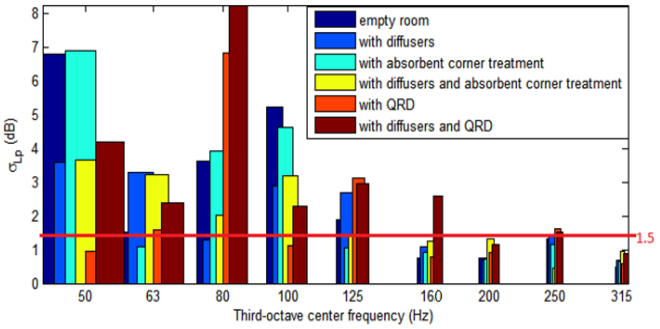


Figure 3: SPL deviations in third-octave bands for different test configurations.

expected value of 0.76 for the room with absorbent corner treatments (Room #3) and the room with both diffusers and absorbent corner treatments (Room #4). Surprisingly, the rooms with QRDs also give good accuracies in some frequency bands, which is not clear and cannot be explained.

4 DISCUSSION

Based on the results discussed above in terms of different descriptors, it is clear that reverberation rooms equipped with either diffusers or absorbent corner treatments, or both, yield better field diffuseness than other test configurations. It was also noticed that the prediction accuracy of the absorption coefficients of the sample absorber is better for the room equipped with both diffusers and absorbent corner treatments than for the other test configurations. Overall, it was noticed that the rooms with QRDs yield less accurate results; this can be attributed to the increased surface absorption, which results from a significant increase in surface area (due to the installation of the QRDs).

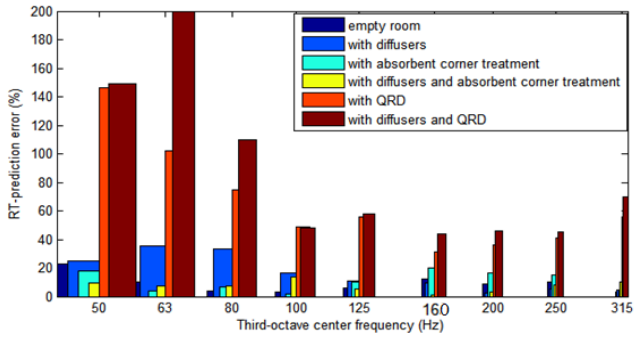


Figure 4: RT-prediction errors in third-octave bands for different test configurations.

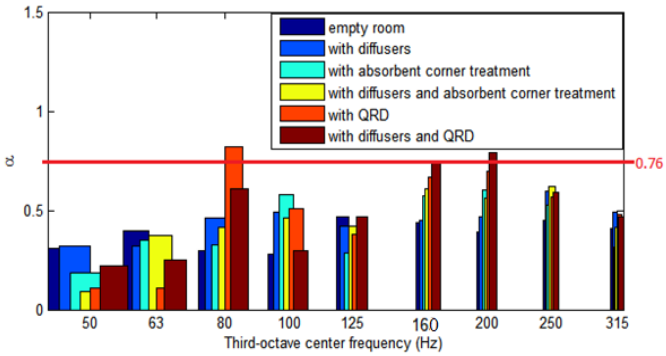


Figure 5: Sample absorber absorption coefficients in third-octave bands for different test configurations.

5 CONCLUSION

Overall, it can be concluded that an oblique-shaped reverberation room of volume 150 m^3 with regular dimensional orientation and equipped with diffusers and absorbent corner treatments would provide better diffuseness compared to other test configurations.

REFERENCES

[1] C. W. Kosten, "International comparison measurement in the reverberation room", *Acustica* 10, 400–411 (1960).

[2] M. Vercammen, "Improving the accuracy of sound absorption measurement according to ISO 354", *Proceedings of the International Symposium on Room Acoustics*, Melbourne, Australia (2010).

[3] M. Hasan and M. Hodgson, "Effectiveness of reverberation room design: Room size and shape and effect on measurement accuracy", *International Congress of Acoustics*, Buenos Aires, Argentina (2016).

[4] ISO 354, Acoustics—measurement of sound absorption in a reverberation room, *International Organization for Standardization*, Geneva, 2003.

SCALE-MODEL INVESTIGATION OF THE EFFECT AND OPTIMAL DESIGN OF SOUND ABSORBERS IN AN OPEN-PLAN OFFICE

Md. Amin Mahmud [*1] and Murray Hodgson[2]
[1,2]Acoustics & Noise Research Group, University of British Columbia,
SPPH, 2206 East Mall, Vancouver, British Columbia V6T1Z3, Canada.

1 Introduction

Porous sound-absorbing materials are frequently used in the ceiling areas of open-plan offices to reduce high reverberation times and improve speech-communication and speech-privacy conditions. The UBC Vancouver AERL building fourth-floor open-plan student area has a distinctive arrangement of porous sound absorbers. Tectum boards [www.tectum.com] are suspended from the ceiling in rows of 4 boards across the width of the space at two-metre intervals and also mounted flat against the ceiling at eight-metre intervals. It is not known how this configuration was chosen. Therefore, it was of interest to determine whether this existing configuration of the porous absorbers is the optimum one for acoustical performance. While a previous research study of Orlowski [1] investigated the effect of porous-absorber arrangements on acoustical performance experimentally in a 1:16-scale model from 1 to 64 kHz, it only investigated the effect of suspended baffles in a reverberation chamber (ie., in a diffuse sound field). The present work intends to investigate 15 different configurations of ceiling-mounted, suspended and sidewall baffles and their various combinations, including the current configuration, and find the optimum arrangement of the porous absorbers.

2 Methodology

2.1 Model construction

A 1:8-scale model of the selected zone of the AERL student space was used to investigate the performance of each proposed absorber configuration. The model was constructed as a rectangular box from sheets of 9-mm-thick plywood for simplicity of construction. To conserve materials, a concrete floor and concrete wall were considered as two of the surfaces of the space. 18 workstations of the selected zone were built on the roof of the model. The partitions of the workstations were made of cardboard covered with a single thickness of 3-mm-thick felt materials to add absorption similar to the existing AERL space. The top surface of the model is hinged to allow for convenience in placing the porous absorbers inside. The dimensions of the rectangular box and workstations are 2.45 × 1.4 × 0.46 m^3 and 0.25 × 0.25 × 0.25 m^3, respectively. Figure 1 shows a photograph of the model. Figure 2 shows its ceiling with the 18 workstations, and the source and absorbers on the floor.

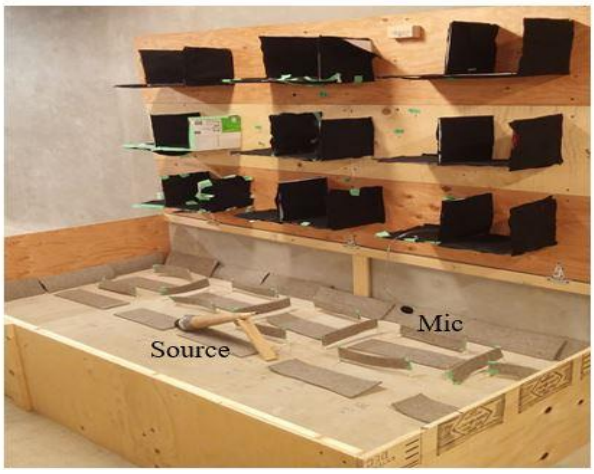


Figure 1. Photograph of the model (open roof) with work -stations showing source, receiver and sound absorbers.

2.2. Sound-absorbing material

The AERL student space has two sizes of tectum boards; size A (110 × 39 × 1.96 in^3) was used as the ceiling-mounted absorber and size B (78 × 21 × 1.96 in^3) as the suspended absorber. Hence, the baffles used in the test configurations were designed exactly as 1:8-scale versions of those boards using the 6-mm-thick felt material. Moreover, Size B absorbers were also used in a sidewall-mounted configuration with baffles on the floor inclined against the walls for convenience of arrangement.

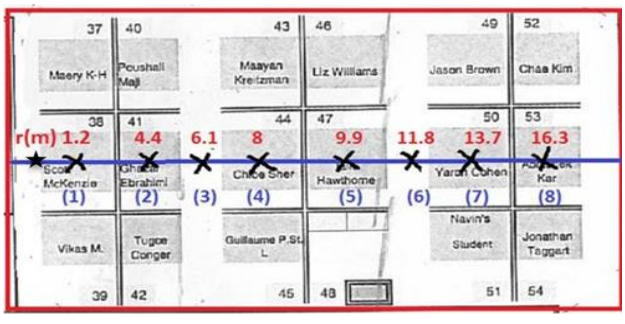


Figure 2. Photo of DL2 (★) source and (X) receiver positions in AERL floor plan.

[*1] amahmud@alumni.ubc.ca
[2] murray.hodgson@ubc.ca

2.3 Experimental setup and testing

The upper range of the available test equipment was limited to 20 kHz but, at 1:8 scale, it covers most of the dominant bands associated with human speech, which is the main source of sound in the AERL student space. Hence, the frequencies tested were from 1 to 16 kHz in octave bands corresponding to a range from 125 Hz to 2 kHz at full scale. To reduce the experiment set-up time, the floor of the model was considered as the roof of the student space. This allowed the sound-absorbing materials in the model to be placed on the floor as opposed to being hung from the roof of the model. The source sat on the floor and receiver microphones were inserted into the space through holes in the top of the model. The reverberation time of each configuration was calculated by averaging the values measured at six source-receiver positions near the centre of the model from 125 to 2000 Hz full scale, corresponding to the test 1:8-scale frequency bands from 1 to16 kHz. In this experiment Sound Level Decrease per Doubling of Distance (DL2) tests were done along a line along the length of the room in the centre of the model (to a maximum full-scale distance of 16.7 m – see Figure 2). Both the source and receiver were located at 15 cm from the ceiling of the model, corresponding to the approximately 1.2-m height of the listener's position above the floor in the workstations in a typical open-plan office area.

3 Results

Both the reverberation time, T_20, and DL2 of the 15 configurations of ceiling-mounted, suspended, current and sidewall configurations were compared with the values in the model with no absorber, and with the whole of the ceiling covered with absorbers.

3.1 Reverberation time

Results shown in Figure 3 show that the whole-ceiling configuration is the optimum configuration, while the combination of three absorbers is the second best and the no-absorber configuration is worst of all, as expected.

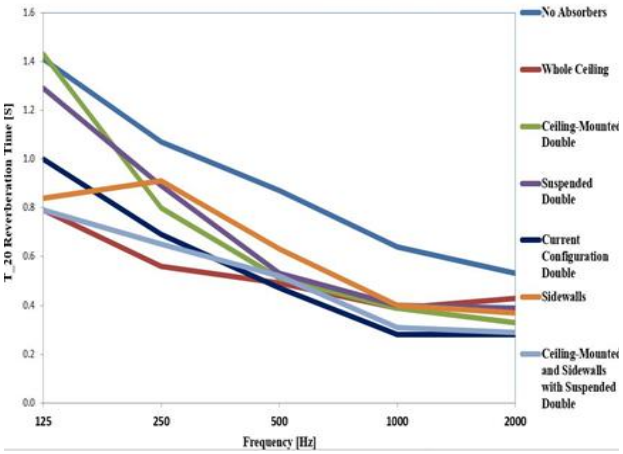


Figure 3. Overall comparison of reverberation time T_20 [s] for all absorber configurations.

3.2 Sound level decrease per distance doubling of (DL2)

As shown in Figure 4, the whole-ceiling configuration is the optimum of all absorber configurations, the current configuration doubled is the second best and the configuration with no absorbers is expectedly the worst of all configurations. It is very interesting to observe that DL2 for the current configuration double is better than the combination of three absorbers, which is the opposite trend of the T_20 results above.

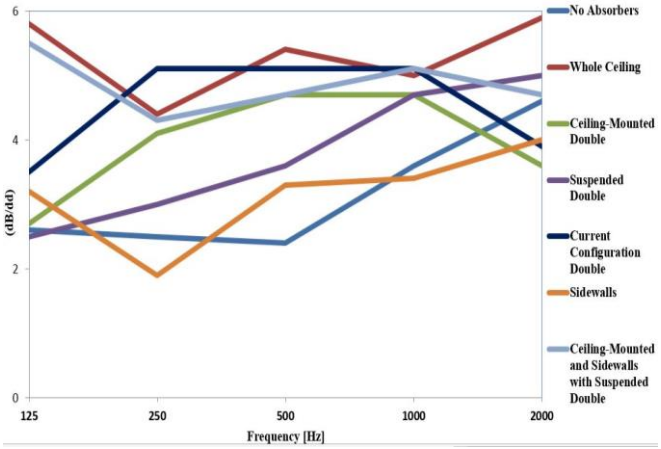


Figure 4. Overall comparison of sound level decrease per doubling of distance (DL2 in dB/dd) for all absorber configurations.

4 Conclusion

On the basis of both reverberation-time and sound level decrease per doubling of distance measurements, both the whole-ceiling configuration and the combination of all three types of absorbers perform much better than the current configuration of the AERL space. Doubling the absorber area in the current configuration seems to give a comparatively better DL2, but higher reverberation time, in contrast with the combination of all three absorbers. For both aspects, the double ceiling-mounted configuration performs the best among these three absorber types. Future work could investigate absorbers distributed over the sidewalls and determine the optimum absorber configuration for speech privacy between the workstations of the AERL open-plan area.

References

[1] Hodgson, M. R. and R. J. Orlowski. "Acoustic scale modelling of factories, Part I: principles, instrumentation and techniques." *Journal of Sound and Vibration*, 113(1): 29-46, 1987.

[2] Orlowski, R. J. "The arrangement of sound absorbers for noise reduction--results of model experiments at 1: 16 scale." *Noise Control Engineering Journal*, 22 (2): 54, 1984.

OBJECTIVE ACOUSTICAL EVALUATION OF HEALTHCARE OFFICE FACILITIES

Murray Hodgson*[1]
[1]Acoustics & Noise Research Group, University of British Columbia.
2260 West Mall, Vancouver, BC, Canada V6T1Z3

1 Introduction

The work reported here evaluated the acoustical quality of healthcare office facilities by way of physical measurement. The objectives were to determine the quality of the environments, the relationships of design features to it and how to design better environments?

2 Site descriptions

The study sites consisted of 17 healthcare office facilities on 30 floors in 17 buildings. In these, administrative staff employed by four local health authorities work. The floors were generally 2.7-m high from the carpeted floor to the suspended acoustical-tile ceiling (SAC). All buildings were mechanically ventilated. On five of the floors, professionally-designed sound-masking systems were installed in parts of the floors, and were in operation during testing. The spaces studied in these facilities included private offices (PO), shared offices (SO), open-plan offices (OPO), meeting rooms (MR), breakout and telephone rooms (B/TR) and lunchrooms (LR). OPOs varied considerably in size, shape and the number of cubicles/workstations. They had workstations arranged in various configurations, some separated by barriers, some separated by partial width or height barriers with heights varying from 1.2 – 1.8 m. In many cases these were located in cubicles of various configurations surrounded by partitions with heights varying from 1.5 - 2.0 m. While exact details of the construction of many of the internal partitions (IP) separating spaces in the facilities were not known, evidently many were of conventional gypsum wall board, metal-stud, usually with glass-fiber insulation, in some cases of modular construction as opposed to built-in-place. Some IP, particularly around enclosed offices, meeting and breakout rooms had glass sections; in one or two cases the entire IP was glass. Some IP rose to the ceiling slab, but many stopped at the SAC. Entrance IP contained hinged or sliding, solid wooden doors which closed against a jamb at the tops and sides, but with a gap of 0.5 to 2 cm at the bottom; in one or two cases, there was a door sweep.

3 Acoustical measurements

The following acoustical measurements were performed:

• back-ground noise levels: these were measured throughout the occupied spaces (BNLo); levels were measured in octave bands from 16-8000 Hz; from these total, A-weighted and NCB levels were determined.

• in selected unoccupied spaces the following para-meters were measured in 125 to 8000 Hz octave bands:

- Reverberation time (RT): average EDT and T20 values were determined from average impulse responses measured between 2 and 4 pairs of source and receiver positions. Values relevant to verbal-communication quality (vcq) were calculated by averaging values at 500, 1000 and 2000 Hz;

- Speech levels (SL), Speech Intelligibility Index (SII): sound-pressure levels generated by an omnidirectional source of pre-calibrated output-power levels were measured at selected receiver positions in the same and/or in adjacent spaces. From these, the source powers and the output powers of talkers using 'normal' voice level (SLn) [1], realistic octave-band and total, speech and A-weighted SLs (SLAn) were calculated. At each receiver position, the source-receiver distance, speech levels, EDTs and BNLo levels were used to calculate normal-voice speech-intelligibility indices (SIIn) [1, 2]. These were used to rate the speech intelligibility or speech-privacy quality at the receiver;

- Sound reduction with distance doubling (DL2): in 33 selected open-plan offices, DL2SA was determined from speech and A-weighted sound-pressure levels measured at available and convenient distances of 1 to 30 m from a calibrated omni-directional loudspeaker along lines crossing workstations;

- IP noise isolation (NI): the NIs of 99 selected IPs were measured as the difference between the sound levels on the source and receiver sides; from these Noise Isolation Class (NIC) values were determined.

4 Results

• Variation of BNLo with room type: Figure 1 shows the variation by room type of BNLo. Average noise levels were similar in SO, OPO and LR (LAtot = 50-53 dBA; NCB 45-50), and were slightly higher than in PO, BR/TR and MR, which were similar (LAtot = 45-47 dBA; NCB 40-43), as expected given the number of occupants in each type of room.

• Variation with RT with room type: Figure 2 shows the variation of average EDT and T20 with room type. All

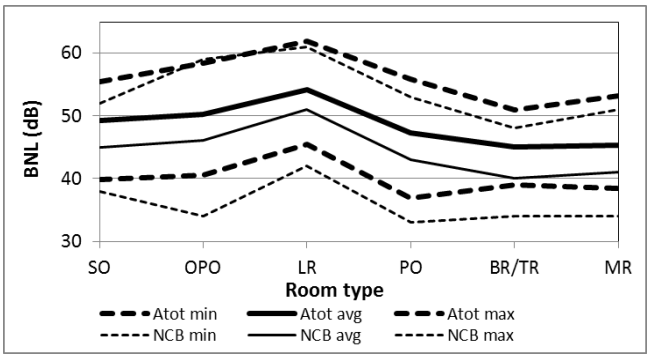


Figure 1: Variation by room type of the overall BNLo.

* murray.hodgson@ubc.ca

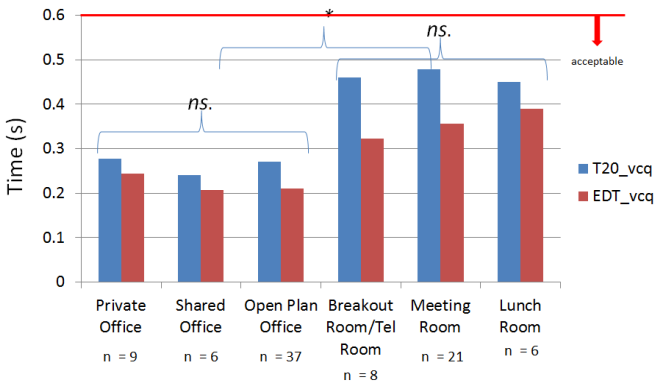


Figure 2: Variation of average EDT/T20 with room type.

average RTs were low – less than 0.5 s. Average RTs were similar in PO, SO and OPO (EDTvcq = 0.2-0.25 s; T20vcq = 0.24-0.28 s), and were slightly higher in BR/TR, MR and LR, which were similar (EDTvcq = 0.32-0.39 s; T20vcq = 0.45-0.48 s), The results are explained by the room volumes and shapes and, for OPOs, the high density of furnishings.

• Variation with room type of speech intelligibility, privacy: Figure 3 shows the variation with room type of the average SIIn at receiver positions 1 m from the source within one room or OPO workstation (SIIn,1m) and of the normal-voice speech privacy at the workstation in an adjacent room or OPO nearest to that containing the source (SIIn,nws). Excellent/acceptable speech intelligibility corresponds to SII > 0.75 / 0.6. Acceptable speech privacy corresponds to SII < 0.2. In all room types, speech intelligibility is acceptable (e.g., in SO, LR) or excellent (e.g., in PO, OPO, BR/TR, MR). Speech privacy is acceptable in PO, BR/TR and MR, but unacceptable in SO, OPO and LR. In the former rooms the receiver is separated from the source by an IP, in the latter there is no IP. Clearly acceptable speech privacy between a source and a receiver requires them to be separated by an IP.

• DL2SA: In the 33 cases measured, DL2SA varied from 5.9 - 14.0 dBA/dd (average = 9.0 dBA/dd; standard deviation = 2.1 dBA/dd). ISO 3382-3 standard [3] suggests that values above 11 / 9 / 7 dBA/dd represent 'excellent' / 'good' / 'fair' conditions for speech privacy, respectively, and values below 7 dBA/dd represent 'poor' conditions. If correct, then 2 (6%) / 11 (33%) / 13 (39%) / 7 (21%) of the OPOs measured have 'excellent' / 'good' / 'fair' / 'poor' conditions, respectively. In OPOs, conditions for speech privacy – and DL2SA –

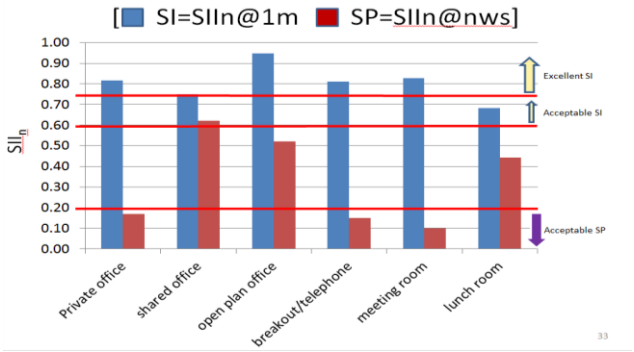


Figure 3: Variation with room type of SIIn,1m and SIIn,nws.

increase with lower ceiling height, sound-absorptive floors and ceilings, increased room lateral dimensions and increased height of partitions separating workstations, among other factors. All OPOs measured had low ceiling heights and absorptive ceilings and floors; partition heights, and room shapes varied.

• The noise isolations of the 99 IP varied from NIC 18-41. Structural concrete or brick walls had the highest values, a glass partition with doors with bottom gaps had the lowest.

• Empirical NIC prediction model: In preparation for developing NIC prediction models using multi-variable regression, all IPs were categorized with respect to the following features: absence or presence of a door; IP rises to SAC or to structural slab; percentage of glass; sliding or hinged door; absence or presence of door sweep; non-structural or structural IP; absence or presence of cavity insulation; modular or built-in-place construction; absence or presence of a flanking path. The only variables that were statistically significant in the analysis were: absence (0) or presence (1) of a door, non-structural (0) or structural (1) IP, absence (0) or presence (1) of cavity insulation, and modular (1) or built-in-place (0) construction. In particular, IP rises to SAC or to structural slab and percentage of glass in partition were not statistically significant. The optimal regression model, which explained 65% of variance, was:

$$NIC = 31.0 - 8.2\ door\ present - 2.0\ modular\ construction + 7.7\ structural\ partition + 2.2\ insulation\ present$$

According to this model, on average, a non-structural, non-modular IP without insulation or door has NIC 31. The presence of a door and of modular construction reduce the partition noise isolation by NIC 8.2 and NIC 2, respectively. Structural construction and cavity insulation increase noise isolation by NIC 7.7 and 2.2, respectively. Surprisingly, constructing the IP to rise to the structural slab instead of to the SAC did not make a statistically-significant difference in the noise isolation.

5 Conclusion

The results of this objective evaluation provide valuable information on the contributions of design features to workplace quality, and how to optimize it.

Acknowledgments

The author acknowledges with gratitude the valuable contributions of Leila Scannell, Denny Ng, Juliette Rauscher, Alice Lam and Tracy Liang in the realization of the work.

References

[1] ANSI S3.5, "Methods for calculation of the Speech Intelligibility Index", American National Standards Institute (1997).

[2] T. Houtgast, H. J. M. Steeneken and R. Plomp, "Predicting speech intelligibility in rooms from the modulation transfer function. I. General room acoustics", *Acustica* **46** (1980) 60-72.

[3] ISO 3382-3, "Measurement of room acoustic parameters – Part 3: Open plan spaces, International Standards Organisation (2009).

SUBJECTIVE ACOUSTICAL EVALUATION OF HEALTHCARE OFFICE FACILITIES

Murray Hodgson*[1]
[1]Acoustics & Noise Research Group, University of British Columbia.
2260 West Mall, Vancouver, BC, Canada V6T1Z3

1 Introduction

The work reported here evaluated the acoustical quality of healthcare office facilities by way of an occupant survey. The objectives were to determine: 1. the quality of the environments and the relationships of design features to it; 2. to what extent perceived acoustical quality is predictive of workplace-perception measures?

2 Site descriptions

The study sites consisted of 17 healthcare office facilities on 30 floors in 17 buildings. In these, administrative staff employed by four local health authorities work. The floors were generally 2.7-m high from the carpeted floor to the suspended, acoustical-tile ceiling (SAC). All buildings were mechanically ventilated. On five of the floors, professionally-designed sound-masking systems were installed in parts of the floors, and were in operation during testing. The spaces studied in these facilities included private offices (PO), shared offices (SO), open-plan offices (OPO), meeting rooms (MR), breakout and telephone rooms (B/TR) and lunchrooms (LR). OPOs varied considerably in size, shape and the number of cubicles/workstations. Many internal partitions (IP) were of conventional gypsum wall board, metal-stud, usually with glass-fiber insulation, in some cases of modular construction as opposed to built-in-place. Some partitions rose to the ceiling slab, but many stopped at the SAC. Entrance partitions contained solid wooden doors.

3 Questionnaire survey

A questionnaire survey to evaluate quality was developed based on work by Veitch [1] and Newsham [2], first by including their Environmental Features Rating (EFR). Factor analysis by Veitch [1] identified three workplace-perception factors: Satisfaction with Lighting (*Sat_L*), Satisfaction with Ventilation and Temperature (*Sat_VT*), and Satisfaction with Acoustics and Privacy (*Sat_AP*). The initial survey also included the following workplace-perception factors:

- Job Satisfaction (*Job_Sat*)
- Job Demands (*Job_Dem*)
- Physical-health symptoms, which involve the perceived frequency and intensity of 11 symptoms. This was grouped into two factors: *Health_Eyes* and *Health_Gen*.
- Emotions While at Work: The PANAS (Positive and Negative Affect Schedule) scale measures how much 10 positive (*PANAS_Pos*) and 10 negative (*PANAS_Neg*) emotions are experienced; from these, *Wellbeing* is calculated as *PANAS_Pos – PANAS_Neg*.

To the initial questionnaire, the following questions were added in preparation for further factor analysis:

- Workspace Overall Environmental Satisfaction section based on questions used by Veitch {1] and Newsham [2], but expanded to improve validity.
- MR/BR/LR-satisfaction section – questions were added to address sponsor interest in these spaces in this study.
- Workplace Related Needs Satisfaction section (WRNS) – this section was adapted from several existing scales with the objective of including questions relating to psychological factors found to be important to workplace satisfaction, including control, place attachment and person-environment congruence.
- respondent demographics section: age, sex, type of job, years in building, years in workstation, educational level.

4 Results

After data cleaning, valid response data was obtained from 635 respondents. Following are key results related to Satisfaction with Acoustics and Privacy (*Sat_AP*). Note that this workplace-perception factor combines acoustics and privacy, implying that the quality of the acoustical environment is uniquely associated with perceived privacy. However, in the questionnaire, privacy includes both acoustical and visual privacy. Attempts to differentiate between these two types of privacy using the questionnaire data were not successful; respondents appear to consider them to be equivalent. Following are some key results:

- Response distribution: Figure 1 shows the *Sat_AP* response distribution. More than 50% of respondents were dissatisfied, a more negative response than for either satisfaction with lighting or ventilation/temperature.
- Variation with office type: Figure 2 shows the *Sat_AP* response variation with office type; the differences were statistically significant. As expected, satisfaction was greatest in POs and worst in OPOs.

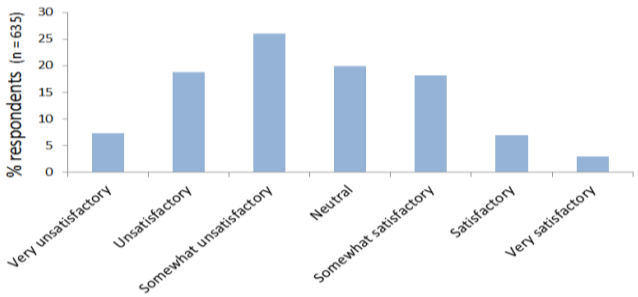


Figure 1: Distribution of responses for *Sat_AP*.

* murray.hodgson@ubc.ca

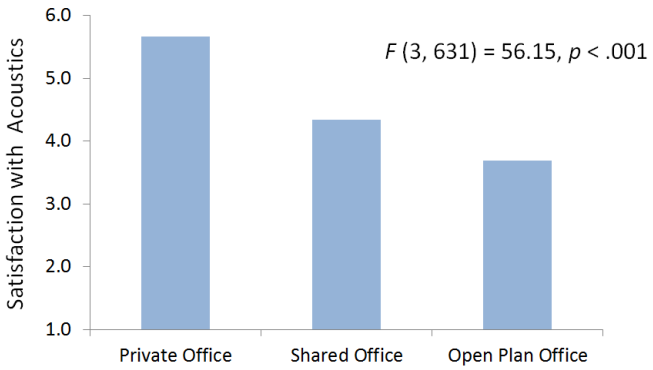


Figure 2: Variation with office type of responses for *Sat_AP*.

- Effect of sound-masking system: Figure 3 shows how a sound-masking system affects *Sat_AP*. Satisfaction was statistically significantly lower on floors with a system in operation. Note that this result does not mean that the masking system is responsible for the decreased satisfaction.

- Factor analysis was done on the responses to 39 questions. Existing previously validated scales discussed above (EFR, Physical health, PANAS, job demands) were excluded. The 39 questions included those in these sections:

- Workspace (overall environmental) satisfaction
- Workspace-Related Needs Satisfaction section
- Section with MR/BR/LR-specific questions, since these were of particular interest to the sponsors of this study.

The new workplace-perception factors identified were *WorkSatExt*, *Detach*, *Control*, *BRMR_L*, *BRMR_VT*, *BRMR_AP* and *LR_Overall*. *WorkSatExt* is an extended version of the Work Satisfaction factor used by Veitch and Newsham. *Detach* is a measure of perceived detachment from the workplace. *Control* is a measure of perceived control over the workplace.

5 Correlation analysis and regression modelling

Demographic factors (e.g., age and gender were not correlated with any workplace-perception factors.

Correlations between workplace-perception factors: Pearson's correlation coefficients between each pair of workplace-perception factors were calculated:

- as indicated by the correlation coefficients, to what extent does *Sat_AP* appear to contribute strongly (|r|>0.4) to various workplace-perception factors?: *WorkSatExt* = 0.759; *Control* = 0.582; *Sat_L* = 0.503; *Detach* = -0.421; *LR_Overall* = 0.414; *BRMR_AP* = 0.412; *Wellbeing* = 0.406; *Sat_VT* = 0.404. It strongly affect many factors.

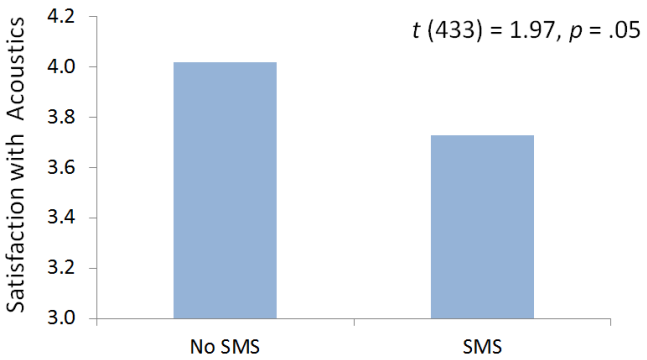


Figure 3: Average *Sat_AP* score on floors without and with a sound-masking system in operation.

Table 1: Relative contributions of (correlation coefficients beween) IEQ aspects and workplace-perception factors.

Factor	*Sat_L*	*Sat_VT*	*Sat_AP*
WorkSatExt	0.525	0.428	**0.759**
Detach	-0.224	-0.208	**-0.421**
Control	0.344	0.422	**0.582**
Health_Eyes	-0.334	-0.**366**	-0.340
Health_Gen	-0.293	-0.**404**	-0.381
PANAS_Pos	0.207	0.217	**0.329**
PANAS_Neg	-0.161	-0.218	**-0.291**
Wellbeing	0.244	0.291	**0.406**
Job_Sat	0.262	0.244	**0.368**

- as indicated by the correlation coefficients, what are the relative contributions of satisfaction with the three aspects of IEQ (lighting, ventilation/temperature and acoustics/privacy to other workplace-perception factors? Table 1 shows that *Sat_AP* is the strongest contributor to *WorkSatExt*, *Detach*, *Control*, *PANAS_Pos*, *PANAS_Neg*, *Welbeing* and *Job_Sat*, but not to *Health_Eyes* and *Health_Gen*, for which *Sat_VT* makes the greatest contribution.

- multi-variable linear regression was used to develop an optimal model for predicting *WorkSatExt* from the other factors. Following is the optimal model obtained. Note that *Sat_AP* is the strongest predictor factor:

$$\textit{WorkSatExt}\ (R^2 = 0.706) = 2.229 + 0.415(\textit{Sat_AP}) - 0.218(\textit{Detach}) + 0.216(\textit{Control}) + 0.154(\textit{Sat_L}) + 0.116(\textit{Job_Sat})$$

6 Conclusion

The results of this subjective evaluation have shown that occupant satisfaction with the acoustical environment often low, but is a major factor in determining the perceived quality of the workplace.

Acknowledgments

The author acknowledges the valuable contributions of Leila Scannell, Denny Ng, Juliette Rauscher, Alice Lam and Tracy Liang in the realization of the work reported here.

References

[1] J. A. Veitch, K. E. Charles, K. M. F. Farley and G. R. Newsham, "A model of satisfaction with open-plan office conditions: COPE field findings", *J. Env. Psych.* **27** (2007) 177-189.

[2] G. Newsham et al., "Do green buildings outperform conventional buildings? Indoor environment and energy performance in North American offices", Research Report RR-329, National Research Council Canada (2012).

OPTIMIZING CEILING SYSTEMS AND LIGHTWEIGHT PLENUM BARRIERS TO ACHIEVE CEILING ATTENUATION CLASS (CAC) RATINGS OF 40, 45 AND 50

Gary Madaras, Ph.D. [†1] and Andrew E. Heuer [‡2]
[1] ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638 USA
[2] NGC Testing Services, 1650 Military Road, Buffalo, NY 14217 USA

1 Introduction

In some buildings, sound isolation between rooms is important. Building codes, standards, guidelines and rating systems often require the level of isolation to be 40, 45 or 50 decibels (dB) at mid-frequencies regardless of whether it is measured as Sound Transmission Class (STC) or Ceiling Attenuation Class (CAC). As an initial construction cost savings in some buildings, interior walls are stopped at the height of a suspended, modular, acoustic ceiling. They do not extend full height up to the structural floor slab or roof above. Noise potentially can transmit more easily from room to room through the ceilings and the open plenum above. This often makes the ceiling the weakest link in the overall room envelope and the component that establishes the inter-room sound isolation performance. Ceiling manufacturers therefore test the sound blocking capacity of their ceiling panels and report the results as CAC_{panel} ratings.

There is approximately a ten CAC point decrease in the ceiling system rating in real buildings (CAC_{system}) due to the penetrations for open return air grilles, recessed lights and supply air diffusers. In the higher frequencies above the 500 Hertz (Hz) octave band, the decrease in isolation can be more than 20 dB.[1]

It is clear that modular acoustic ceilings alone cannot achieve the isolation levels required in the building codes, standards, guidelines and rating systems, especially once the flanking paths are considered. Prior research[2, 3] shows that lightweight plenum barriers that extend vertically from the top of the demising wall to the underside of the structural slab or roof above is the optimal design approach considering both performance and cost. The goal of this research is to optimize the combined performance of a modular acoustic ceiling system and lightweight plenum barriers of various materials and installation techniques in order to find solutions that comply with the required STC/CAC 40, 45 and 50 levels of performance.

2 Method

A series of CAC tests was performed on a suspended, modular, acoustic ceiling system with and without various lightweight plenum barriers under laboratory conditions. The tests were performed per ASTM E 1414 and E 413 at NGC Testing Services, a fully-accredited fire, acoustical, and structural/physical testing facility located in Buffalo, NY by a Senior Test Engineer. The laboratory is accredited by the National Voluntary Laboratory Accreditation Program (NVLAP) (Laboratory Code 200291-0). For the baseline test, the specimen comprised a standard, 24 mm (15/16") wide, metal, suspension grid, 16 mm (5/8") thick stone wool ceiling panels (CAC_{panel} 22), recessed troffer light fixtures, return air grilles and supply air diffusers per the layout shown in Figure 1, but no plenum barrier above the demising wall. Subsequent tests added various lightweight plenum barriers. Figure 1 also shows the location of a duct, PVC pipes and metal conduits that were suspended in the plenum above the ceiling and penetrated the plenum barriers. These elements were included so the study represented real-world applications more closely.

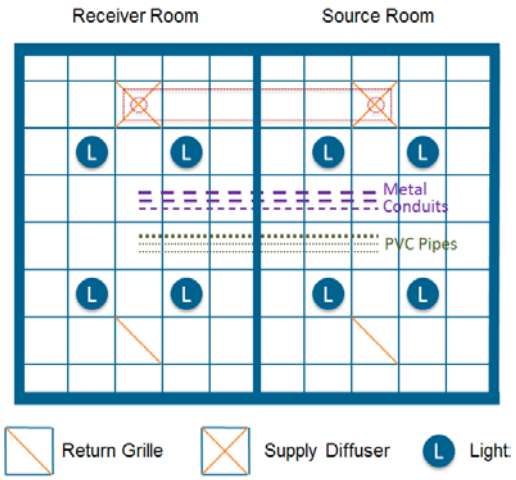


Figure 1: Reflected ceiling plan of test chambers showing locations of lights, return grilles, supply diffusers, supply duct, pipes and conduits.

Three types of plenum barriers were tested:

1. Drywall barriers (1 layer, 16 mm or 5/8" thick)
   a. Drywall alone, quick install method
   b. Drywall alone, sealed install method
   c. Drywall, sealed install, stone wool insulation 89 mm (3-1/2")
2. Mass loaded vinyl barriers (MLV, 4.88 kg/m^2 or 1 psf)
   a. MLV alone
   b. MLV with fiberglass insulation (89 mm, 3-1/2")
   c. MLV with stone wool insulation (89 mm, 3-1/2")
3. Stonewool barriers (30 mm, 1-3/16" thick)
   a. 1 layer
   b. 2 layers spaced 19 mm (3/4") apart.

For the drywall barriers, the *quick* installation method used simple, rectangular pieces of drywall. Large holes around elements that penetrated the barriers were stuffed with

[†] gary.madaras@rockfon.com | www.OptimizedAcoustics.com
[‡] aeheuer@ngctestingservices.com | www.ngctestingservices.com

scraps of stone wool insulation. Large gaps up to 13 mm (1/2") wide and 610 mm (24") long between drywall pieces and around edges and penetrations remained open. No sealant or tape was used to make the barrier airtight. For the *sealed* installation method, the drywall was cut close around penetrating elements and all joints, gaps and screw heads were sealed airtight with either sealant or metal tape.

The MLV barriers were screwed into metal channels along the tops and bottoms. Adjacent pieces of MLV overlapped 152 mm (6") and the joints were taped on one side to hold them in position. No sealant or tape was used around the perimeter or around penetrations. For the stone wool insulation barriers, the 30 mm (1-3/16") thick panels were oriented vertically, friction fitted at the tops and bottoms and taped along the tops, bottoms and abutted sides with metal tape.

3 Results

Table 1 provides the results of the study. CAC_{system} ratings and normalized transmission loss (Dn,c) in three, 1/3 octave bands are given for the base line stone wool ceiling and a reference 16 mm (5/8") thick wet-felted, mineral fiber ceiling (CAC_{panel} 37) as well as for the drywall, MLV and stone wool plenum barriers when they were combined with the baseline, stone wool, ceiling system. The mineral fiber ceiling results are included so that the relatively small effect of the type of ceiling panel on overall isolation can be seen compared to the more substantial effect of the plenum barriers being studied. Also, people are more familiar with mineral fiber panels than with stone wool ceiling panels, so inclusion of their data preempts the anticipated question of how a mineral fiber ceiling panel would compare.

Table 1: Ceiling attenuation class (CAC_{system}) rating and normalized transmission loss (Dn,c) in three 1/3 octave bands for various drywall, MLV and stone wool plenum barriers in combination with a suspended modular ceiling.

		Transmission Loss *normalized* (Dn,c)		
	CAC System	125 Hz	500 Hz	2000 Hz
Baseline Ceilings (no barrier)				
Stone wool	**22**	14	21	22
Mineral fiber	**27**	18	28	27
Drywall Barriers				
Quick install	**41**	21	37	44
Sealed install	**46**	25	46	50
Sealed install, stone wool	**50**	26	52	55
Mass Loaded Vinyl (MLV) Barriers				
MLV alone	**37**	18	34	42
With fiberglass	**42**	20	39	51
With stone wool	**44**	21	43	53
Stone Wool Barriers				
Stone wool - 1 layer	**38**	22	36	50
Stone wool - 2 layers	**48**	24	51	55

4 Discussion

Acoustic standards, guidelines and rating systems generally require isolation between rooms to be STC/CAC 40, 45 or 50. One might be led to believe that the lowest level of isolation (STC/CAC 40) can be achieved simply by using ceiling panels with CAC_{panel} rating of 40 or higher, but the noise flanking paths must also be controlled. This could affect aesthetics. For example, suspended lights avoid large ceiling penetrations, but they change the look of the room compared to recessed lights and could require higher ceilings. Custom, site-built noise control measures are required for specialty devices that penetrate the ceiling such as bed lifts, projector mounts and exhaust hoods. Controlling flanking path noise through the ceiling system is not typically part of a ceiling contractor's daily routine, so there is also an increased risk of failure. Lastly, once the building is occupied, flanking noise control measures above the ceiling can make access into the plenum difficult and maintenance personnel must understand the importance of the noise control measures and replace them correctly when their work in the plenum is complete. Using ceiling panels below CAC_{panel} 40 alone to block sound between rooms cannot provide adequate sound privacy. Using a modular acoustic ceiling system alone cannot provide the two higher performance levels required in the standards (STC 45 and 50).

5 Conclusion

The results of this research are consistent with prior research[2, 3] by others and suggest an alternative approach of using a lightweight plenum barrier that extends from the top of the demising wall to the underside of the deck above, in combination with a suspended ceiling system, may be optimal. These plenum barriers can be made of drywall, MLV, stone wool insulation or a combination. The resulting CAC_{system} values can reach the STC/CAC 40, 45 and 50 levels required by the acoustic standards, guidelines and rating systems.

Acknowledgments

The authors would like to thank ROCKFON and NGC Testing Services for their support and assistance in conducting this study and disseminating the findings.

References

[1] G. Madaras, A. Heuer, "Effects of Noise Flanking Paths on Ceiling Attenuation Class (CAC) Ratings of Ceiling Systems and Inter-Room Speech Privacy," Proc. Inter-Noise 2015, edited by Courtney Burroughs and George Maling, paper *in15_083.*

[2] K. Kruger, "The Effect of Various Parameters on the Sound Isolation between Offices with Suspended Ceilings," *Canadian Acoustics*, 16 (2), (1988, pgs 9-23).

[3] R.E. Halliwell and J.D. Quirt, "Controlling Interoffice Sound Transmission through a Suspended Ceiling," *J. Acoust. Soc. of Am.*, 90 (3), (September 1991).

THE DYNAMIC RESPONSE OF COMPOSITE FLOATING FLOORS IN BUILDINGS

Max de Castro Magalhaes [*1]
[1]Structural Engineering Department, Universidade Federal de Minas Gerais, Av. Antonio Carlos, 6627, 31270-901 Belo Horizonte, Brazil

1 Introduction

Recently, concerns about sustainability and green technology have been focused on the construction industry. Many buildings are nowadays constructed considering energy saving methods, alternative materials and the use of residues.

Nowadays vegetal fibers are widely used as a substitute to synthetic fibers for acoustics absorption purposes. Vegetal fibers such as the coconut fiber might be considered an alternative option to be used as acoustic materials. The coconut is a natural organic resource which is the seed-hair fiber obtained from the husk (mesocarp) of the coconut. There are many advantages on using vegetal fibers such as those made of coconut. For instance, they are renewable, cheaper and offer no risk to the human health during handling and processing. In the building construction industry most sound absorbing products are based on foams, glass or mineral fiber materials.

The use of floating floors on building construction is well-known among civil engineers, architects and acoustic space designers. They are popular not only for their ability to decrease the transmission of structure-borne sound throughout the building structural components but also for their slender dimension which may be relevant on the calculation of the building total cost price. Thus, various researchers have concentrated their work on presenting the main advantages of recycled materials, such as rubbers, in floating floor systems for sound isolation purposes.

The aim of this paper is to investigate the efficiency of using a combination of coconut fiber and recycled rubber for the vibration isolation of structural slabs in buildings. In section 2, experimental tests were made in order to characterize the coconut fiber boards and rubber in terms of their acoustic impedance (sound absorption) and mobility respectively. In section 3, a modal model was implemented. For this model, the screed and the resilient material are assumed as a one-dimensional mass-spring system lying on a simply-supported elastic floor. In section 4, a Finite Element (FE) model was developed using commercial software. The floating floor system was modeled using laminated (layered) shell elements composed of two different elastic materials, i.e. coconut fiber and recycled rubber. Finally, the conclusions are presented in section 5.

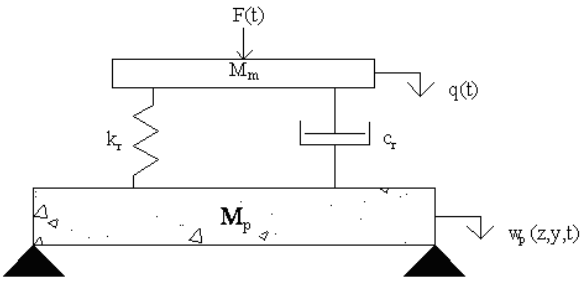
2 Method

Structure-borne vibration transmission was measured between a reinforced concrete plate and its connecter support at low frequencies.

Two different configurations (with and without a resilient layer) were taken into account. The structural plate considered was made of reinforced concrete and simply-supported on its four corners. Its dimensions were 10 cm, 1.2m and 1.16m for the thickness, length and width respectively. The concrete density was 2500 kg/m^3. A rubber quilt made from 100 % recycled rubber (thickness equal to ¼ ") was glued to the topside of the structural plate. A thinner concrete plate was placed on the resilient material, representing the screed.

The average vibration level of the structural concrete plate was measured when the screed was excited by an impulsive force. The vibration source was a plastic-headed hammer. It was used to hit the floor at different locations (in order to obtain space averaging) over a 5 second measuring duration. The average point mobility of the wall was obtained by measuring the impact force and the acceleration. The velocities were determined from the accelerations using the assumption of harmonic time dependence. The coherence function and the corresponding space-averaged mobility amplitude were measured. It is seen that the resonance frequencies are sensitive to the boundary conditions, i.e. the viscoelastic properties of the resilient material (recycled rubber). The results are shown for a narrow frequency band 50-500Hz. It is seen that with the introduction of a resilient quilt the sound insulation was significantly improved. The use of viscoelastic rubber under the screed caused a drastic reduction of at least 20 dB over the whole frequency range 50-500 Hz. The dynamic stiffness per unit area of the specimen was then evaluated.

Figure 1 below shows the modal model considered. It was composed of two connected subsystems: a mass-spring (screed and resilient material) and a plate (concrete slab). The structure subsystem corresponds to the interaction between a floating screed, a resilient quilt and a floor structural plate. For the resilient quilt, two different materials were considered herein: a) recycled rubber and b) coconut fibber.



[*] maxdcm@gmail.com

Figure 1: Floating floor system used in buildings in order to improve the sound insulation properties of the structural floor.

The modal response of the plate w_p (modal displacement) was calculated as:

$$-\omega^2 w_p + j\omega\beta_p w_p + \omega_p^2 w_p = \frac{F_p}{\Lambda_p} \quad (2)$$

$$-\omega^2 q + j\omega\frac{C_r}{M_m}q + \omega_o^2 q = \frac{F_o}{M_m} + \left[\omega_o^2\phi_p(z_o,y_o) + j\omega\frac{C_r}{M_m}\phi_p(z_o,y_o)\right]w_p \quad (3)$$

$$F_p = K_r\left[q\phi_p(z_o,y_o) - w_p\phi_p^2(z_o,y_o)\right] + j\omega C_r\left[q\phi_p(z_o,y_o) - w_p\phi_p^2(z_o,y_o)\right] \quad (4)$$

where q is the steady-state amplitude of the mass M_m (screed), β_p is the generalized modal damping coefficient for the structural floor; ω is the excitation frequency in radian/s and F_o .is the force applied on the limp mass M_m. The terms $j\omega\beta_p$ and $j\omega\frac{C_m}{M_m}$ are added on the left-hand side of Eqs. (2) and (3) in order to represent the damping of the flexible structural partition and lump mass. Λ_p is the modal mass of the structural mode *p*.

3 Results

Figure 2 shows the average mean-square velocity of a simply-supported structural floor subjected to a concentrated load. A point load was applied at the point (z_o, y_o) where z_o =0.3 m, y_o =0.4 m. It is seen that the spatial average mean-square velocity decreased as the frequency increases. The performance of the floating floor system is similar for the use of either rubber or coconut fiber.

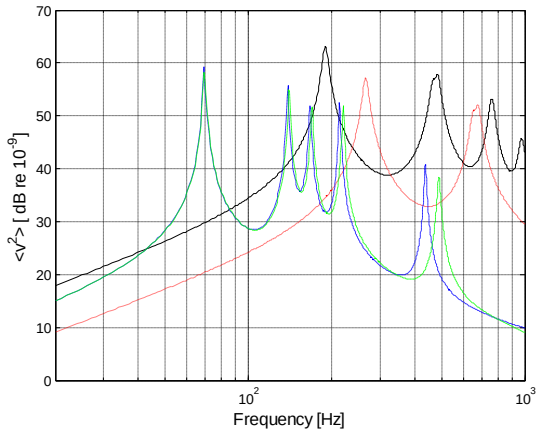


Figure 2: Average mean-square velocity of a simply-supported structural floor. Top black curve (bared structural floor); Middle red curve (floor+screed); Bottom blue curve (floor+rubber + screed); Bottom green curve (floor+coconut fiber + screed)

Figures 3 below shows the layout configuration used on the FE model analysis.

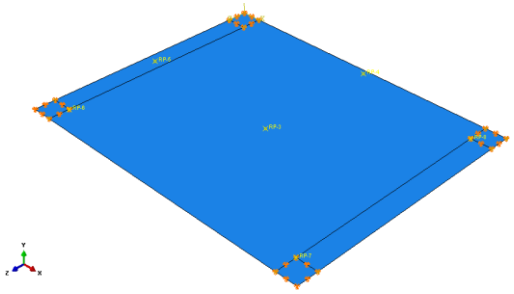


Figure 3: Boundary condition for the floating floor which is supported on its four cornes.

4 Conclusions

The main goal of this paper was to examine the variability of floating-floor efficiency, in terms of its mobility, to different configurations of rubber-coconut layouts. This study was aimed at providing not only a better understanding of the impact transmission mechanism in itself but also to produce a useful set of data which for instance can be used by acousticians as input data for a SEA analysis. This data might be useful for optimizing sound insulation in buildings at low frequencies, where the modal behaviour of rooms strongly influences the transmission.

Although research presents the results for a few resilient material configurations, in principle the same procedure can be applied to any other floating floor system with complex features. As future work, the possibility of using other types of material configuration (e.g. rubber//polymer, rubber//glass fiber, etc) might be another area of investigation. In addition, it will also be very useful to compute the sound radiation pattern.

Acknowledgments

I gratefully acknowledge the financial support provided by CNPQ (National Counsel of Technological and Scientific Development), UFMG (Federal University of Minas Gerais) and FAPEMIG (Research Foundation of Minas Gerais).

References

[1] Cox, T. J. And D'Antonio, P. *Acoustic Absorbers and Diffusers.* London, Spon Press, 2000.

[2] Kino, N. and Ueno, T. Improvements to Johnson-Allard model for rigid-framed fibrous materials; *Applied Acoustics Journal* 68, 1468-1484, 2007.

[3] Santiago, B. X. Development of project for the process of generation of coconut fiber residuos. In *5th European Conference on Steel and Composite Structures*, 2:1659–1664, 2008.

EXTERIOR NOISE INGRESS IN GREATER VANCOUVER RESIDENTIAL BUILDINGS: PREDICTION VERSUS FIELD MEASUREMENTS

G. Mak, P.Eng. [*1] and M. Bliss, P.Eng. [†1]
[1]BKL Consultants Ltd., #308-1200 Lynn Valley Road, North Vancouver, BC V7J 2A2

1 Introduction

Many municipalities in Greater Vancouver prescribe acoustical requirements for residential and mixed-used development permitting applications in order to control exterior noise (e.g., road, rail and aircraft traffic noise) to noise sensitive portions of dwelling units such as bedrooms, living/dining rooms, kitchen, and bathrooms. Hence, an assessment is performed to determine the necessary façade construction details to meet prescribed interior noise limits.

ISO 15712-3 *Building acoustics – Estimation of acoustic performance of buildings from the performance of elements – Part 3: Airborne sound insulation against outdoor sound* [1] is considered by many to be the current best practice for predicting noise transmission through the building envelope.

IBANA-Calc [2], an indoor noise prediction software developed by the National Research Council Canada (NRC), is an alternative commonly used in Canada for similar noise ingress calculations. IBANA-Calc was specifically developed to predict sound insulation against aircraft noise, but the software allows for other noise source inputs.

A comparison of noise predictions using the two calculation methods, with the same inputs, shows that they generate different results. For a number of residential development projects, BKL had the opportunity to conduct post-construction measurements to determine the actual 24-hour noise levels inside suites and compare them with predictions performed with IBANA-Calc and ISO 15712-3 to assess the accuracy and identify potential causes for differences.

2 Sound Insulation Predictions Overview

With ISO 15712-3 and IBANA-Calc, interior noise levels are calculated based on the facade noise exposure, transmission loss of the building envelope elements, receiving room dimensions and the amount of absorption in the receiving room. This calculation is performed for each exterior facade of a room and added together to estimate the indoor noise level.

2.1 Facade Noise Exposure

Exterior noise levels are typically estimated using site measurements, theoretical calculations, or a combination of the two.

2.2 Transmission Loss

Spectral transmission loss (TL) data for each facade component (wall, window, etc.) is used to perform sound insulation calculation for ISO 15712-3 and IBANA-Calc. TL data is often obtained from laboratory measurements, field measurements, or theoretical calculations depending on which is available and judged to be more reliable.

2.3 Receiving Room Absorption

In IBANA-Calc, the room absorption can be specified as a percentage of floor area, ranging from 50% to 150%. 100-120% are commonly used values, which corresponds to reverberation times of 0.3-0.4 seconds for a typical floor-to-ceiling height of 2.4 m.

Instead of specifying a specific amount of absorption in the receiving room, ISO 15712-3 normalizes all indoor noise sound pressure levels to a reference reverberation time of 0.5 seconds.

2.4 Calculation Method

ISO 15712-3 defines the indoor sound level resulting from traffic noise transmission through a building element as:

$$L_2 = L_1 - TL + 10 * log\left(\frac{S}{A}\right) + 3,$$

where,
L_1= free-field sound level on the outside surface
L_2 = sound level in the receiving room
TL = transmission loss value
S = area of building facade
A = equivalent sound absorption area in receiving room

However, IBANA-Calc, being developed for aircraft noise ingress measurements, uses the following equation, which better correlated with their test results:

$$L_2 = L_1 - TL + 10 * log\left(\frac{S}{A}\right)$$

3 Post-Construction Measurements Overview

BKL conducted post-construction measurements in four wood-frame residential construction developments where a pre-construction acoustic assessment had also been completed. Indoor and outdoor measurements were performed simultaneously. Reverberation time measurements were also conducted to compare with the assumptions made in IBANA-Calc. The sound absorption for the modelled rooms was then adjusted to reflect actual conditions during measurements by matching the broadband reverberation time that was measured.

Table 1 below summarizes the measurements taken:

[*] mak@bkl.ca
[†] bliss@bkl.ca

Table 1: Summary of measurements taken

Building	Floor/ Room Type	Noise Source	Description[1]
Richmond House	2/F Bed	Hwy/ Aircraft	• No carpet; unfurnished • 3-13-3 window • VIN_PLY9_WS89_ GFB89_G13
Vancouver Townhome	2/F Living	Arterial Road	• No carpet; Unfurnished • 3-13-3 window • HARDI9_PLY13_ SWS89_2G13
Vancouver Townhome	1/F Den	Arterial Road	• Carpeted; Unfurnished • 6-13-4 window • HARDI9_PLY13_ SWS89_2G13
Vancouver Condo	2/F Living	Arterial Road	• Carpeted; unfurnished • 5 window • STUC23_PLY9_WS140 _ GFB152_G13
Vancouver Condo	2/F Living	Arterial Road	• Carpeted; unfurnished • 5 window • STUC23_PLY9_WS140 _ GFB152_G13
Vancouver Condo	4/F Bed	SkyTrain	• Carpeted; unfurnished • 3-13-3 window • 5-13-3 sliding glass door • VIN1_PLY13_WS140_ GFB152_G13

[1] Numbers denote thickness in mm
Window/Sliding Glass Door: Glazing-Airspace-Glazing thicknesses
VIN: Vinyl Siding
HARDI: Hardi Panel Siding
PLY: Plywood
WS: Wood Studs
GFB: Glass Fibre Batt Insulation in Stud Cavity
RC: Resilient Channels
G: Gypsum Wall Board

4 Results

Measured indoor noise levels were corrected to a reverberation time of 0.5 seconds for comparison with ISO 15712-3 predictions. As-measured results were compared with IBANA-Calc predictions that were adjusted to match the measured reverberation times. Table 2 below shows the summary of the comparison between predicted and measured indoor sound levels.

Table 2: Calculated difference between predicted and measured indoor sound levels (dBA) for ISO 15712-3 and IBANA-Calc

Building	Floor and Room Type	ISO 15712-3	IBANA-Calc
Rmd House	2/F Bed	-2	-5
Van TH	2/F Living	4	-1
Van TH	1/F Den	1	-8
Van Condo	2/F Living	-3	-9
Van Condo	2/F Living	-4	-10
Van Condo	4/F Bed	-2	-6
Average Difference		-1.0	-6.5
Standard Deviation		3.0	3.3

5 Discussion

There are a number of factors which could contribute to the discrepancies between the prediction results.

Noise Transmission Equation: The most significant difference between the ISO 15712-3 and the IBANA-Calc calculation is the +3 dB term which results in lower predictions from IBANA-Calc.

Exterior Noise Exposure Definition: In this study, exterior noise was measured using microphones mounted on the surface, 2 m from the surface or in the free-field, and then corrected to match the specific input of ISO 15712-3 and IBANA using the assumption $L_{1,free\text{-}field} = L_{1,2m} - 3 = L_{1,surface} - 6$, as described in ISO 1996-2 [3]. However, this equation may not hold true in some cases, which may result in incorrectly defining the exterior noise exposure.

Transmission Loss Assumptions: Since lab TL data was used in the calculations, the in-situ quality of construction could be a significant source of error.

Flanking Transmission: ISO 15712-3 also considers flanking noise transmission, which depends on the wall, floor and ceiling material properties and the type of connections.

Room Reverberation: Considering the reverberation time assumption difference between ISO 15712-3 and IBANA-Calc (0.1-0.2 s), this could result in up to a 2 dBA difference. IBANA-Calc predictions would be lower by using the default values.

6 Conclusion

Post-construction measurements were performed at four wood-frame construction buildings and the results were compared with noise predictions calculated using ISO 15712-3 and IBANA-Calc. Assuming the measurements conducted are representative, the ISO 15712-3 method provides significantly more accurate results for ground-based transportation noise.

References

[1] International Organization for Standardization (ISO). 2005. Building acoustics – Estimation of acoustic performance of buildings from the performance of elements – Part 3: Airborne sound insulation against outdoor sound. Reference No. ISO 15712-3:2005. Geneva, ISO.

[2] Birta, J., Bradley, J.S., Estabrooks, T. 2001. IBANA-Calc User's Manual. Ottawa, National Research Council.

[3] International Organization for Standardization (ISO). 2007. Acoustics – Description, measurement and assessment of environmental noise – Part 2: Determination of environmental noise levels. Reference No. ISO 1996-2:2007. Geneva, ISO.

ACOUSTICAL DESIGN FOR 21ST CENTURY LEARNING

Paul E Marks
BKL Consultants Ltd., 308-1200 Lynn Valley Road, North Vancouver, BC V7J 2A2, Canada.

1 Introduction

The contemporary design of schools and buildings used for educational purposes has to not only consider the need of students and teachers but also reflect the world of business, retail and commerce in the rapidly changing 21st century world. The use of web-connected interactive, multimedia tools permits access to information and resources at fibre-optic speeds without the need for any of traditional learning media or instruction. At home, children and young adults have access to a wide variety of information from an array of online devices, which are enabling the quest for greater knowledge and control.

The contemporary role model works in an open-plan hi-tech office environment with a wireless connection to a mobile tablet device, communicating over social media using a 4G smart phones whilst drinking an almond milk americano misto from the in-house corporate barista. OK, so maybe this is not everyone's workplace, but in today's interconnected and technology driven world, a working environment does not need to be a physical location like a standard 10' by 15' office. It is more likely to be a shared or serviced open plan space with hot-desks but it can also be virtual, remote or, perhaps more succinctly, an online IP location somewhere on the web.

Educational experts believe a similar approach should be used for schools. The 21st century learning environment should enable and accommodate the learning needs of every individual learner while supporting the betterment of a group through positive relationships. 21st century learning should take place in the context of "promoting interaction and sense of community that enable formal and informal learning"[1]. To this end, 21st century architecture in schools or learning studios should allow for group, team and individual learning with a design flexibility that includes moveable furniture and walls that allow space to be re-configured.

Such flexibility in room-spaces can provide challenges in terms of the acoustic environment. The need for children to have a clear understanding of speech, whether from teachers or their contemporaries remains a critical requirement for any learning environment. Traditional classroom arrangements with a teacher providing instruction from the front of the room to children sat in rows is a relatively simple acoustic model – a single sound source located broadly one position transmitting instruction to receivers throughout the space. Providing the space contained sufficient reverberation control in terms of a fiberglass ceiling tile and perimeter walls that adequately isolated sound from entering the space the signal to noise ratio was sufficient for individual learning.

* marks@bkl.ca

In 21st century spaces, the sound transmission path from source to receiver is not so clearly defined as the primary source (i.e. teacher) is encouraged to move around the space and the receivers (i.e. students) may be in a large group, or small teams or may even be on the third floor atrium listening via the loudspeakers on a laptop. There simply is no "*simple*" model anymore.

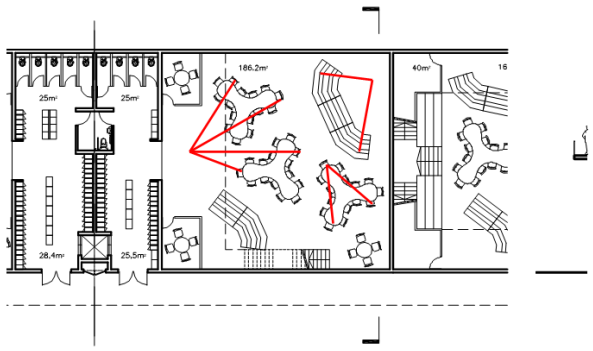


Figure 1: Potential source and receiver locations within a 21st century learning environment.

Furthermore, the sound source maybe a child or young adult with a distinct voice spectrum or an interactive MP3 file being played over the web for the benefit of a wider listening group. In these circumstances, the clarity of sound will inevitably be in a direct relationship to not only the amplitude but also the reverberant conditions and underlying sound levels in the space, which may be attributable to building services noise or noise ingress from exterior road traffic.

2 Design standards and speech intelligibility

Acoustic design standards for schools are typically defined by the School Board, with a general reliance on objective standards from good practice guides such as ANSI/ASA S12.60[2] and the ASHRAE Handbook[3]. Whilst these guidelines provide a wealth of guidance for the acoustic requirements of sound isolation, underlying background sound levels and reverberation in core-learning spaces little or no information is provided about speech intelligibility. Speech intelligibility is an important measure of the effectiveness or adequacy of a communication system in a noisy environment.

The UK Government Department for Education and Skills document Building Bulletin 93 (BB93) Acoustic Design of Schools: Performance Standards[4] provides design guidance for educational boards, architects and engineers involved in the design of both new and renovated school buildings. The guidance recognizes that open plan spaces require additional specification due to the complex acoustic conditions and provides specific recommendations for the speech intelligibility and privacy performance standards

during instructional group or critical activities and between workgroups. In some cases, supporting computational predictive acoustic modelling of the open plan space is required as part of the building code approval.

3 Acoustical design to improve speech intelligibility

As a starting point, the acoustician should seek to minimize the variables under their direct control. The potential for intrusive sounds can be reduced, but not necessarily eliminated, by the provision of large amounts of acoustic absorption within the space. Acoustically absorbent surfaces should be maximized with the use of not only highly efficient ceiling and wall treatments but also soft furnishings and carpeted floors. The use of "soundclouds" or acoustically absorbent wall panels can offer potentially delineated areas within the space that have low reverberant conditions ideal for individual learning.

The underlying background sound level should be low, although not unduly quiet, and more importantly constant and steady in nature. The over-reliance on mechanical services to provide background sound should be avoided and consideration should be given to the use of white or colored noise systems in open plan areas as the potential acoustics benefits are well-merited.

Finally, the use of operable screens or moveable furniture should be utilized to provide barriers to sound transmission paths within the open plan areas. Acoustic glass wall dividers and operable walls are capable of achieving levels of sound isolation commensurate with the objective standards outlined within the best practice guidelines. However, in areas that are recognized as noise generating (e.g. music practice rooms or drama spaces) extra provisions would be necessary to achieve higher sound isolation.

The use of moveable raked bench seating not only provides opportunistic physical barriers but also increases the potential for sound absorption within the space. Moveable furniture allows schools to not only orientate spaces as they require but also enables short-lines of sight between teachers and students, which increases the speech intelligibility within the workgroups.

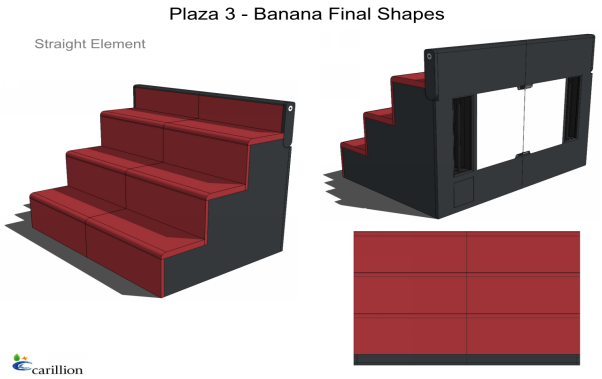


Figure 2: An example of fabric covered raked mobile seating (*courtesy of Carillion UK plc.*)

4 Conclusion

Overall, while 21st century learning presents considerable challenges to the acoustician, securing appropriate acoustic conditions for group, team and individual learning that also includes design flexibility is fully achievable with adequate planning and the inclusion of measures that increase sound absorption, reduce noise intrusion and promote clear, short-lines of sight for improved levels of speech intelligibility.

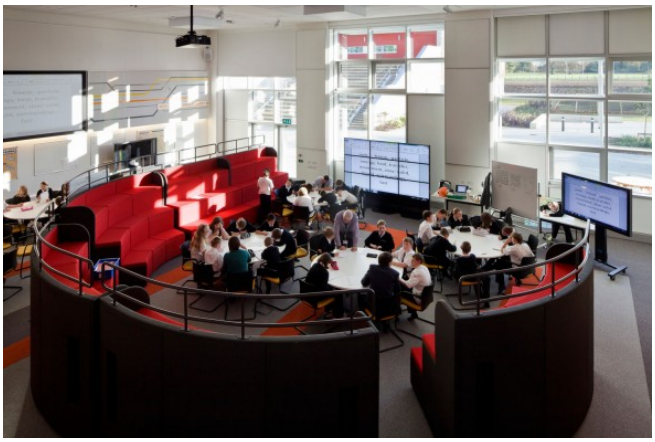


Figure 3: An example of a 21st century learning environment.

References

1 Cornell, P. (2002). "The Impact of Change in Teaching and Learning on Furniture and the Environment." In Chism, N. & Bickford, D. (2002). The Importance of Physical Space in Creating Supportive Learning Environments. New Directions for Teaching and Learning. No. 92, Winter 2002. San Francisco: Jossey Bass

2 ANSI/ASA S12.60-2010/Part 1, American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools, Part 1: Permanent Schools

3 ASHRAE Handbook. Heating, ventilating, and air-condition systems and applications

4 Building Bulletin 93 Acoustic design of schools: performance standards, December 2014 UK Department of Education & Education Funding Agency

NOISE CONTROL ISSUES IN A STEEL-FRAME INSTITUTIONAL BUILDING

Clair W. Wakefield [1]
[1]Senior Consultant, RWDI (formerly Wakefield Acoustics Ltd.)
301-2250 Oak Bay Avenue, Victoria, B.C. V8R 1G5

1 Introduction

The replacement of Oak Bay High School's original 1930's and 1950's buildings with a single school/community facility was largely completed for the start of the 2015-16 school year. The new building features two gymnasia, a large atrium/common space, band and choral rehearsal rooms, drama and dance studios, and a 400-seat community theatre. The classrooms, laboratories and Community and First Nations spaces are housed in two and three-storey sections of the building.

The building is of steel-frame construction with floor sections of concrete-topped, corrugated steel (Q-deck). In most interior locations, the steel columns and diagonal cross-bracing are located inside conventional steel stud and gypsum board walls. Because of the need to accommodate structural steel elements of various widths and achieve a range of lab-rated Sound Transmission Class (STC) targets (from STC 53 for classrooms to STC 60 for the wall separating the Theatre form the Atrium), a large number of unique wall sections were required. The band, choral and dance spaces are over 6 m high with exposed steel roof trusses.

The steel-frame constructions described above led to several acoustical separation issues including excessive sound transmission directly through the demising walls between certain adjacent spaces, sound transmission between third floor classrooms via the Q-deck roof, and via large, round supply air ducts serving the band and dance spaces. These issues, and how they were overcome, are discussed below.

2 Acoustic Separation Issues

2.1 Steel Framing Effects on Wall STC's

The structural steel framing (columns and cross bracing) supporting the building had to be accomodated within the cavities of the steel stud and gypsum board demising walls. While these demising walls do not support the building, the heights of the walls separating the various laboratories as well as the band, choral, and dance spaces, required that the studs be load-bearing (i.e., not lightweight). The acoustical separation was then compromised in two ways :

- Excessive transmission via the load-bearing studs,
- Resilient channel ineffective - "squashed" between steel framing and gypsum board.

[1†] clair.wakefield@rwdi.com

The wall cavities should have been wide enough to accommodate the steel framing elements, however, it was subsequently learned that the framing sizes had been increased late in the design process and it is possible that the wall cavity widths were not adjusted accordingly. The effect was to limit the Apparent Sound Transmission Class (ASTC) ratings achieved in field tests as follows:

- Wall between laboratories achieved ASTC 37 when lab-rated objective was STC 53,
- Wall between Theatre and Atrium achieved ASTC 37 when lab-rated objective was STC 60.

Figure 1 shows an example of the problems created in accommodating the steel cross braces within the Theatre/Atrium wall.



Figure 1: Bridging of Theatre/Atrium wall cavity by structural steel cross bracing.

2.2 Effect of Q-deck Roof on Acoustic Separation

When an insulated double-stud (steel studs and drywall) wall separating two classrooms on the third floor was field tested, it yielded STC 38, in spite of being laboratory rated at well over 50. Since this double stud wall contained no structural steel elements, it was at first unclear why it performed so poorly. Upon investigating more closely, it was found that sound was primarily being radiated from the Q-deck ceiling, which was the underside of the school's roof, and was concealed by a conventional T-bar acoustic ceiling. Unlike the floor assemblies between the first and second and second and third levels, the roof assembly has no concrete topping layer, only rigid insulation and roofing

material. Without the mass and stiffness provided by a concrete topping, the Q-deck was able to respond to sound field in the source room and provide a structure-borne sound path between the two rooms. Upon reviewing the structural plans, it was seen that, in only a few situations, were the Q-deck corrugations perpendicular to the demising wall, thereby facilitating the transmission of vibration/sound between rooms. Where the corrugations ran parallel to the wall, sound transmission was greatly reduced.

2.3 Sound Transmission via Supply Air Ducts

Figure 2 shows the supply air ducts serving the school's performing arts area. The ducts shown are located in the band room. They pass through the wall at the right and continue into the dance and drama studios beyond. This wall had a STC objective of 55, but when first tested it yielded only STC 45. Investigations showed that, while the supply air ducts were round and featured spiral-wound sheet metal, sound created in the band room was able to pass through the duct walls, travel along the duct and pass through the demising wall. There sound was transmitted through the duct walls again and into the dance studio. Music created in the dance studio did the same, but in the opposite direction.



Figure 2: Supply Air Ducts passing between Band Room and Dance Studio.

3 Noise Control Measures

3.1 Improving the Transmission Loss of Walls

The problems created in certain demising walls by the presence of structural steel elements and inadequate cavity width were overcome by creating separate, free-standing (self-supporting) barrier layers (steel studs and gypsum board or dense board) over one side of each wall. The newly created wall cavities were insulated. In some cases the existing gypsum board surface was first removed so that a wider cavity could be created, in other cases, where space was less critical, the existing gypsum board was retained. The most challenging situation was that between the Theatre and the Atrium, both because of the size of the wall (see Figure 3) and because of the high STC objective. For this reason, a localized mock-up section was first created and tested before the entire wall was treated.

These wall treatments generally increase classroom wall performance from about ASTC 37 to ASTC 47. In the case of the Theatre/Atrium wall, the ASTC was increased from 37 to 55. Testing a wall of this size is challenging, however, the wall now functions satisfactorily.



Figure 3: Atrium/Theatre wall under construction - Atrium side.

3.2 Treating Q-Deck Sound Transmission

The problem of sound transmission between some third floor classrooms via the Q-deck roof was addressed by back loading the acoustic ceiling tile with 13 mm gypsum board. While it may not have been necessary, the builder back loaded the tiles on both sides of the wall. The result was to increase the ATSC from 37 to 46.

3.3 Treatment of Supply Air Ducts

Several options were considered, including internal duct lining, external duct lagging and silencers. Rather it was decided to build a acoustically-lined gypsum board bulkhead around the two ducts within the band room only. This treatment may be seen in Figure 4. The outcome of this treatment was to increase the ASTC between the band room and dance studio from 45 to 51. However, the ASTC rating would likely have been higher but for the presence of HVAC noise in the dance studio.



Figure 3: Bulkhead Around Supply Air Ducts in Band Room.

ACOUSTICAL PROPERTIES OF HOMOGENEOUS PLANAR SURFACES IN ROOM ACOUSTICS: A THEORETICAL REVIEW

Behrooz Yousefzadeh [*1] **and Murray Hodgson** [†1]
[1]Department of Mechanical Engineering, University of British Columbia, 2054-6250 Applied Science Lane, Vancouver, BC

1 Introduction

This work reviews the fundamental boundary conditions used in room acoustics from a theoretical perspective. In particular, it is assumed that sound propagates in air as plane waves and the boundaries are planar walls. Moreover, the walls are assumed to be infinite in lateral extent and have uniform properties along their surface. These two conditions ensure that sound waves are reflected wholly specularly, as opposed to being scattered as well ; the assumption of infinite lateral extent avoids scattering from the edges of the walls, and the assumption of uniform properties avoids scattered radiation caused by the inhomogeneity of the wall. Parameters used in describing the acoustical properties of the walls are introduced and, using the wave equation, expressions are derived for them in terms of the wall impedance. Theoretical relations between the parameters are then discussed based on (i) physical restrictions such as energy conservation and the causality condition and (ii) simplifying assumptions used in room acoustics such as local reaction of surfaces and the minimum-phase condition. Finally, some practical aspects of these theoretical relations are discussed using simple examples.

2 Acoustical Descriptors of Room Surfaces

The effects of a surface on an incident acoustic wave are typically described by the following parameters : surface impedance (or admittance), reflection coefficient, transmission coefficient and absorption coefficient. These parameters are traditionally defined in the frequency domain and, depending on the context, are used in either frequency and time domains. The first two relate to sound pressure and are complex-valued quantities, while the second two relate to sound energy and are real-valued quantities.

Surface impedance, Z, is defined as the ratio of sound pressure, p, and the normal component of particle velocity, V_n, at the surface. In the frequency domain, the real part of Z is called resistance and its imaginary part is called reactance :

$$Z(\omega) = R(\omega) + iX(\omega) \tag{1}$$

In this work, we mainly focus on surface impedance. See [1] for the relations among Z and the other parameters.

3 Physical Restrictions on the Impedance Function

When working in the frequency domain, plane-wave reflection of harmonic sound from a uniform surface can be described using a complex number for each angle of incidence. However, for an incident wave with an extended frequency spectrum (e.g. an impulse), one needs to know the impedance as a function of frequency. This information can be used to translate the impedance into an equivalent function that can be used in the time domain. Time-domain analysis is used in auralization [2] and study of flow ducts [3], for example, and recently in low-frequency room acoustics as well [4].

To extend the definition of impedance, one needs to introduce certain conditions on the impedance function. These conditions ensure that the wall does not violate the conservation of energy, that physical parameters (pressure and velocity) remain real-valued quantities, and that the problem satisfies the principle of causality.

3.1 Energy conservation

Conservation of energy requires that the energy of the wave reflected by the wall be smaller than or equal to the energy of the incident wave. In other words, the power reflection coefficient should not exceed unity. This condition requires the surface resistance to be non-negative : $R(\omega) > 0$. A surface with this property absorbs sound energy, and is referred to as a passive surface. Note that if a sound source is present on the other side of the wall, and energy is transmitted through the wall, the calculations for conservation of energy are carried out differently.

3.2 Reality condition

To ensure that the impedance function remains a real-valued quantity, resistance and reactance, $R(\omega)$ and $X(\omega)$, should be even and odd functions of frequency, respectively. This property is often used to remove negative frequencies from the analysis.

3.3 Causality condition

The causality condition states that the pressure at the wall cannot depend on future values of velocity, and vice versa. For this condition to hold, $Z(\omega)$ should not have any poles or zeros in the upper complex half-space [3] ; i.e. $X(\omega) > 0$. If the negative Fourier sign convention is used, the upper half-space becomes the lower half-space. Note that violation of the causality condition brings about unphysical instabilities in time-domain analysis [3].

One can show that the real and imaginary parts of a causal impedance function are related by the Hilbert transform [5]. Therefore, knowledge of resistance $R(\omega)$ is sufficient to completely specify the system, and reactance $X(\omega)$ carries redundant information. The underlying assumption here is that the system is linear and causal.

∗. behroozy@alumni.ubc.ca
†. murray.hodgson@ubc.ca

4 Common simplifying assumptions

4.1 Locally-reacting walls

Local reaction of walls is perhaps the most widely used simplifying assumption used in room acoustics to describe the properties of surfaces. The impedance of a locally-reacting surface is independent of the direction of the incident sound. This is equivalent to assuming that the particle velocity generated by the incident sound at any point on the surface is related linearly to the local pressure only, and is therefore independent of the form of the incident sound field [1]. A surface that is not locally-reacting exhibits extended reaction. A special case of extended reaction is a surface with a reflection coefficient that does not depend on the angle of incident sound. Thus, the walls of an ideal anechoic chamber have extended reaction to the sound field inside the chamber.

Local reaction is encountered wherever the wall itself or the space behind it is unable to propagate waves in a direction parallel to its surface [1]. In room acoustics, walls are not locally-reacting at relatively low frequencies [2]. In particular, modelling multilayer surfaces as locally reacting can make audible mistakes in prediction of parameters that describe the subjective acoustical characteristics of rooms [6, 7]. In general, the assumption of local reaction gives acceptable results in the frequency range in which the sound field may be assumed to be diffuse. It also increases the range of applicability of the data collected from an impedance tube test.

4.2 Minimum-phase condition

An assumption that greatly simples the relation between the acoustical descriptors of surfaces is the minimum-phase condition. For a minimum-phase function, its magnitude and phase are related through a Hilbert transform [8]. An important implication here is that by measuring the real-valued absorption coefficient of a surface, one can obtain the complex-valued reflection coefficient, and thereby the impedance, of the surface. One of the early applications of the minimum-phase condition in room acoustics is auralization of echogram obtained from ray-tracing by Kuttruff [9].

Interestingly, a surface with a minimum-phase reflection coefficient has the property of reflecting the acoustic energy faster than a surface with the same magnitude of reflection coefficient but a non-minimum phase [8]. It is noted, however, that neither the reflection coefficient nor the transfer function of a room are necessarily minimum phase.

5 Concluding Remarks

We gave an overview of some of the theoretical aspects of modelling homogeneous planar surfaces in room acoustics. In particular, we focused on physical restrictions and common simplifying assumptions applied to the surface impedance function. These considerations are particularly important for auralization [2], time-domain simulations of sound fields [4] and improvements of existing prediction models (such as Miki's correction of the Delany-Bazley model [10]). Other aspects of modelling that warrant more research include (i) a detailed study of the minimum-phase condition in relation to the room transfer function and surface properties ; (ii) influence of large objects on the acoustics of a room ; (iii) sound fields in rooms coupled by a common wall. An aspect of modelling that has recently received attention is the effects of the finite size of the boundaries [11]. It is worth noting that many of these features are significant for low-frequency acoustical modelling of rooms (up to the Schröder frequency). We hope that this discussion ignites interest and further research on these topics.

References

[1] Heinrich Kuttruff. *Room acoustics*. Spon Press, USA, 2009.

[2] Michael Vorländer. *Auralization*. Springer, Germany, 2007.

[3] Sjoerd W Rienstra. Impedance models in time domain, including the extended Helmholtz resonator model. *AIAA paper*, 2686 :2006, 2006.

[4] Yiu Wai Lam and Jonathan Hargreaves. Time domain modelling of room acoustics. In *Acoustics 2012*, 2012.

[5] Yves H Berthelot. Surface acoustic impedance and causality. *The Journal of the Acoustical Society of America*, 109(4) :1736–1739, 2001.

[6] Behrooz Yousefzadeh and Murray Hodgson. Energy- and wave-based beam-tracing prediction of room-acoustical parameters using different boundary conditions. *The Journal of the Acoustical Society of America*, 132(3) :1450–1461, 2012.

[7] Murray Hodgson and Andrew Wareing. Comparisons of predicted steady-state levels in rooms with extended- and local-reaction bounding surfaces. *Journal of Sound and Vibration*, 309(1) :167–177, 2008.

[8] J Gregory McDaniel and Cory L Clarke. Interpretation and identification of minimum phase reflection coefficients. *The Journal of the Acoustical Society of America*, 110(6) :3003–3010, 2001.

[9] Heinrich Kuttruff. Auralization of impulse responses modeled on the basis of ray-tracing results. *Journal of the Audio Engineering Society*, 41(11) :876–880, 1993.

[10] Yasushi Miki. Acoustical properties of porous materials. modifications of delany-bazley models. *Journal of the Acoustical Society of Japan (E)*, 11(1) :19–24, 1990.

[11] Cheol-Ho Jeong. Improved low frequency room responses by considering finiteness of room boundary surfaces. *Acta Acustica united with Acustica*, 99(4) :545–556, 2013.

Dl2 Requirements For Speech Intelligibility And Privacy In Rooms

Denny Ng, Murray Hodgson

Sound pressure level decay per doubling of distance (DL2) is an attractive room acoustic parameter as it, by design, bears easily understandable spatial information. Since DL2 is dependant on local geometry in addition to all room acoustics parameters, it can be viewed as an all-encapsulating metric. The theoretical study discussed in this presentation shall serve as the framework for implementation of DL2 as a universal room acoustics parameter. Target values for DL2 for unamplified talkers were investigated in rooms of varying purposes and sizes (e.g. classrooms, dining establishments, offices, study spaces). These rooms were assumed to be classifiable under one of two use purposes: (1) speech intelligibility is desired throughout the space, (2) speech intelligibility is desired until a specific distance, and speech privacy is desired thereafter. Background noise levels (BNL) and reverberation times (RT) collected in sample spaces were averaged to find representative quantities. The BNL spectra were then adjusted to match A-weighted levels prescribed by ANSI S12.2 for corresponding spaces. As ISO 9921:2002 lacks speech intelligibility index (SII) values for various intelligibility ratings, ratings were translated from the speech transmission index (STI). In turn, the SII values were assigned at typical talker-listener distances depending on intelligibility or privacy objectives. Analysis will be completed over the next month, focusing on finding the frequency-dependant DL2 values required to achieve target SII values, and assessing feasibility based on experimental data.

GEOACOUSTIC INVERSION AND MEASUREMENT - INVERSION ET MESURES GÉOACOUSTIQUE

The Potential Of Remote Real-Time Noise Monitoring
Henk De Haan, Virgini Senden 106

Bayesian Inversion Of Time-Difference-Of-Arrival Data To Localize Bowhead Whales In The Chukchi Sea
Graham A. Warner, Stan E. Dosso, David E. Hannay 108

Geoacoustic Inversion Of Reflection Data In A Water Tank
Junjie (Jason) Shi, Stan Dosso, Dajun Sun 110

Advances In Acoustic Monitoring
Corey Kinart, Bill Gastmeier 112

The Evolution Of Environmental Ground Borne Vibration Measurement, Assessment And Reporting Over The Last 50 Years
Kathryn Katsiroumpas, Ian Matthew 114

Gradient Representations In Seabed Geoacoustic Inversion By Bernstein Polynomials
Jorge E. Quijano, Stan E. Dosso, Charles W. Holland, Jan Dettmer 116

THE POTENTIAL OF REAL-TIME NOISE MONITORING

Henk de Haan*[1] and Virgini Senden†[2]
dBA Noise Consultants Ltd, RR1 Site 14 Box 55, Okotoks, AB (403) 836 8806

1 Introduction

NoiseNet is the Canadian representative of Sensornet.nl. For over a decade, we have provided real-time web access to unmanned noise- and vibration monitors (including audio recordings) around airports, along railroads, highways and around industry. We helped clients make sense of Big Data by automatically analyzing data, developing source recognition and by providing customized reports. Real-time access to noise monitoring data results in increased confidence from the public, and drastically reduces response time and cost when dealing with complaints. Initially, these possibilities have been used to improve upon traditional noise monitoring, with identical goals for that kind of program. Recently however, unmanned noise measurements along light rail systems have been applied to assist in planned maintenance (rail re-surfacing, rolling stock maintenance) thus reducing maintenance cost.

2 Method

2.1 Traditional Monitoring

In traditional monitoring, data gathering, data analysis and reporting are separated in time. Disadvantages of this approach are the often fixed and limited duration of the monitoring, the impossibility to assess (the quality of) data while monitoring, the time-lapse between data acquisition and availability of analysis. Sometimes there's public distrust about the whole process. Which consultant has not been confronted with a remark along the lines of "It's not noisy now, you should have been here yesterday!"

2.2 Real-Time Monitoring

Real-Time Noise Monitoring (RTNM) allows access to gathered data (e.g. noise and vibration data, audio recordings, meteorological data, rail car speed) in real time by means of a dedicated website, with automated data analysis and reporting possibilities. This allows to evaluate and adjust the measurement program or the analysis while the program is active. Making certain webpages accessible to the public will build public confidence and is invaluable for complaint resolution.

A real-time monitoring solution consists of one or more sensors (e.g. a microphone and data acquisition devices (e.g. an SLM or RTA), a connection to the internet, a server with a database to gather and store the collected data, analysis software to extract information from the data, reporting tools and an interface to the client – a webpage.

* henk@dbanoise.com
† virgini@dbanoise.com

Custom solutions have been developed, e.g. for air force bases, where source recognition techniques were developed to distinguish between several types of helicopters (Chinook, Apache) and between helicopters and other sources (e.g. car passage). Other solutions were provided for construction projects in close proximity to 3rd party stakeholders, for communities exposed to high levels of multiple noise sources (road, rail and air traffic, shipping), around wind farms, race circuits and along rail lines.

3 Asset management using noise data

3.1 Rail Road Noise

RTNM projects typically start as compliance verification projects or in response to (potential) complaints, much the same as in traditional (noise) monitoring projects. Call it a better mousetrap. Recently, RTNM projects also proved useful for asset management purposes.

An important component of rail road noise has its root cause in the contact between wheel and the rail track. The smoother the track and the rounder the wheel, the less vibrations will result from their interaction. However, wear and tear will increase surface roughness of the track, and will cause the wheel to become less than perfectly round. Track can be resurfaced by grinding the rail, and wheels can be trued on a lathe. Ideally, this is done only when needed to minimize wear and tear, schedule interruptions and cost.

3.2 Rotterdam - The Hague Metropolis

One of the regional Light Rail Systems (LRS) in the Rotterdam – The Hague metropolis in the Netherlands is the Randstad rail, connecting Rotterdam, The Hague and Zoetermeer, a satellite city of The Hague. The area is densely populated, with residences in close proximity to the high-use tracks; noise is a potential issue.

However, noise could also provide insight in the condition of tracks and railcars, thus assisting in deciding when and where maintenance is required. A program was therefore designed to collect noise and vibration levels, train speed, weather data and track data (temperature, moisture) in one second intervals at four different locations. Also, the identification of every passing railcar is collected. All this results in approximately 9 billion data points a year; far too many for any human to extract useful information from. Dedicated software was developed to extract information from this mountain of data. An advantage of the software is

that analysis and reporting can be adapted as a result of learnings extracted from the information. Refer toFigure 1, displaying the vibration and sound levels before and after resurfacing the rails.

Figure 1 Effects of Resurfacing on Sound and Vibration Levels

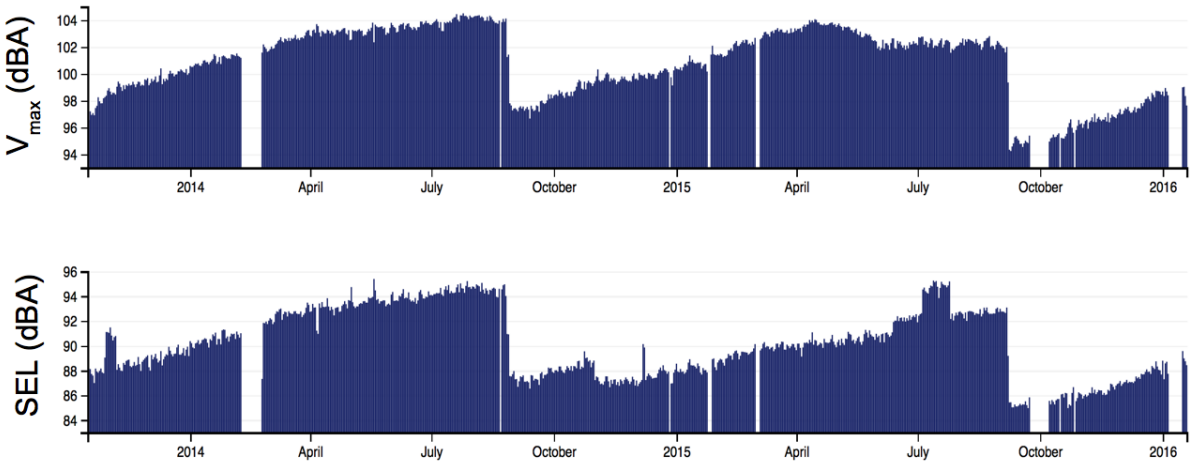


Figure 1 not only reveals the reduction in sound and vibration levels achieved as a result of resurfacing, but also provides a useful indication for when to resurface. This allows for planning the procedure to be conducted at the most optimal moment, considering both maintenance and schedule requirements.

Utilizing the available data, it is also possible to answer other questions that might arise, such as the average speed, the number of railcars per weekday or per hour, the average speed – the sky really is the limit. This offers quite a few possibilities for improvement. An example was combining location (GPS) data with railcar speed. A glance on a map displaying this information revealed that the trams frequently had to stop for traffic lights and were not enjoying the promised "green wave", allowing them to keep their schedule. The map was essential in moving the discussion forward; see Figure 2.

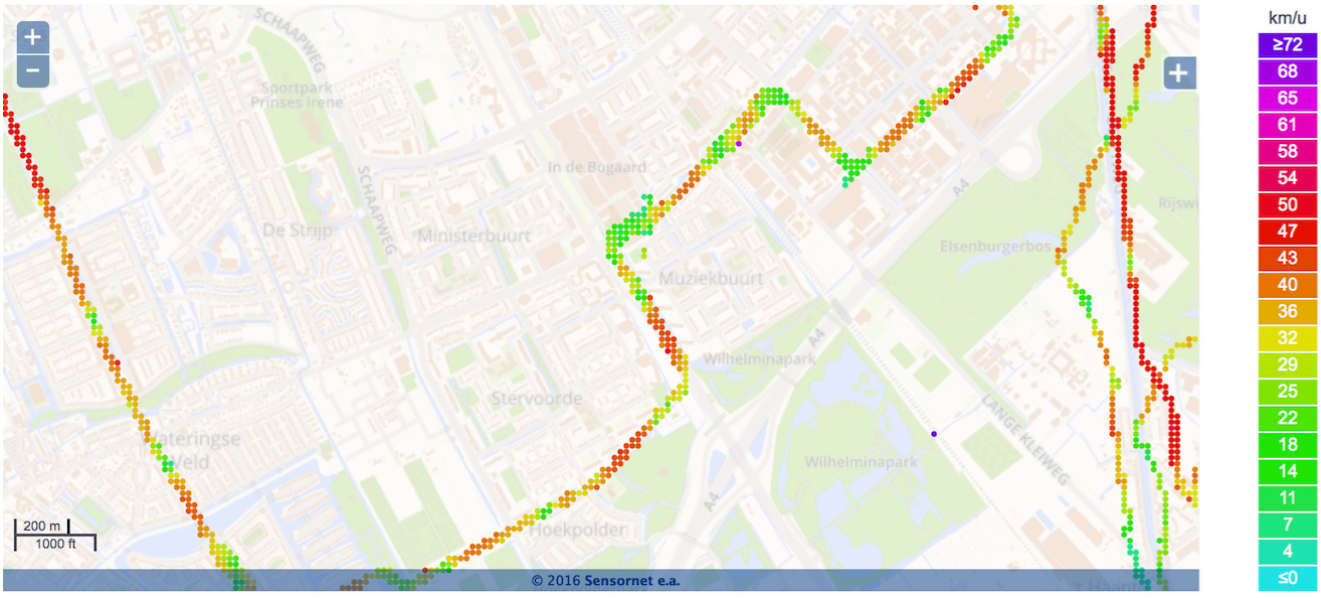
4 Conclusion

RTNM opens up fascinating new possibilities and opportunities by combining sensors, internet and Big Data analysis instruments. This include optimizing asset management, reducing maintenance cost and reducing the noise impact for residences in close proximity. It also increases public confidence in our work and provides a solid basis for discussions between stakeholders.

Remerciements/Acknowledgments

We want to than mr. Ron Maas from Sensornet.nl for allowing the use of their information.

Figure 2 Average Speed (km/hr)



BAYESIAN INVERSION OF TIME-DIFFERENCE-OF-ARRIVAL DATA TO LOCALIZE BOWHEAD WHALES IN THE CHUKCHI SEA

Graham A. Warner[*1,2], **Stan E. Dosso**[†2], **and David E. Hannay**[‡1]
[1]JASCO Applied Sciences, 2305-4464 Markham St, Victoria BC V8Z 7X8
[2]School of Earth and Ocean Sciences, University of Victoria, 3800 Finnerty Rd, Victoria BC V8P 5C2

1 Introduction

Localizing marine mammals without disrupting their behaviour is often performed using passive acoustic monitoring techniques. Passive localization over large areas requires widely-spaced acoustic recorders, which often precludes inter-recorder clock synchronization and applications of well-established matched-field or time-difference-of-arrival (TDOA) localization techniques. In this paper, we localize bowhead whales in the northeastern Chukchi Sea using a nonlinear Bayesian inversion of whale-call TDOA data obtained from non-synchronized recorders. The inversion is applied to batches of TDOA data in which the whale locations, relative recorder clock drifts, effective environmental sound speed, and recorder locations are all treated as unknown parameters (subject to prior information) and estimated in the inversion. Rigorous posterior uncertainty estimates for the parameters are provided by the Bayesian formulation. A more complete treatment of this work can be found in [1].

2 Bowhead-Whale Call Data

JASCO Applied Sciences recorded thousands of low-frequency bowhead-whale calls in the Chukchi Sea on a number of Autonomous Multichannel Acoustic Recorders (AMARs) during August–October, 2013 [2]. Although the AMAR internal clocks were synchronized at deployment, the clocks drift out of synchronization relative to each (a common problem for long-term instrument deployments). This paper considers whale calls recorded on up to seven AMARs (herein labelled A–G) over a 30 minute period on 11 October. The recordings were initially aligned roughly by observing similar time-frequency characteristics of transient sounds (e.g., seismic pulses or specific bowhead-whale calls). Whale calls were then manually associated between recorders by listening to the recording and visually comparing call spectrograms. Each call annotation defined lower and upper frequencies and start and end call times. TDOA data were then derived according to Sec. 3.1.

3 Methods

3.1 Data Processing

For each bowhead-whale call, recordings were band-pass filtered according to the frequency annotations and cross-correlated with all other recordings of the same call. TDOA data were determined from the maximum of the cross-correlation functions, providing time delays relative to an arbitrarily-assigned reference recorder. In this manner, each whale call yielded up to 21 TDOAs, which were ranked in descending order according to the the maximum value of the cross-correlation function (a measure of signal similarity). A minimum of 2 and up to 6 linearly-independent TODAs were selected for the inversion (linearly-dependent TDOAs were discarded for consistency with the assumption of independent data errors in Sec. 3.3).

3.2 Bayesian Inversion

Whale-call TDOA data depend on the whale and recorder locations, relative recorder clock drifts, and the environmental sound speed. Using a straight-path, time-of-flight propagation model, the TDOA for call w between recorders i and j is

$$t_{wi} - t_{wj} = \frac{\sqrt{(x_w - X_i)^2 + (y_w - Y_i)^2}}{c} - \frac{\sqrt{(x_w - X_j)^2 + (y_w - Y_j)^2}}{c} + \Delta_i - \Delta_j, \quad (1)$$

where x_w and y_w are the easting and northing coordinates of the whale, X_i and Y_i are the coordinates of the ith recorder, c is the effective waveguide sound speed (discussed below), and Δ_i and Δ_j are the recorder clock drifts relative to the reference recorder (this formulation assumes the relative clock drifts remain constant between the first and last whale calls in the data batch). Since the bowhead-whale calls propagate as normal modes in the shallow-water environment ($\sim$40 m depth), the sound speed c is an effective average modal group speed (less than the water sound speed) weighted by the frequency and modal content of the calls. The recorder locations are also considered unknown parameters in the inversion, subject to Gaussian priors centred at the deployment locations with 50 m standard deviations to account for uncertainties in GPS ship locations and lateral drift of the AMARS as they are lowered through the water. This is done so the whale-location uncertainties account for recorder-location uncertainties; high-precision recorder localization is not expected. In the Bayesian formulation the solution consists of properties of the posterior probability density (PPD) of the unknown parameters given the measured data and prior information. The PPD is approximated numerically for non-linear problems using a Markov-chain Monte Carlo algorithm [3].

3.3 Likelihood

Residual errors between measured and predicted data are assumed to be uncorrelated and Gaussian-distributed within a batch of whale calls, with an unknown standard deviation σ

[*]graham.warner@jasco.com
[†]sdosso@uvic
[‡]david.hannay@jasco.com

treated as a nuisance parameter in the inversion. For N data the corresponding likelihood function is

$$L(\mathbf{m}) = \frac{1}{(2\pi)^{N/2}\sigma^N} \exp\left[-\frac{|\mathbf{d}-\mathbf{d}(\mathbf{m})|^2}{2\sigma^2}\right], \qquad (2)$$

where $\mathbf{m}$ is the vector of unknown model parameters (whale and recorder locations, relative recorder clock drifts, and sound speed), and $\mathbf{d}$ and $\mathbf{d}(\mathbf{m})$ are measured and predicted data, respectively. Setting $\partial L/\partial\sigma = 0$ leads to a maximum-likelihood estimate for the error standard deviation σ:

$$\hat{\sigma}(\mathbf{m}) = \left[|\mathbf{d}-\mathbf{d}(\mathbf{m})|^2/N\right]^{1/2}. \qquad (3)$$

Equation (3) allows implicit sampling of $\hat{\sigma}$ by sampling explicitly over the parameters in $\mathbf{m}$.

4 Inversion Results

Figure 1 shows examples of the marginal probability densities for bowhead-whale locations in the horizontal plane computed for 12 whale calls via the Bayesian inversion algorithm. This figure illustrates the variety of uncertainty distributions obtained, which are strongly dependent on the source-recorder geometry for the specific case in each panel (note that not all recorders detected all calls). Localization uncertainties generally increase with the whale's distance from the centre of the AMAR cluster. When the off-axis recorders do not detect a call, the probability densities are symmetric about the cluster axis, as shown in Fig. 1(d), (j), and (l). Further, the marginal densities in Fig. 1(d) and (j) are multi-modal, and the marginal in Fig. 1(j) is notably curved (non-Gaussian). These three marginal PPDs illustrate the potentially strong non-linearity of the localization problem, and could not be recovered by a linearized inversion algorithm. Resolving such probability densities with the Bayesian sampling algorithm required parallel tempering [4] to provide sufficiently wide, but efficient, sampling. Call locations oriented broadside to the axis of the AMAR array are generally well localized, with the highest precision for calls that are also relatively close to the array (e.g., Fig. 1(a) and (k)). Endfire call locations are poorly constrained in terms of range to the AMAR cluster; however, the direction of arrival is well determined (e.g., Fig. 1(b) and (h)).

In terms of the other parameters included in the inversion, the effective sound speed was estimated at 1403 ± 18 m/s (one standard-deviation uncertainty), which corresponds approximately to the average group speed of the first two modes computed for the shallow-water environment at the whale-call frequencies. Relative clock drifts were estimated to uncertainties of generally less that 50 ms, and an essentially linear clock-drift trend over a two-week period was determined by considering multiple TODA data sets (not shown here). The posterior localization uncertainties for the recorders were similar to their prior uncertainties indicating the inversion did not significantly improve these locations; however, this procedure accounts for the recorder-location uncertainties in the whale-location uncertainty estimates. The estimated data standard deviations varied from 0.10–0.14 s, indicating the TDOA data are well fit by the inversion.

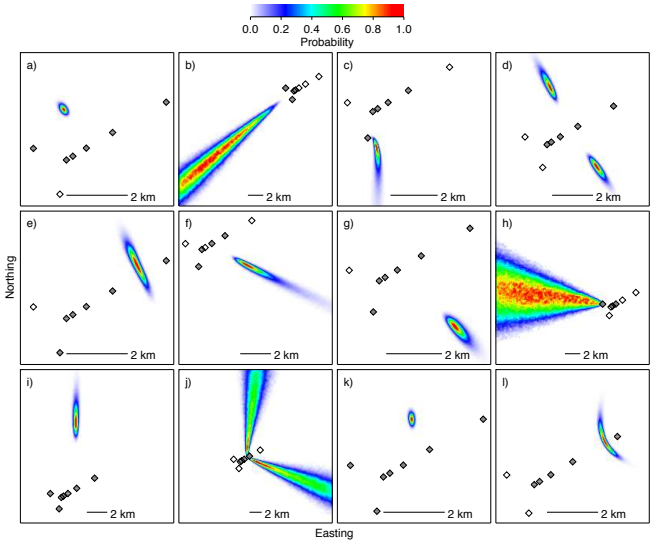


Figure 1: Posterior marginal probability densities for bowhead whale locations for 12 selected calls. Prior recorder (AMAR) locations (GPS deployment positions) are shown with diamond symbols; filled diamonds indicate recorders which detected the call considered in that panel, open diamonds represent recorders which did not detect the call.

5 Summary

This paper presented a nonlinear Bayesian inversion to passively localize bowhead whales in the Chukchi sea based on TDOA data from an array of autonomous, non-synchronized recorders. The inversion rigorously quantifies localization uncertainties, accounting for unknown recorder clock drifts and for uncertainties in the TODA data, recorder locations, and effective environmental sound speed. Localization uncertainties are found to be strongly dependent on the particular source-recorder geometries with strong nonlinear effects.

Acknowledgments

The authors thank Shell Exploration and Production Company for permission to publish this analysis of their data.

References

[1] G. A. Warner. Bowhead whale localization and environmental characterization in the Chukchi Sea using nonlinear Bayesian inversion. *PhD Thesis*, University of Victoria, 2016.

[2] J. Delarue, J. MacDonnell, B. Martin, X. Mouy, and D. Hannay. Northeastern Chukchi Sea joint acoustic monitoring program 2012–2013. Technical Report 00808, JASCO Applied Sciences, 2014.

[3] G. A. Warner, S. E. Dosso, J. Dettmer, and D. E. Hannay. Bayesian environmental inversion of airgun modal dispersion using a single hydrophone in the Chukchi Sea. *J. Acoust. Soc. Am.*, 137:3009–3023, 2015.

[4] S. E. Dosso, C. W. Holland, and M. Sambridge. Parallel tempering in strongly nonlinear geoacoustic inversion. *J. Acoust. Soc. Am.*, 132:3030–3040, 2012.

GEOACOUSTIC INVERSION OF REFLECTION DATA IN A WATER TANK

Junjie Shi [*1,2,3], Stan E. Dosso [†3] Dajun Sun [‡1,2]
[1] Key Laboratory of Underwater Acoustic Technology, Harbin Engineering University, China.
[2] College of Underwater Acoustic Engineering, Harbin Engineering University, China
[3] School of Earth and Ocean Sciences, University of Victoria, Canada.

1 Introduction

Geoacoustic inversion surveys conducted in ocean environments often require expensive resources and challenging logistics and sometimes involve computationally intensive calculations. For example, moored or towed horizontal arrays or vertical arrays of hydrophones are often employed and complicated setups can introduce operational difficulties and increase costs in shallow-water inversion methods such as matched-field inversion, reverberation inversion, transmission-loss inversion, etc. However, inversion of reflectivity data can be carried out by employing a single receiver, and holds considerable promise owing to more convenient instrument deployments and recoveries and less time-consumption for forward-model computations as full-wave calculations may not be required.

This paper describes a nonlinear Bayesian inversion approach applied to broadband reflection data [1] at multiple frequencies for geoacoustic properties of river sand at the base of an anechoic water tank at Harbin Engineering University, China.

2 Bayesian Inversion Theory and Algorithms

2.1 Bayesian formulation

The nonlinear approach to invert reflectivity data for geoacoustic properties is based on Bayes rule, which can be expressed as [1]

$$P(\mathbf{m}|\mathbf{d},F)=\frac{P(\mathbf{d}|\mathbf{m},F)P(\mathbf{m}\,|\,F)}{P(\mathbf{d}\,|\,F)}=\frac{P(\mathbf{d}|\mathbf{m})P(\mathbf{m})}{P(\mathbf{d})}\ . \qquad (1)$$

The goal of Bayesian inversion is to interpret the multi-dimensional posterior probability density (PPD), $P(\mathbf{m}|\mathbf{d},F)$, for optimal estimates of the model parameters $\mathbf{m}$ and their uncertainties based on observed data $\mathbf{d}$ and prior information $P(\mathbf{m})$; here F represents the choice of model parameterization (e.g., the number of sediment layers). These properties include the maximum *a posteriori* (MAP) model and marginal probability distributions etc, which are defined, respectively, as

$$\hat{\mathbf{m}}=\mathrm{Arg}_{\max}\left\{P(\mathbf{m}|\mathbf{d})\right\} \qquad (2)$$

$$P(m_i\,|\,\mathbf{d})=\int\delta(m_i-m_i')P(\mathbf{m}|\mathbf{d})d\mathbf{m}'\ . \qquad (3)$$

Joint (two dimensional) marginal densitites can be computed analogously to Eq. (3). Parameter uncertainties are also quantified in terms of credibility intervals. For example, the highest probability density credibility interval represents the interval of minimum width containing of the area of the marginal distribution (50% and 95% credibility intervals are used in this paper).

2.2 Numerical Optimization

In this paper, delayed rejection adaptive Metropolis (DRAM), a Markov Chain Monte Carlo (MCMC) sampling method, is applied to nonlinear Bayesian inversion of reflectivity data. It can provide an effective numerical approach to determine MAP estimates and to evaluate integrals Eq. (2)-Eq. (3) to quantify parameter estimates, uncertainties and inter-relationships [2].

DRAM is a combination of two ideas for improving the efficiency of Metropolis-Hastings type MCMC algorithms, i.e. adaptive Metropolis (AM) and delayed rejection (DR). In adaptive Metropolis, the covariance matrix of a Gaussian proposal distribution is adapted on the history of the Markov chain. In delayed rejection, upon rejecting a proposed candidate model, a second (smaller) move is proposed instead of simply rejecting the initial move. Moreover, the acceptance probability of the second candidate move is formulated so as to preserve reversibility of the Markov chain. Once convergent MCMC chains are established, the MAP parameter estimates can be determined by Eq. (2), while parameter uncertainties and inter-relationships will be further analyzed according to Eq. (3).

2.3 Model Choice

Choosing an appropriate model parameterization F is an important aspect for Bayesian reflectivity inversion. For parameterized models, measuring the probability of observed data arising under the assumed model is an effective way to estimate the most appropriate model parameterization. The Bayesian information criterion (BIC) is applied here to approximate Bayesian evidence in terms of minimizing

$$\mathrm{BIC}=-2\log P(\mathbf{d}|\mathbf{m}^*,F)+M\log N\ . \qquad (4)$$

3 Inversion Results and Analysis

3.1 Model Selection Results

MAP models were estimated by DRAM for model parameterization including 1, 2, and 3 layers (including a halfspace). The results of the model choice are shown in Figure 1. Figure 1(a) shows the misfit function (negative log likelihood) decreases rapidly with layers changing from 1 to

[*] junejieshi@outlook.com
[†] sdosso@uvic.ca
[‡] sundajun@hrbeu.edu.cn

2, while decreases much less in going from 2 to 3 layers. Considering the penalty term for the number of parameters for every possible model shown in Figure 1 (b), Figure 1 (c) indicates BIC has its minimum at 2 layers, showing this is the maximum number of layers that can be reliably estimated from the observed data.

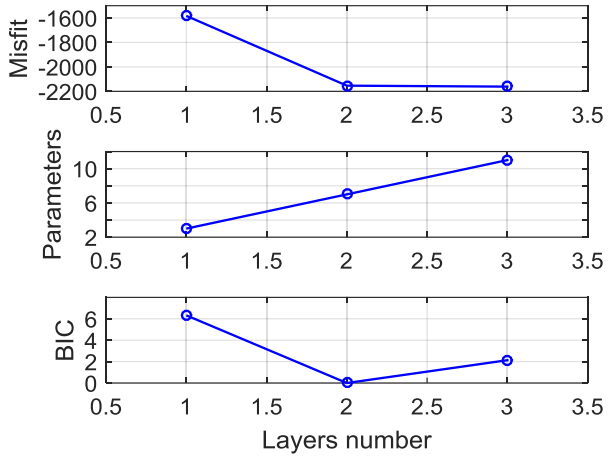


Figure 1: Results of the model choice in terms of: (a) misfits, (b) number of parameters, (c) BIC as a function of layers numbers with its value shifted to unity at its minimum and shown in log scale for display purposes.

3.2 Inversion Results

Figure 2 shows good agreement between the observed data and data predicted for the inversion results at 31.5 kHz, with 50% and 95% credibility intervals quantified. Figure 3 shows joint marginal probability distributions of parameters of interest for the two-layer model to illustrate parameter uncertainties and inter-parameter relationships.

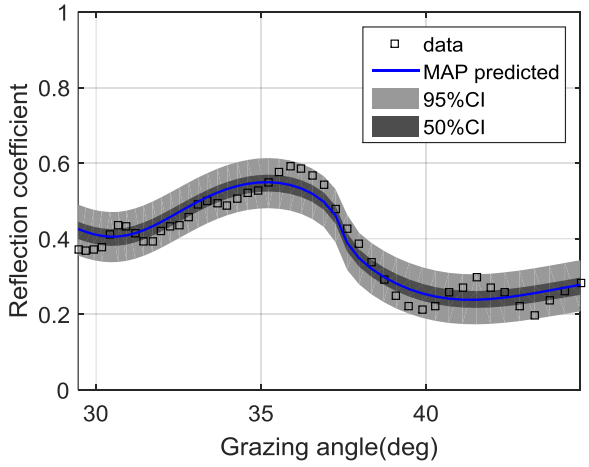


Figure 2: Measured reflection coefficient data (squares) and predicted data for the MAP model (line) and 50% and 95% credibility intervals from the inversion at 31.5 kHz.

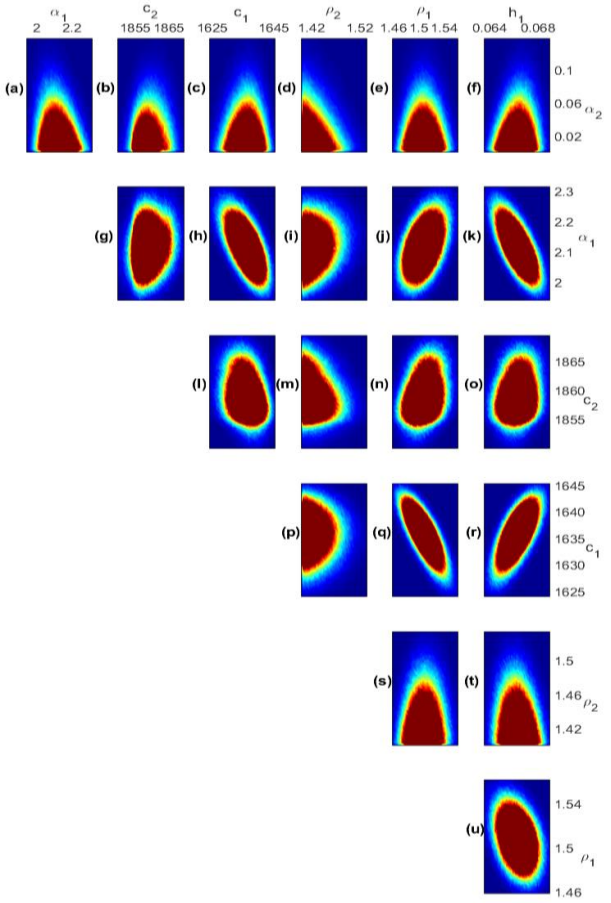


Figure 3: Joint marginal probability distributions of parameters of interest

4 Summary and Discussion

This paper presented a nonlinear Bayesian inversion for reflection coefficient data to estimate geoacoustic properties and uncertainties of sand layers at the base of a water tank. The MAP inversion results for the first layer suggested that this layer is clayey sand which is consistent with a visual observation.

References

[1] Dosso, S. E. and Holland, C. W. Geoacoustic uncertainties from viscoelastic inversion of seabed reflection data. IEEE Journal of Oceanic Engineering, 31(3): 657-671, 2006.

[2] Laine, M. Adaptive MCMC methods with applications in environmental and geophysical models. Finnish Meteorological Institute, 69: 1–46, 2008.

[3]Jensen, F. B., Kuperman, W. A., Porter, M. B. and Schmidt H. Computational Ocean Acoustics. Springer-Verlag: New York, USA, 2000.

ADVANCES IN ACOUSTIC MONITORING

Corey D. Kinart, MBA, PEng[*] and William J. Gastmeier, MASc, PEng[†]
HGC Engineering, 2000 Argentia Road, Plaza 1, Suite 203, Mississauga, Ontario, Canada L5N 1P7

1. INTRODUCTION

Advances in the technology of digital sound level analyzers have made automated acoustical monitoring an even more powerful tool for assessing and investigating environmental noise. In particular, advances in digital memory allow vast amounts of high resolution data to be gathered, which can be very useful in the context of temporal and spectral analysis. Perhaps the most significant aspect of increased memory capacity in acoustical monitoring equipment is the option to collect continuous audio recordings, remotely, for days to weeks at a time. Not only do audio recordings allow us to "listen in" on monitoring periods of interest during post analysis, they turn acoustic monitoring equipment into silent sentinels that never need to eat or sleep. In some cases, practitioners no longer need to remain as continuous observers in the field (both figuratively and literally); coupled with advances in deployable solar power systems and cellular modems, acoustical data and audio recordings can be collected remotely for weeks, months or even longer without the need for a technician or acoustician to be present. However, the old engineering adage of "too much data can be a bad thing" is an important consideration when it comes to advanced acoustic monitoring. Not only can gathering vast quantities of data present a challenge to analyze and derive meaningful insight from, it can also become a formidable challenge to maintain and protect the data itself.

This paper explores the evolving uses of acoustic monitoring, as well as advances in monitoring equipment and new capabilities afforded therefrom. Also discussed are some of the challenges related to conducting long term outdoor acoustic monitoring with regard to deployment and maintenance logistics, concurrent data collection (e.g. meteorological and/or operating parameters of a facility of or specific equipment of interest) and post analysis, including lessons learned from field experience.

2. EVOLVING USES

Monitoring of environmental noise is most certainly not a new phenomenon – we have been monitoring noise from airports, large industries, major transportation thoroughfares and the like for decades. However, such monitoring systems have, in past, been typically permanent stations equipped with continuous grid power and hard-wired data transfer systems, all wrapped up in bulky housings to protect the devices within from the elements. Such systems are quite expensive, and offer very little operational flexibility, particularly with regard to mobility.

The latest in acoustic monitoring technology no longer consists of a permanent fixture; smaller, more advanced and robust systems can be deployed nearly anywhere and for any length of time. Transfer of measurement data, via either a hard internet connection or cellular modem, allows for virtually real-time data acquisition. Behind us are the days where one must wait until the end of a deployment period to begin analyzing the recorded data or, in some cases, find out that some component of the deployed system had suffered a failure during the deployment period. Moreover, autonomous monitoring systems require significantly less resources to maintain; one needn't visit the equipment on a regular basis to refresh batteries or offload saved data to relieve what, in past, has been very limited storage capacity. Such advancements offer significant efficiencies for long-term monitor deployment, such as that routinely undertaken for assessment of wind turbine noise. Such monitoring campaigns can span many months in order to capture sound levels both with and without the wind turbines operating, under a prescribed variety of meteorological conditions.

Noise complaint investigations, particularly when the sound(s) of concern are intermittent, are dramatically more sophisticated when monitoring equipment can be deployed at or moved amongst multiple locations within a given community for any length of time. Audio recordings, either continuous or for short durations triggered by a measured sound level exceeding a specified threshold, allows investigators to listen to a given period or event, and focus on the sound(s) of concern, separate from other background sound. An acoustical specialist needn't be present in the field for long periods in order to "catch" the sound(s) of concern.

Construction vibration monitoring is becoming a more common occurrence in highly urbanized areas where there is increased potential for damage to structures adjacent to a demolition or construction site. Continuous vibration monitors equipped with real-time communication capability can issue a report the moment that vibration levels exceed a threshold of concern; construction activities can then be halted and the approach adjusted to ensure protection of surrounding structures.

The following sections discuss the primary components of an acoustic monitoring package, along with the associated strengths and weaknesses of each.

3. PRIMARY COMPONENTS

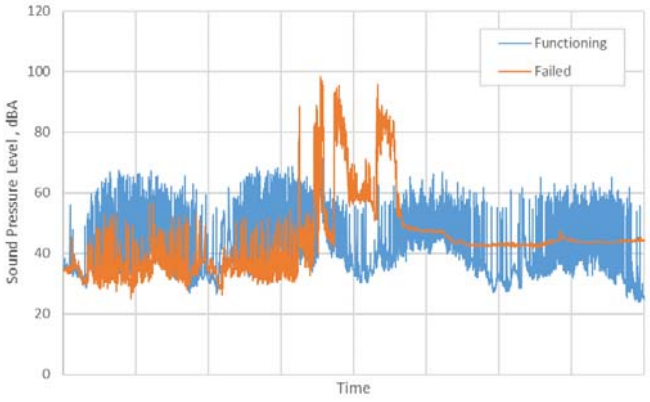
3.1 Measurement Device

The central component of any monitoring system is the measurement device itself, be it for sound, vibration or both. When equipped with a means of remote communication (discussed further in a following subsection), the latest measurement devices can be configured and re-configured remotely, which can reduce occurrences of costly configuration errors (that may not be discovered until the end of a deployment) and offers greater adaptability. As noted above, some measurement devices can be configured to trigger recordings of a pre-determined dataset when a sound or vibration threshold is exceeded, which reduces the total amount of data stored in local memory, thereby extending the deployment period in cases where the data is not being downloaded remotely. Another significant feature

[*] ckinart@hgcengineering.com
[†] bgastmeier@hgcengineering.com

of today's most advanced monitoring devices is the ability to self-test the integrity of the microphone on a regular basis to ensure that the device is functioning correctly. Such a feature can help to avoid costly losses of key datasets and/or monitoring time. Figure 1 provides an example of an in situ failure of a deployed sound level monitor.

Fig. 1. Example of sound levels recorded by two nearby monitors where one device failed in situ, as evidenced by the "flat lining" of one dataset.



3.2 Power

Advances in portable solar power systems have made it possible to deploy acoustic monitors for extended periods in very remote areas (where grid power is not available), without the need for regular maintenance visits by a technician in order to refresh batteries. This represents a significant boost in efficiency of time and resources for a given acoustic monitoring campaign. In the event of a temporary power interruption (resulting from a solar power system that becomes temporarily covered by snow, for example), many advanced measurement devices will simply restart when the power supply becomes available again, and resume monitoring immediately.

3.3 Memory

The built-in (or replaceable) memory capacity of the latest acoustical monitoring devices has increased significantly in recent years, with capacities of 32 GB or more being common. Such significant memory capacity means that equally significant amounts of data can be collected and stored, including triggered audio recordings, and limited periods of continuous audio. Lower quality audio recordings (which consume far less memory) can be quite sufficient for listening to identify sounds during periods of interest; audio recordings of sufficient quality to be post-processed can consume significant amounts of memory over relatively short periods.

3.4 Data Transfer

Whether via a hard-wired or Wi-Fi connection to the internet, or by cellular modem, being able to transfer data at will from a remotely deployed acoustic monitoring system allows data to be analyzed sooner (rather than having to wait until a technician retrieves the data from the equipment in the field), and facilitates ensuring that the monitor is still functioning as expected. If a monitor is deployed for a period of several weeks and experiences a hardware failure early in the deployment, being able to detect and resolve such an issue early is a significant strength of recent, downloadable monitoring systems (rather than finding out upon retrieving the equipment after a long-term deployment that some component failed two days into the campaign). With a hard-wired connection to the internet, data can be downloaded from monitoring equipment essentially in real-time.

4. LIMITATIONS

Although significant advances have been made in the complexity and flexibility of acoustic monitoring equipment, there remain several notable limitations with regard to power, memory and data transfer. With regard to power, fully autonomous solar charging systems must be deployed in areas with sufficient solar exposure, which can be limited due to foliage, snowfall, or even limited sunlight due to prolonged periods of heavy cloud cover or short days in extreme northern/southern locales. In addition, the size of solar panels and weight of accompanying batteries can be a challenge to deploy and retrieve in very remote areas. Finally, large solar panels can make the monitoring equipment quite visible, increasing the risk of being vandalized or stolen.

Despite the obvious benefits afforded by cellular modems, they consume significant amounts of power (relative to the power needs of most measurement devices), and can thus strain a system that relies on batteries alone, or limited solar systems. While this can be mitigated by scheduling "offline" time to conserve energy (the modem automatically goes online at scheduled intervals to report equipment integrity, upload data, etc.), this can nevertheless be a significant design consideration when developing a fully portable system capable of long-term deployment. In addition, cellular modems introduce a potential failure point in an acoustic monitoring system that is secondary to the measurement device(s) themselves, and can sometimes consume valuable troubleshooting time. Finally, depending on the data transfer rate of a given data transfer media, the rate at which data is collected may exceed the transfer speed, such that the internal memory will reach capacity at some point, limiting the deployment life of the system.

5. LESSONS FROM FIELD EXPERIENCE

Indeed, too much data can be a bad thing. A noise complaint investigation involving deployment of six acoustical monitors collecting sound levels and continuous audio recordings for a period of approximately three months generates a tremendous amount of data, both in terms of digital memory, and for post-processing. Synchronized with concurrent meteorological data and varying operating conditions of an industrial site the size of a small town, deriving meaningful conclusions from such a volume of information can be a significant challenge. Moreover, the maintenance and backup of several terabytes of data for just a single project can be a daunting task.

The application of advanced acoustical monitoring must therefore be carefully thought out in order to ensure that an appropriate amount of data is collected in the context of a given challenge. When advanced monitoring assets are readily available and easily configured to collect massive sums of data, it can be all too easy to bring to bear far more resources than are necessary in order to complete a given task. In that regard, all of the efficiencies afforded by advances in monitoring technology can be offset by the time spent administering and deriving meaning out of the tremendous amount of data that can be readily collected.

THE EVOLUTION OF ENVIRONMENTAL GROUND BORNE VIBRATION MEASUREMENT, ASSESSMENT AND REPORTING OVER THE LAST 50 YEARS

Kathryn Katsiroumpas, B.Sc.E., P.Eng. [*1] and Ian Matthew, M.Sc., P.Eng. [†1]
[1]Valcoustics Canada Ltd., 30 Wertheim Court, Unit 25, Richmond Hill, L4B 4B9

1 Introduction

Urbanization has increased the proximity of residential development to sources of environmental ground borne vibration, including at and below grade rail, construction activities, quarries and heavy industrial processes. Measurement, assessment and reporting on ground borne vibration induced by these sources have always been an integral component of acoustical consulting practice. This retrospective highlights the evolution of measurement equipment and reporting techniques over the last 50 years. Often, changes to the methodology are driven by new or modified regulatory guidelines as opposed to advances in available technology. Other factors, such as capital costs and efficiency are also considered. Limitations in current measurement and reporting methodologies give an indication of where improvements in technology may be made to environmental ground borne vibration assessment in the future.

2 Discussion

2.1 1950s to early 1980s

In the early 1950s, the issue of environmental vibration relative to perception and comfort were not of particular concern. Most monitoring focussed on activities that generated vibration capable of structural damage such as blasting at quarries. A good example of this is a 30 year long monitoring program at a quarry in Ontario of weekly blasting activities. There were few regulations at that time and the project was initiated by the quarry owners in case of future complaints. The results of these measurements were analysed and compiled in reports issued yearly.

These measurements were done with a portable seismograph, the Cambridge vibrograph, shown in Figure 1. The Cambridge vibrograph etched directly transmitted vertical movements on celluloid tape with a stylus. The celluloid drum was driven by an internal wind up clockwork. There was no need for a power source, which for a portable unit, was beneficial. The amplitude of the recorded movements were reduced by 10 to 50 times and could only be seen using a simple microscope. This made checking results in real time impossible. A timing signal and an electrically operated event marker would also be noted on the celluloid alongside the deflection trace. The vibrograph was capable of measuring 10 to 100 Hz but could be extended down to 1 Hz.

Given that the movements were directly measured, the vibrograph itself had to be set up at the measurement point which would have been a concrete monument. Since remote measurement was not possible, there was an inherent risk to those operating the equipment, particularly when measuring quarrying activities in close proximity.

The measurement, analysis and reporting procedures were all done by hand and were time consuming. Reports were book bound and either hand typed or sent out to be published.

Surprisingly, the methodology for these measurements remained largely unchanged over the 30 year period. A second less sensitive Leet seismograph was added in order to measure two locations in 1956. It should be noted that even in the 1950s, the Cambridge vibrograph was not cutting edge technology. Nonetheless, it was robust and functional and there was no impetus to invest additional funds in new equipment.

It wasn't until 1974 that the Cambridge vibrograph was replaced with another seismograph, the VME Velocity Recorder. The VME had an external geophone and using a mirror galvanometer, recorded results on photo sensitive paper. The drawbacks to this seismograph were the limitations (and weight) of the internal rechargeable battery, the expense of the photo sensitive paper and the short measurement duration of 60 seconds.

The early 1970s also saw the beginning of the use of the metric system, although the earliest reports referred to metric as a fashion and not yet a standard. Reports through to the early 1980s were still hand typed and most results tabulated in the same fashion as the previous 20 years.

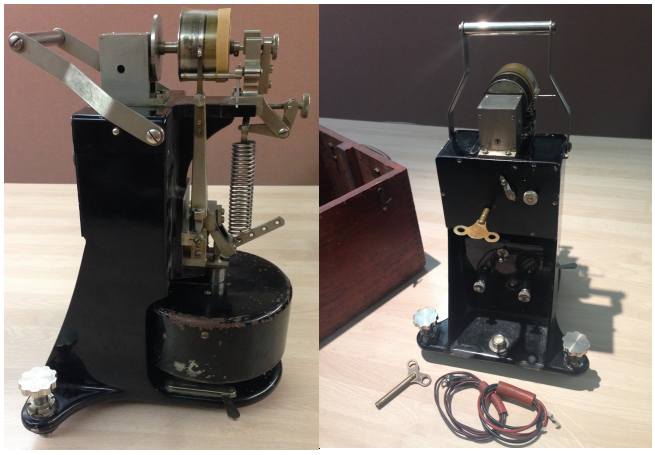


Figure 1: Cambridge Vibrograph

[*]kathy@valcoustics.com
[†]ian@valcoustics.com

2.2 Early 1980s to mid 2000s

In the early 1980s, railway authorities - Canadian Pacific (CPR) and Canadian National (CNR) - started to develop their own land use guidelines for new development adjacent to railways. These included criteria for railway induced ground borne vibration. A boom in new residential construction meant that railway vibration studies became a larger part of acoustical consulting.

The older seismographs were not particularly useful in this application. Thus, new equipment was acquired in order to adequately measure and assess vibration from railway train passbys. The methodology of choice was to record the signal from vibration velocity transducers. Initially this was done on analog devices, a Nagra IV-SJ and more frequently, a Sony Beta tape recorder. This transitioned to a digital recording device in the late 1990s.

The recordings had the advantage of providing two channels on which the signals from two transducers could be recorded simultaneously. The recording would then be played back and analyzed in the office. Time histories were produced using the Metrosonics Chart Recorder and if octave band vibration velocity levels were required, the signal was run through the Brüel and Kjær 1233 Real Time Analog Frequency Analyzer. See Figure 2.

Early on, the data output would be manually recorded off the screen of the Frequency Analyzer for inclusion in reports. However, the early 1980s was also the beginning of the age of personal computers. A newly acquired Osborne1 portable microcomputer provided an interface that could directly output the octave band data in graphical presentation. Computers in the office also meant that reports were no longer typed but prepared using word processing programs.

Similarly to the previous twenty years, the methodology of measurement and assessment remained largely unchanged until the turn of the century. The Osborne1, Chart Recorder and Frequency Analyzer, long obsolete yet still functional, remained in service for this sole purpose. Again, there was little impetus to invest funds in new equipment.



Figure 2: Brüel and Kjær 1233 Real Time Analog Frequency Analyzer (left) and Metrosonics Chart Recorder (right)

2.3 Mid 2000s to today

Although old technology at the time, the new millennium finally brought forward a jump to "direct to hard drive" analysis for vibration measurements. The LMS Pimento system included a front end A/D box (which provided not only the converters but also the biasing output) as well as a back-end software application to manage data capture, storage, and post-processing. With the implementation of time domain and frequency domain processing, the analysis possibilities were essentially endless. However, with the new applications and possibilities came new challenges. Laptop computers became an integral part of the measurement signal chain and as such needed to be purchased and managed. Software licensing became a key component of the overall system as "pay-for-functionality" was now part of the commercial landscape for equipment manufacturers. And of course, transporting and powering the entire data acquisition system was a logistical puzzle.

Later in the decade, the City of Toronto recognized the potential adverse impact that construction vibration could have on neighbouring structures and enacted the "Construction Vibration By-Law" (514-2008)[1]. The By-Law required that any building which could be negatively impacted by adjacent construction (as defined in the By-Law) be monitored for vibration according to a specific set of frequency-dependent criteria. With the By-Law came a further equipment boom with companies previously known for blast vibration monitoring driving a new market in construction vibration monitoring equipment. The blasting equipment brought with it advances including powerful onboard processing, full-time cellular connectivity, and real-time notification of events over a given threshold. Owing to its blasting roots, the equipment was also extremely durable and portable with long-lasting onboard batteries (although the cellular modems meant that AC power was still required).

2.4 Future

Demands for new advances in equipment are constantly arising. Construction specifications are being written with more and more stringent vibration monitoring requirements and often include criteria which require new flexibility in equipment operation. Based on the current trend, the equipment of tomorrow will likely be:

- Wireless with central data collection and cloud connection for data transfer and processing;
- Durable and portable;
- Long on battery life and may include alternate power supply options;
- Low-cost on a per-sensor basis to allow more discrete measurement points;
- Powerful with regard to onboard processing and rapid data presentation.

3 Conclusion

Measurement, assessment and reporting on environmental ground borne vibration has changed significantly over the last 50 years. However, the integration of new technology into acoustical consulting practice is highly influenced by changes to regulatory guidelines as well as capital cost.

References

[1] City of Toronto Municipal Code, Chapter 363, Building Construction and Demolition, Bylaw 514-2008.

GRADIENT REPRESENTATIONS IN SEABED GEOACOUSTIC INVERSION BY BERNSTEIN POLYNOMIALS

Jorge E. Quijano*[1], Stan E. Dosso†[1], Charles W. Holland‡[2] and Jan Dettmer‖[1]
[1]School of Earth and Ocean Sciences, University of Victoria, Victoria BC Canada.
[2] Applied Research Laboratory, The Pennsylvania State University, State College PA, USA

1 Introduction

Geoacoustic properties of the upper-most transition layer of mud seabed sediments often change rapidly with depth as continuous gradients, rather than discontinuous layers. However, most geoacoustic inversion approaches are based on layered sediment models. This paper presents a seabed parameterization approach that represents continuous geoacoustic gradients as a sum of Bernstein polynomial basis functions weighted by unknown coefficients which are estimated by Bayesian inversion of seabed acoustic reflectivity data. The Bernstein representation is efficient/effective in representing a wide variety of gradients with a small number of coefficients, and has optimal numerical stability to perturbation of the coefficients in the nonlinear inversion scheme. The Bernstein parametrization in geoacoustic inversion is applied to simulated data to illustrate the ability of the Bernstein polynomial parameterization to represent steep and strongly-variable gradients.

The development of acoustic remote sensing techniques for seabed parameter estimation is an active area of research, relevant to a variety of fields such as sonar signal processing, underwater noise modelling, and bioacoustics.[1,2] With a trend of increasing computational power accessible to most researchers [including parallel computing architectures and graphic processing units (GPUs)], application of sophisticated Bayesian geoacoustic inversion methods can be used to exploit the information content of acoustic data to reveal fine-scale seabed layering structure, and to estimate sediment sound speeds, densities, attenuations, and parameter uncertainties.[3,4] A key component in geoacoustic inversion is the efficient representation of the unknown media under study (i.e., the seabed) by a proposed mathematical model using only a few parameters. The majority of works on seabed geoacoustics have focused on layered models [Fig. 1(a)], appropriate to represent seabeds with sharp depth-dependent transitions between different kinds of sediment as a function of depth. However, such representation is highly inefficient in cases of smooth depth-dependent variations [Fig. 1(b)], such as those observed in the sound speed and density of mud sediments. This work proposes the use of Bernstein polynomials as the natural choice to represent continuous sediment properties.

The acoustic data considered in this work are the seabed bottom loss (BL) defined as[1,2]

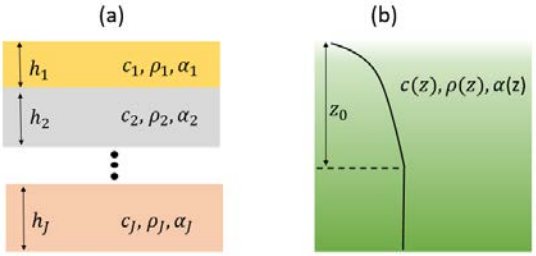


Figure 1: Representations of depth-dependent variation of seabed parameters: (a) discrete layering with constant geoacoustic properties within each layer and (b) continuous variation at $0 \le z \le z_o$.

$$BL(\theta, f) = -10\log_{10}|R(\theta, f)| \quad (1)$$

where θ is the grazing angle, f is the frequency, and $R(\theta, f)$ is the seabed reflection coefficient, which in field experiments can be obtained by the wide-angle reflection technique.[5] This quantity has been shown to carry substantial information content on seabed geoacoustics at and near the angle of intromission for muddy (silty-clay) sediments. The inversion approach in this work exploits this information.

2 Method

2.1 Gradient Bernstein representation

Let $c(z)$ and $\rho(z)$ be continuous functions representing the (smooth) depth-dependent variation of the sound speed and density in the seabed, respectively, as illustrated in Fig.1(b). Their polynomial representations in Bernstein form are[6]

$$\begin{aligned} c(z) &= \textstyle\sum_{j=0}^{J^c} g_i^c b_j(z, J^c), \\ \rho(z) &= \textstyle\sum_{j=0}^{J^\rho} g_i^\rho b_j(z, J^\rho), \end{aligned} \quad (2)$$

where J^c and J^ρ are the order of the polynomial, g_i^c and g_i^ρ are real coefficients determined by Bayesian inversion, and[6]

$$b_j(z, J^{c,\rho}) = \binom{J^{c,\rho}}{j} (1-z)^{J^{c,\rho}-j} z^j, \quad (3)$$

are the Bernstein basis functions for $0 \le z \le z_o$.

2.2 Bayesian geoacoustic inversion

Bayesian geoacoustic inversion methods estimate the posterior probability density (PPD) defined as[3,4]

$$P(\mathbf{m}|\mathbf{d}) = \frac{P(\mathbf{m})P(\mathbf{d}|\mathbf{m})}{Z}, \quad (4)$$

where $\mathbf{d}$ is a vector containing $BL(\theta, f)$ at N_θ angles and N_f frequencies, Z is a normalizing constant, $P(\mathbf{m})$ represents prior information on the sediment geoacoustics, the

* jorgeq@uvic.ca
† sdosso@uvic.ca
‡ cwh10@psu.edu
‖ jdettmer@uvic.ca

conditional probability $P(\mathbf{d}|\mathbf{m})$ is interpreted as the likelihood function model, and

$$\mathbf{m} = \left[z_o\ g_0^c\ g_1^c\ \ldots\ g_{J^c}^c\ g_0^\rho\ g_1^\rho\ \ldots\ g_{J^\rho}^\rho\ \alpha\right], \qquad (5)$$

is the vector used to parametrize the seabed according to the Eqs. (2) and (3), with α representing the sediment attenuation (assumed constant over depth in this study). Predicted BL data, $\mathbf{d}(\mathbf{m})$, can be obtained as the plane wave reflection coefficient (PWRC) corresponding to sediments described by $\mathbf{m}$. Assuming that data residuals $\mathbf{e} = \mathbf{d} - \mathbf{d}(\mathbf{m})$ are Gaussian distributed with covariance matrix $\mathbf{C_d}$, the likelihood is defined as[3-4]

$$P(\mathbf{d}|\mathbf{m}) = \frac{1}{(2\pi)^{N_\theta N_f/2}|\mathbf{C_d}|^{1/2}} \exp\left(-\frac{1}{2}\mathbf{e}^T\mathbf{C_d^{-1}}\mathbf{e}\right). \qquad (6)$$

In this work, PWRC predictions were implemented on a GPU for efficient computation. The PPD was quantified using Metropolis-Hastings sampling.[3,4] To provide a more complete search over the parameter space, parallel tempering is applied by running 16 interacting Markov chains at a sequence of sampling temperatures between 1 and 5, with chains at higher temperatures sampling wider regions of the parameter space.

3 Results

Simulated BL data with added zero-mean Gaussian noise (0.5 dB standard deviation) were computed for a sediment profile that mimics core measurements[7], with a steep positive depth-dependent gradient for the density and a mild negative gradient for the sound speed. For the inversion, non-informative uniform distributions with bounds given in Table 1 were assumed.

Table 1: Bounds of the uniform distributions used as priors.

Parameter	Units	min	max
z_o	m	0	1
α	dB/m/KHz	0	0.05
g_i^ρ	g/cm^3	0.5	2
g_i^c	m/s	1400	1700

Figure 2 shows marginal PPD profiles for sound speed, density, and attenuation obtained by Bayesian sampling, with $J^c = 5$ and $J^\rho = 6$. At most depths, the PPD overlaps the true sound speeds and densities, capturing some of the fine scale features of the profiles but requiring only a small number of parameters. Attempting such inversion with a layered modeled would substantially enlarge the dimensionality of the parameter space: instead of the $2+J^c + J^\rho$ parameters used by the Bersntein representation, a layered model would require many layers with four unknowns per layer to represent a continuous profile. Inversions in large parameter spaces are computationally challenging, and are prone to introduce artifacts in the form of sharp transitions in the estimated PPDs.

4 Conclusions

This paper discusses the use of Bernstein polynomials to represent continuous variation in seabed parameters. In addition to efficient representation of smooth variations, Bernstein polynomials can be shown to exhibit optimal numerical stability when its weighting coefficients are perturbed, which is a highly advantageous property for gradual exploration of multi-dimensional parameter spaces in Bayesian sampling inversions

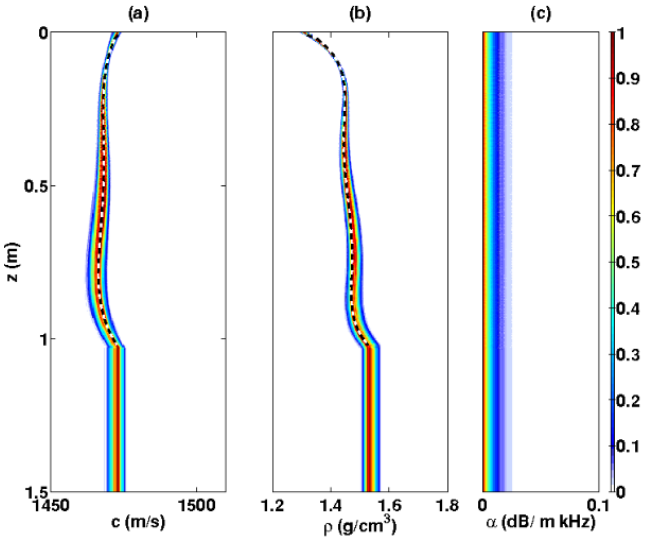


Figure 2: (a) Sound-speed, (b) density, and (c) attenuation profiles obtained by Bayesian inversion of simulated BL data. Dashed lines indicate true parameters.

References

[1] R. J. Urick, Principles of underwater sound (McGraw-Hill, Inc., New York, 1983), pp. 1-423.

[2] F. B. Jensen, W. A. Kuperman, M. B. Porter, and H. Schmidt, Computational Ocean Acoustics (American Institute of Physics, New York, 1994), pp. 1-794.

[3] J. Dettmer, C. W. Holland, and S. E. Dosso, "Analyzing lateral seabed variability with Bayesian inference of seabed reflection data", J. Acoust. Soc. Am. 126, 56–69 (2009).

[4] G. Steininger, S. E. Dosso, C. W. Holland, and J. Dettmer, "A trans-dimensional polynomial-spline parameterization for gradient-based geoacoustic inversion", J.Acoust. Soc. Am. 136, 1563–1573 (2014).

[5] C. W. Holland and J. Osler, "High resolution geoacoustic inversion in shallow water: a joint time- and frequency-domain technique", J. Acoust. Soc. Am. 107, 1263–1279 (2000).

[6] R. T. Farouki, "The Bernstein polynomial basis: A centennial retrospective", Comp. Aided Geometric Des. 29, 379–419 (2012).

[7] C. W. Holland, J. Dettmer, and S. E. Dosso, "Remote sensing of sediment density and velocity gradients in the transition layer", J. Acoust. Soc. Am. 118, 163–177 (2005).

Acknowledgments

The authors gratefully acknowledge NSERC support. Jan Dettmer acknowledges the support of the Office of Naval Research grant N000140910394.

WE CAN MAKE WHATEVER YOU MAKE

BETTER.
SAFER.
QUIETER.



Custom insulation solutions for acoustical panels start here.

Made from basalt rock, ROXUL® Core Solutions (OEM) insulation provides excellent acoustical dampening and fire resistant properties. So whether you're designing acoustical panels, a music studio or modular office – our team of experts will work with you before, during and after fabrication to create the custom sound solution you're looking for.
Visit us at **roxul.com/oem**

CONROCK® Series.
For the better way to build.

ROXUL®
The Better Insulation™

HEARING HEALTH AND AGING - SANTÉ AUDITIVE ET VIEILLISSEMENT

Audiologic Care For Clients With Multiple Medical Comorbidities: Modifications To Clinical Practice And Inclusion Of Family Members And Caregivers
Kate Dupuis, Ulrike Lemke, Marilyn Reed, M. Kathleen Pichora-Fuller 120

Improving Communication For Persons With Dementia
Sara Mamo, Samantha J. Mayhew, Matthew McNabney, Esther S. Oh, Frank R. Lin 122

Association Between Sensory Loss And Social Outcomes: A Preliminary Report
Maksim Parfyonov, Paul Mick, Kathy Pichora-Fuller, Walter Wittich 124

Acoustic Factors Related To Emotional Responses To Sound
Erin Margaret Picou 126

Addressing Social Isolation In Hearing Rehabilitation
Marilyn Reed 128

Promoting Hearing And Communication Health In Long-Term Residential Care Through The Use Of Communication Apps
Jeff Small, Rozanne Wilson, Alex Milhailidis, Diana Cochrane 130

AUDIOLOGIC CARE FOR CLIENTS WITH MULTIPLE MEDICAL COMORBIDITIES: MODIFICATIONS TO CLINICAL PRACTICE AND INCLUSION OF FAMILY MEMBERS AND CAREGIVERS

Kate Dupuis*[1,2], Ulrike Lemke†[3], Marilyn Reed‡[1], and M. Kathleen Pichora-Fuller[Δ2,4]
[1]Department of Audiology, Baycrest Health Sciences, Toronto, Canada
[2]Department of Psychology, University of Toronto, Toronto, Canada
[3]Science and Technology, Phonak AG, Stäfa, Switzerland
[4]Rotman Research Institute, Toronto, Canada

1 Introduction

Older adults seeking treatment for hearing loss often experience problems in daily listening and communication situations. These self-reports illustrate the negative effects of hearing loss on everyday activities, and the importance of considering a client's typical listening environments when creating a rehabilitation plan.

Hearing impairment is the third most common chronic condition in older adults [1], affecting 25%-40% of people aged 65 years and older, and more than 90% of people aged 90 years and older [2,3]. With the aging of the population, it is anticipated that there will be an increasingly larger number of older adults who have hearing impairment.

People may become aware of changes to their hearing in middle age, but the average age of a first-time hearing aid user is 63 years old [4]. Indeed, individuals often wait ten or more years after hearing loss is diagnosed before pursuing treatment [5]. Most people with hearing loss will never seek treatment; only 14% of adults aged 50 years and older use a hearing aid [6]. Even if an individual seeks treatment and obtains a hearing aid, this does not guarantee successful use of their device because care and maintenance can be a barrier to successful use of hearing aids [7].

A number of medical conditions could affect an individual's ability to successfully use and maintain their hearing aids, including vision impairment, manual dexterity issues, and cognitive impairment. Approximately 15% of individuals aged 75 years and older experience vision loss even when using glasses or contact lenses [8]. An individual with vision loss may have difficulty operating and handling their device (e.g., inserting a battery upside-down). Changes to hand or arm function (e.g., osteoporosis, hand osteoarthritis, hand tremor) are also common in older age [9], and can affect the ability to properly insert and remove hearing aids and operate controls. Finally, cognitive impairment affects approximately 7-11% of individuals aged 65 years and older [10], and may make it difficult for a client to pay attention during testing, or to recall the operating instructions for their hearing aids.

Audiologists need to be aware of the special needs of their older clients who have health issues that could negatively affect all aspects of audiologic rehabilitation, including hearing testing, hearing aid evaluation, hearing aid fitting, and check-ups. Audiologists may modify assessment and treatment plans to accommodate health issues, and may include family members and caregivers in rehab planning as part of a family-centered approach to client care [11].

Older adults with sensory and cognitive losses will also have special needs in everyday life and their ability to function can be facilitated or foiled by the properties of the physical environment. The design of spaces to optimize acoustics, lighting, and other features to create supportive environments for communication and navigation will be tremendously important as the population ages.

In the current study, a chart review was conducted in a geriatric audiology clinic to determine: Goal 1: Frequency of medical comorbidities related to vision, manual dexterity, and cognition, as noted by the audiologists; Goal 2: Specific modifications to clinical practice made by the audiologists to accommodate these comorbidities; Goal 3: Level of involvement of family members/friends/caregivers in audiologic rehabilitation.

2 Method

2.1 Participants

This study was conducted in the Audiology Clinic at Baycrest Health Sciences, a geriatric healthcare facility in Toronto. Participants were 136 clients who completed a Hearing Aid Evaluation (HAE) with a clinical audiologist in 2015. The mean age of the participants (61% female) was 86.2 years (*SD* = 7.5). Fifty-six percent of the participants were experienced hearing aid users.

2.2 Data abstraction

Data from each client's Audiology chart were abstracted by a research assistant trained by the project lead (KD). Inter-rater reliability on six charts used for training was > 80%. Data were entered manually from each chart into specially created Medialab (Empirisoft, 2014) programs. All sections of the charts were abstracted; the sections most relevant for the current study include HAE (client's amplification needs, social support, etc.), Hearing Aid Prescription and Fitting (HAF: details about prescribed device(s)), Hearing Aid Check (HAC: information regarding the client's success/challenges with the hearing aid(s)).

* kdupuis@baycrest.org
† ulrike.lemke@sonova.com
‡ mreed@baycrest.org
Δ k.pichora.fuller@utoronto.ca

3 Results

3.1 Goal 1

Prevalence rates were 52% (70/136 clients) for visual issues, 38% (52/136) for manual dexterity issues, and 44% (59/136) for cognitive issues. The most common elaborations for each type of issue are shown in Table 1.

Table 1. Examples of most commonly noted comorbid medical issues (with *n*s and % of responses corresponding to that example in brackets)

Visual Issues	Wears glasses/reading glasses (39/50; 78%)
	Macular degeneration (5/50; 10%)
	Cataracts (3/50; 6%)
Manual Dexterity Issues	Arthritis (11/38; 39%)
	Clumsiness (3/38; 8%)
	Tremors (2/38; 5%)
Cognitive Issues	Alzheimer's disease (6/47; 13%)
	Dementia (not otherwise specified) (6/47; 13%)
	Mild cognitive impairment (3/47; 6%)

3.2 Goal 2

The audiologists noted how these medical issues could influence the rehabilitation process at the HAF and HAC appointments. Examples of prescription modifications include using the largest removal cord possible and use of rechargeable aids (to avoid having to deal with replacing batteries). Examples of check modifications or commentary include showing the spouse how to use the client's remote control, re-instructing caregivers on insertion techniques, noting that the daughter changes the client's batteries, and postponing modifications to hearing aid output to the next appointment due to the time spent on counseling.

3.3 Goal 3

In 42% of the cases (57/136 clients), the audiologist indicated that the client would need daily help to successfully use their hearing aid(s). For 46/57 clients, they elaborated on who would provide this help (see Table 2). The audiologists also gave specific examples of the different categories of help provided for the clients, including: Overall management (e.g., "Requires others to manage"); Battery management (e.g., "Can manage aid except for changing batteries"); Spouse involvement (e.g., "Wife will help, wants it as simple as possible"); Device cleaning (e.g., "Caregiver cleans the hearing aids").

Table 2. Individuals providing assistance ($n = 50$)

Individual providing care	
Formal caregiver	22 cases
Family member- spouse	10 cases
Family member- child	6 cases
Family member- grandchild	1 case
Family member- sibling	1 case
Close friend	1 case
Employee at long-term care facility (LTC)	7 cases
Caregiver & LTC employee	1 case
Family member & LTC employee	1 case

4 Discussion and Conclusions

There were high levels of comorbid medical conditions in the older clients seeking treatment for hearing loss. These findings underscore the importance of involving family members and caregivers in audiologic rehabilitation and of being sensitive to the effects of other medical conditions in order to optimize treatment success.

Although hearing aids are one important aspect of audiologic rehabilitation, audiologists also commonly counsel clients about the importance of choosing (where possible) the best possible communication environments (e.g., ensuring lights are on to improve speech reading cues, choosing quiet restaurants with carpeted floors, or sitting at a corner table). While communication partners can assist clients in this endeavor, with a rapidly growing population of older adults, those designing the spaces where these clients congregate and communicate (e.g., recreation centers, long-term care facilities) must consider sensory, physical, and cognitive abilities in their planning.

References

[1] Collins, J.G. (1997). Prevalence of selected chronic conditions: United States 1990-1992. *Vital and Health Statistics, 10*(194). Hyattsville, MD: National Center for Health Statistics.

[2] Yueh, B., Shapiro, N., MacLean, C.H., & Shekelle, P.G. (2003). Screening and management of adult hearing loss in primary care: Scientific review. *JAMA, 289,* 1976-1985.

[3] Cruickshanks, K.J., Wiley, T.L., Tweed, T.S., Klein, B.E.K., Klein,R., Mares-Perlman,J.A., & Nondahl, D.M. (1998). Prevalence of hearing loss in older adults in Beaver Dam, WI: The Epidemiology of Hearing Loss Study. *Amer J Epidemiology, 148*, 879-886.

[4] Abrams, H.B., Kihm, J. (2015). An introduction to MarkeTrak IX: A new baseline for the hearing aid market. *Hearing Review, 22*(6), 16.

[5] Davis, A., Smith, P., Ferguson, M., Stephens, D., & Gianopoulos, I. (2007). Acceptability, benefit and costs of early screening for hearing disability: A study of potential screening tests and models. *Health Technology Assessment, 11,* 1-294.

[6] Chien, W., & Lin, F. R. (2012). Prevalence of hearing aid use among older adults in the United States. *Arch Internal Medicine*, *172*(3), 292-293.

[7] McCormack, A., & Fortnum, H. (2013). Why do people fitted with hearing aids not wear them? *Int J Audiol*, *52*, 360-368.

[8] Schiller, J.S., & Peregoy, J. A. (2012). Provisional report: Summary health statistics for U.S. adults: National Health Interview Survey, 2011. *National Center for Health Statistics, 10(256)*.

[9] Singh, G. (2010). The aging hand and handling of hearing aids: A review. In L. Hickson (Ed). *Hearing care for adults 2009- The challenge of aging* (pp. 265-276). Chicago, IL: Phonak AG.

[10] Alzheimer's Association (2016). 2016 Alzheimer's disease facts and figures. Retrieved from http://www.alz.org/facts/

[11] Singh, G., Hickson, L., English, K., Scherpiet, S., Lemke, U., Timmer, B., … & Saunders, G. H. (2016). Family-centered adult audiologic care: A Phonak position statement. *Hearing Review, 23*, 16-21.

IMPROVING COMMUNICATION FOR PERSONS WITH DEMENTIA

Sara K. Mamo[1,2], Samantha J. Mayhew[4], Matthew McNabney[1], Esther S. Oh[1], and Frank R. Lin[1,2,3]
[1]Johns Hopkins School of Medicine, [2]Johns Hopkins Center on Aging and Health,
[3]Johns Hopkins Bloomberg School of Public Health, [4]University of Cincinnati College of Medicine

1 Introduction

Older adults with hearing loss and dementia may be particularly at risk for further cognitive and functional decline due to the combined effects of hearing loss and dementia on communication and social engagement. Further, the presence of hearing loss can exacerbate dementia symptoms, making the disease stage appear more severe than the person's true cognitive function. Importantly, there is evidence that treating hearing loss can reduce dementia-related problem behaviors [1, 2].

The purpose of this project is to provide basic hearing intervention services in a group care environment to improve access to hearing loss treatment for older adults with dementia and their families. By integrating services into an interdisciplinary, comprehensive health program, this project aims to improve communication for at risk adults attending a Day Health Center (DHC) for social engagement opportunities.

During Phase I of developing an affordable and accessible group care hearing intervention, a series of observations and focus groups were undertaken to assess the social environment at the DHC. Using exploratory qualitative methods, findings from Phase I will inform the development of staff training and small group therapy sessions to support communication for individuals with age-related hearing loss and dementia.

2 Methods

2.1 Participants

The participants in this research project include employees at the Johns Hopkins ElderPlus Program of All-inclusive Care for the Elderly (PACE; n=13) and PACE participants (n=5) who were selected in partnership with the PACE recreation and rehabilitation therapists as well as members of the medical team.

2.2 Direct observation

Group activity

Over a two-week period, activities in the DHC were observed and field notes were recorded by two researchers (SKM and SJM). Observations were completed in 2-hr blocks and sampled throughout the entire day of services. The researchers collected field notes together for the first two sessions in order to compare field notes and develop a similar system, including rich description of communication interactions. Through an iterative process, a codebook was developed for coding the field notes. The coded field notes were then analyzed for themes, with a particular interest in communication between staff and participants as well as between participants.

smamo1@jhmi.edu
mayhewsj@mail.uc.edu; mmcnabne@jhmi.edu
eoh9@jhmi.edu; flin1@jhmi.edu

Individual activity

Behavioral coding was undertaken with five target individuals at the DHC. The individual participants were invited to participate based on having hearing loss and cognitive impairment. The invited individuals were chosen due to having low activity levels at the DHC per reports by the recreation therapy staff.

Based on pilot observations, a coding system was developed through an iterative process to capture frequency counts of communication-related behaviors. All coding sessions were completed by at least two researchers and compared after each session to improve the fidelity of the coding system.

2.3 Staff focus groups

Three focus groups (n=13) were held as an opportunity to learn about the staff's perception of hearing loss and communication challenges with PACE participants. Topics in the focus group guide included communication challenges and repair strategies, training interests, and amplification devices. The audio recordings were transcribed and a research team read and discussed the text in order to develop codes. The coded passages in the transcripts were compared and discussed to improve the reliability of the coded passages.

3 Results

3.1 Communication environment

Overall impressions of communication in the DHC included that it was a friendly and interactive environment that was, at times, too loud for clear communication. Through analysis of the coded transcripts, a pattern emerged such that the staff-participant interactions fell into categories that mapped onto common communication strategies. A brief sample of categories are shown in Table 1.

Table 1: Observations at the DHC were coded and grouped into examples of communication strategies.

Strategy	Successful Example	Unsuccessful Example
Get Face-to-Face	*[Participant] is calling out about things she needs. Two staff members talk to her right next to her ear…*	*The noise level began to increase …Participants & staff were trying to talk…across the room. A lot of 'huhs' were heard.*
Use Nonverbal Cues	*A staff member tells 3 participants to close their eyes…The staff member then repeats herself, closes her own eyes…points to her eyes.*	*A staff member pushes the wheel chair directly behind the participant and tells him to sit down…He pulls the participant into his wheel chair from behind by the hips.*

3.3 Behavioral characteristics of individual participants

Between individual participants, the levels of behavioral responses varied, but within subjects, steady behavioral responses were seen across 2-weeks of baseline recording. An example of responses for one individual is shown in Figure 1. This figure shows the frequency count of the 'look' behavior, which was operationally defined as orienting to a person in the environment in a purposeful way (e.g., a person walking past, saying hello, giving instructions).

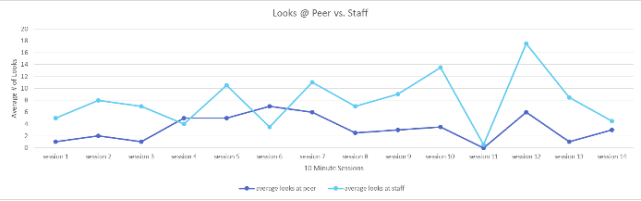


Figure 1: 'Look' responses for one individual at 13 observation blocks. Shown are number of 'looks' at peers (blue) and staff (teal). Each observation block is a 10 minute observation episode collected across multiple days and activity opportunities. Response range = 0-18.

3.2 Staff perceptions and communication strategies

Analysis of the coded transcripts from the staff focus groups revealed several specific training and communication support opportunities. Common compensatory strategies (e.g., getting closer to the person or taking the person to a quiet place) that the staff developed through experience were discussed across the groups as well as a desire to improve knowledge regarding signs of hearing loss, degrees of hearing loss and its impact on communication, and how to use over-the-counter amplification to improve hearing, especially during medical appointments.

4 Discussion and conclusion

The purpose of this exploratory data set is to build an effective multicomponent intervention that supports communication in a group care environment for individuals with hearing loss and dementia. Direct observation at the DHC revealed opportunities to incorporate real-life scenarios into a staff training focused on communication strategies. Moreover, the observational coding of individual participants provides the groundwork for using behavior in the activity hall as the primary outcome of a future intervention investigation. Finally, the focus groups provided insight into the barriers experienced by staff in terms of communicating with some participants and identified topics that would be welcomed with regards to training.

Nearly all persons with dementia experience distressing problem behaviors at some point in the disease progression [3, 4]. Such behavioral symptoms are often treated with psychotropic medications that can have negative side effects. Developing and implementing an affordable and accessible intervention that improves the listening environment, staff awareness, and use of amplification devices among individuals with dementia in group care settings provides a low-risk, low-cost, non-pharmacological treatment that may improve social engagement and reduce dementia-related behavioral symptoms.

Acknowledgments

This work was supported by AAA (SKM), AFAR and JHSOM Div of Geriatrics and Gerontology (SJM), NIH/NIDCD K23DC011279 (FRL), NIH/NIA K23AG043504 (ESO), the Roberts Fund (ESO), and the Eleanor Schwartz Charitable Foundation (FRL).

References

[1] Allen, N.H., et al., *The effects of improving hearing in dementia*. Age Ageing, 2003. **32**(2): p. 189-193.

[2] Palmer, C.V., et al., *Reduction in caregiver-identified problem behaviors in patients with Alzheimer disease post-hearing-aid fitting*. J.Speech Lang Hear.Res., 1999. **42**(2): p. 312-328.

[3] Lyketsos, C.G., et al., *Neuropsychiatric symptoms in Alzheimer's disease*. Alzheimers Dement, 2011. **7**(5): p. 532-9.

[4] Kales, H.C., L.N. Gitlin, and C.G. Lyketsos, *Assessment and management of behavioral and psychological symptoms of dementia*. BMJ, 2015. **350**: p. h369.

ASSOCIATION BETWEEN SENSORY LOSS AND SOCIAL OUTCOMES: A PRELIMANARY REPORT

Maksim Parfyonov*[1], Paul Mick[† 2], M. Kathleen Pichora-Fuller[‡ 3] and Walter Wittich[Δ4]
[1]Faculty of Medicine, University of British Columbia
[2]Division of Otolaryngology, Department of Surgery, University of British Columbia
[3]Department of Psychology, University of Toronto, Ontario
[4]School of Optometry, University of Montreal

1 Introduction

Social relationships are an important determinant of health. Indeed, a recent meta-analysis of 148 studies found a 50% greater likelihood for survival for individuals with stronger social relationships, comparable to the effect on mortality of smoking and alcohol consumption, and exceeding the influence of physical inactivity and obesity [1]. The effects of social relationships on health may be mediated through social network size or characteristics, participation in social activities, availability of social support, or loneliness [2]. Our population is growing older, and a large segment of seniors report feeling isolated or lonely. Improving social function is therefore an important public health goal. One of the ways in which we can work towards this goal is by addressing modifiable risk factors or by directing social interventions towards at-risk populations.

Hearing and sight affect mobility and communication and thus may influence an individual's social function. The degree to which sensory loss affects social lives at the population level is not well understood. Presbycusis and uncorrectable age-related vision loss are common, increasing in prevalence, and important for social interaction. Studies examining the relationship between sensory loss and social outcomes have been qualitative, measured only single aspects of social function, focused on hearing or vision loss in isolation, or have not been conducted in the Canadian population. Therefore, we set out to determine if hearing loss, vision loss, or dual sensory loss are associated with social network diversity, social participation, availability of social support, and loneliness in a nationally-representative sample of Canadians ranging in age from 45 to 85 years, and whether age or gender modified these associations.

2 Method

2.1 Study cohort

The Canadian Longitudinal Study on Aging (CLSA) [3] is comprised of English- and French-speaking Canadians who, at baseline, were 45-85 years of age, did not live on First Nations reserves, were not institutionalized, and had normal cognition. Our analysis was conducted using the first wave of data released for the CLSA tracking cohort, a nationally representative sample of 21,241 adults who underwent a telephone survey [4]. Participants were excluded if they had missing data for any of the variables in the analytic dataset.

2.2 Outcome variables

Associations were analyzed between hearing, vision and dual sensory loss and social network diversity, social participation, availability of social support, and loneliness.

Social network diversity was determined using a slightly modified version of the Social Network Index (SNI) [5]. The original SNI measures 12 possible social roles. In the present analyses, participants were scored on the SNI out of 10 instead of 12 roles because the frequency of contact with parents or parents-in-law is not measured in the CLSA. Participants received 1 point if they were married or in a domestic partnership. They also received 1 point (each) if they had interpersonal contact at least every 1-2 weeks (over the past year) with children, other close family members, friends, neighbours, work colleagues, schoolmates, fellow volunteers, members of non-religious community groups, of members of religious groups.

Social participation was measured using 8 items developed for the Canadian Community Health Survey 4.2 [6]. Participants were classified as having low social participation if they did not participate in any social activities at least once per week. Activities included family or friendship activities outside the household, church or religious activities, sports or physical activities with others, educational or cultural activities with others, service club activities, community or professional association activities, volunteer work, or any recreational activity with others.

Availability of social support was determined using the Medical Outcomes Study-Social Support Survey [7], a validated scale of overall social support with four domains of social support (emotional/informational, tangible, affectionate, and positive interactions). For the overall score and the score for each domain, participants were categorized as having low availability of social support if their scores were below the median.

Loneliness was determined using the following questionnaire item: "In the past week, how often did you feel lonely?" Participants were classified as lonely if they responded "some of the time (1-2 days), "Occasionally (3-4 days)" or "All of the time (5-7 days)". Those who responded "Rarely or never (less than 1 day)" were considered not to be lonely.

* parfyonovm@gmail.com
† ptmick@gmail.com
‡ k.pichora.fuller@utoronto.ca
Δ walter.wittich@umontreal.ca

2.3 Sensory variables

Self-reported hearing was determined using the multiple-choice item: "Is your hearing, using a hearing aid if you use one…" Participants were classified as having hearing loss if they responded "fair" or "poor or non-existent/deaf" (versus "good", "very good" or "excellent"). Self-reported vision was determined using the item: "Is your eyesight, using glasses or corrective lens if you use them…" Participants were classified has having vision loss if they responded "fair" or "poor or non-existent/blind" (versus "good", "very good" or "excellent"). Individuals were considered to have dual sensory loss if they had both hearing loss and vision loss defined according to the criteria described above.

2.4 Covariates

In multivariate models, we adjusted for age, sex, race/ethnicity (white/non-white), annual household income (<$20,000, $20,000-49,999, $50,000-99,999, $100,000-149,999, >$150,000), education level (less than secondary school, secondary school graduate, some post-secondary school, post-secondary graduate), smoking status (never smoker, former smoker, current smoker), and dichotomous self-reported medical histories of diabetes, hypertension, angina, myocardial infarction, transient ischemic attack, stroke, cancer, kidney disease, hypo- or hyper-thyroidism.

2.5 Statistical analysis

Univariate analyses were performed to assess cross-sectional associations between hearing, vision and dual sensory loss and the social, demographic, and medical variables. The chi-square test was used to assess differences in proportions for categorical variables. Analysis of variance was used to test for differences between groups (defined by sensory status) on means for continuous variables.

Multivariate regression models were used to analyze associations between sensory loss (hearing, vision, dual) and SNI score, lack of social participation, low availability of social support and loneliness. Linear regression was used for SNI score and logistic regression for the other outcomes. The CLSA is a complex sample survey requiring mathematical variance approximation procedures to estimate sampling errors and produce estimates that are representative of the Canadian population. Inverse probability weights were used in univariate and multivariate analyses as per CLSA guidelines [8].

3 Preliminary Results & Discussion

Hearing loss was independently associated with low availability of social support and loneliness, but not low social network diversity or social participation. In contrast, vision loss and dual sensory loss were independently associated with poor scores on all social outcome variables.

These results are important for a number of reasons. Living with chronic health conditions often entails relying on others for help with instrumental tasks and emotional support. Social support can provide coping mechanisms that may mitigate the effects of sensory loss. Hearing loss has been associated with cognitive decline and dementia, and low social support or loneliness may mediate this pathway. The focus of therapy for these individuals is usually on augmenting sensory abilities (e.g., with hearing aids or lenses), without addressing activity and participation consequences in more comprehensive auditory or visual rehabilitation programs. Individuals with sensory impairments may also be benefit from community interventions that aim to increase participation and social engagement and reduce loneliness and isolation. Such community-based approaches could include more attention to the needs of older adults with sensory impairments when buildings and public spaces are designed and built.

Acknowledgments

The study is supported by the Canadian Consortium on Neurodegeneration in Aging funded by the Canadian Institutes of Health Research (CIHR). This research was also made possible using data/samples collected by the Canadian Longitudinal Study on Aging (CLSA), which is funded by the Government of Canada through the CIHR and the Canada Foundation for Innovation.

References

[1] Holt-Lunstad, J., Smith, T. B., & Layton, J. B. (2010). Social relationships and mortality risk: A meta-analytic review. *PLoS Medicine*, 7(7), e1000316.

[2] Berkman, L. F., & Glass, T. (2000). Social integration, social networks, social support, and health. In Berkman L.F. & Kawachi, I. (Eds.), *Social Epidemiology* (pp. 137-173). New York: Oxford University Press.

[3] Raina, P. S., Wolfson, C., Kirkland, S. A., Griffith, L. E., Oremus, M., Patterson, C., et al. (2009). The Canadian longitudinal study on aging (CLSA). *Can J Aging*, 28(03), 221–229.

[4] Kirkland, S. A., Griffith, L. E., Menec, V., Wister, A., Payette, H., Wolfson, C., & Raina, P. S. (2015). Mining a unique Canadian resource: The Canadian longitudinal study on aging. *Can J Aging*, 34(3), 366–377.

[5] Cohen, S. (1991). Social supports and physical health: Symptoms, health behaviors, and infectious disease. In Greene A.L., Cummings, M., & Karraker, K.H. (Eds.), *Life-span developmental psychology: Perspectives on stress and coping* (pp. 213-234). Hillsdale, NJ:Lawrence Erlbaum Associates, Inc.

[6] Statistics Canada. (2013). Canadian Community Health Survey (CCHS) – Cycle 4.2 Healthy Aging. Retrieved from Statistics Canada website: http://www23.statcan.gc.ca/imdb/p2SV.pl?Function=getSurvey&SDDS=5146&language=en&db=imdb&adm=8&dis=2

[7] Sherbourne, C. D., & Stewart, A. L. (1991). The MOS social support survey. *Social Science & Medicine*, 32(6), 705-14.

[8] Canadian Longitudinal Study on Aging. (2015). Canadian longitudinal study on aging protocol. Retrieved from Canadian Longitudinal Study on Aging website: https://datapreview.clsa-elcv.ca/sites/default/files/2015Mar27_sampling%20weights_technical%20document.pdf

ACOUSTIC FACTORS RELATED TO EMOTIONAL RESPONSES TO SOUND

Erin M. Picou*[1]
[1]Department of Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave S, Room 8310, Nashville, TN

1 Introduction

People with hearing loss are at increased risk of reduced subjective well-being, increased depressive symptoms and reduced quality-of-life [1]. One of the contributing factors to the relationship between hearing loss and negative psychosocial consequences may be disrupted emotional processing. Positive emotional experiences can improve well-being [2]. Negative emotional experiences can improve cognition and prepare a body for action [3]. Mild-moderate hearing loss has recently been shown to disrupt emotional processing [4] and the range of emotional responses to sound [5]. However, little is known about the effects of hearing loss or amplification on a listener's emotional response to sounds. The purpose of this study was to evaluate the effects of hearing loss and hearing aid use on emotional responses to non-speech sounds and to explore the potential related acoustic factors.

2 Method

2.1 Participants

Participants included 13 adults with bilateral, mild to severe sensorineural hearing loss (M = 65.6 years, range = 49 to 74), in addition to 13 adults with normal hearing (M = 57.3 years, range = 50 to 80). No participant reported middle ear, neurogenic, or psychological pathology. All participants passed a screening survey, indicating low risk of clinical anxiety or depression.

Participants with hearing loss, who were all experienced hearing aid users, were fitted with bilateral, behind-the-ear hearing instruments for research purposes. Hearing aids were verified using probe-microphone measures to match validated prescriptive targets. All digital features were disabled in both programs, except feedback reduction.

2.2 Procedures

2.2.1 Stimuli

Sounds were 75 non-speech sounds from the corpus of the International Affective Digitized Sounds (IADS-2) [6]. Example sounds included human emotion (laughter, crying), machine sounds (dentist drill, helicopter), animal sounds (mooing, barking), and music (classical, guitar). The 15 sounds in each of 5 pre-determined categories were chosen; categories included 1) pleasant, low arousal, 2) pleasant, high arousal, 3) neutral, 4) unpleasant, high arousal, and 5) unpleasant, low arousal. Fifteen sounds were selected to represent each category. Sounds were 1.5 seconds long and were matched to have the same peak level.

Participants used the Self Assessment Manikin (SAM) [7] to make subjective ratings of valence and arousal. For each dimension, the SAM provides 5 pictorial representations of emotional responses (ranging from low to high). Numbers 1 through 9 are indicated under the pictures and indicate low to high feelings on each dimension. During testing, participants listened to a sound and then rated their emotional response using a computer keypad. While responding, the SAM was displayed on a computer monitor. Stimuli were presented from a loudspeaker located at 0˚.

2.2.2 Conditions

Participants with normal and impaired hearing made ratings of valence and arousal for stimuli presented at 60 and 80 dB SPL. Participants with hearing loss also made ratings of valence and arousal for stimuli when using hearing aids (60 dB presentation level). Sounds were blocked such that all items were presented in one condition before progressing to another condition. Within a condition, the sounds were presented in a random order. Condition order was counterbalanced across participants.

3 Results

3.1 Subjective Ratings

Subjective ratings of valence and arousal were analyzed separately. Considering the unaided responses from both groups of participants, analysis results of the arousal ratings revealed significant main effects of Stimulus Category ($F(4, 89.9) = 6.3$, $p < 0.001$), Level ($F(1, 244.7) = 5.9$, $p < 0.05$), and Hearing ($F(1, 244.7) = 9.9$, $p < 0.01$). These results suggest that ratings of arousal were higher with the higher presentation level and were higher for listeners with normal hearing. The interactions were not significant. Analysis results of the valence ratings, displayed in Figure 1, revealed significant main effects of Stimulus Category ($F(4, 103.5) = 155.$, $p < 0.001$), Level ($F(1, 225.5) = 12.7$, $p < 0.001$), and Hearing ($F(1, 225.5) = 15.4$, $p < 0.001$). In addition, there was a significant Hearing x Category interaction ($F(4, 103.5) = 8.2$, $p < 0.001$). These results suggest that ratings of valence were significantly lower with the 80 dB presentation level and for listeners with hearing loss. Furthermore, hearing loss negatively impacted the ratings of sounds in the pleasant categories, but not the unpleasant categories.

Considering only listeners with hearing loss, analysis of arousal ratings revealed only a significant effect of Category ($F(4, 60.0) = 5.2$, $p < 0.01$). Analysis of the valence ratings revealed significant main effects of Condition ($F(2, 94.6) = 6.4$, $p < 0.01$) and of Category ($F(4, 79.9) = 90.9$, $p <$

* erin.picou@vanderbilt.edu

0.001). These results suggest that the ratings of valence were negatively affected by increasing the overall level, but there was a trend for hearing aid use to mitigate the negative effects of an overall increase in level (see Figure 1).

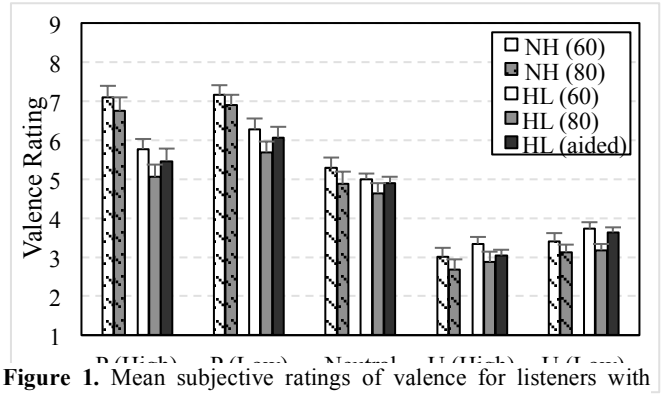


Figure 1. Mean subjective ratings of valence for listeners with normal hearing (hashed bars) and listeners with hearing loss (solid bars). Error bars represent +1 standard error from the mean.

3.2 Acoustic Factors

3.2.1 Hearing Loss

Correlation analysis, focused on ratings of pleasant stimuli, revealed a significant positive correlation between ratings of valence and arousal ($r = 0.17$, $p < 0.05$) and a significant negative relationship between ratings of valence and pure-tone average (better ear, 0.5, 1.0, 2.0, 4.0 kHz; $r = -0.34$, $p < 0.001$). These results suggest that higher ratings of valence were associated with higher ratings of arousal and also better hearing thresholds. Furthermore, sounds that were more sensitive to the effects of hearing loss were more likely to have less low frequency energy (through 600 Hz).

3.2.2. Hearing Aids

There was a significant relationship between a sound's high frequency content (2-6 kHz) and hearing aid effect; sounds with more high frequency energy were more likely to have lower valence ratings in the aided condition ($r = -0.34$, $p < 0.01$). In addition, sounds with the smallest dynamic range were most likely to be negatively affected by hearing aid use for valence ratings ($r = 0.29$, $p < 0.05$).

4 Discussion

The present results confirmed previous findings, suggesting that hearing loss negatively affects emotional responses to sounds [5]; listeners with hearing loss demonstrated a reduced range of ratings of valence and lower general arousal relative to their peers with normal hearing. The most notable effects of hearing loss on emotional responses of valence were related to ratings of pleasant stimuli. Participants with more hearing loss were more likely to be negatively affected. Hearing aid use did not increase ratings of valence for sounds in any category, although the effects were less detrimental than increasing the overall level.

Some of the effects of hearing loss and hearing aids were related to the acoustic properties of a signal. Specifically, signals with more low frequency energy were more resilient to the effects of hearing loss. Conversely, more high frequency energy in a signal was related to negative effects of hearing aid use on ratings of valence. These results suggest that manipulations of hearing aid gain may influence ratings of valence. Future work is warranted to explore the effects of hearing aid gain and other feature parameters on emotional responses to sound.

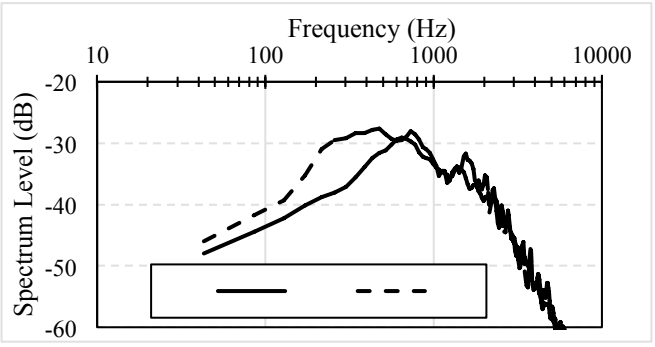


Figure 2. Average long term average spectrum of sounds that were sensitive to the effects of hearing loss (solid line) and of sounds that were not sensitive to the effects of hearing loss (dashed line).

5 Conclusion

Listeners with hearing loss exhibit a smaller range of emotional responses than their peers with normal hearing, primarily as a result of lower ratings of valence for pleasant stimuli. In addition, listeners with hearing loss reported lower levels of arousal. Although hearing aid use did not restore emotional responses, using personalized amplification had a less detrimental effect than increasing the overall level. These findings may have clinical implications for hearing aid fittings.

Acknowledgments

Funding for this project was provided by Sivantos, AG.

References

[1] Dalton, D., Cruickshanks, K., Klein, B., Klein, R., Wiley, T., & Nondahl, D. (2003). The impact of hearing loss on quality of life in older adults. *Gerontologist, 43,* 661 – 668

[2] Arthaud-Day, M. L., Rode, J. C., Mooney, C. H., & Near, J. P. (2005). The subjective well-being construct: A test of its convergent, discriminant, and factorial validity. *Soc Indic Res, 74,* 445 – 476

[3] Hermans, D., Houwer, J. D., & Eelen, P. (1994). The affective priming effect: Automatic activation of evaluative information in memory. *Cogn Emot, 8,* 515-533.

[4] Husain, F. T., Carpenter-Thompson, J. R., & Schmidt, S. A. (2014). The effect of mild-to-moderate hearing loss on auditory and emotion processing networks. *Front Syst Neurosci, 8,* 1 – 13

[5] Picou, E.M. (in press). How hearing loss and age affect emotional responses to non-speech sounds. *J Sp Lang Hear Res*

[6] Bradley, M. M., & Lang, P. J. (2007). *The International Affective Digitized Sounds (IADS-2): Affective ratings of sounds and instruction manual.* University of Florida, Gainesville, FL, Tech. Rep. B-3.

[7] Bradley, M. M., & Lang, P. J. (1994). Measuring emotion: the self-assessment manikin and the semantic differential. *J Behav Ther Exp Psychiatry, 25,* 49 – 59

ADDRESSING SOCIAL ISOLATION IN HEARING REHABILITATION

Marilyn Reed*
Department of Audiology, Baycrest Health Sciences, Toronto, Ontario, Canada

1 Introduction

There is mounting evidence of the link between hearing loss and cognitive decline [1]. While the causal mechanisms remain unknown, one of the most likely contributing pathways proposed thus far is the impact of hearing loss on communication and its effect on social participation [2, 3]. Social interaction is known to be a key determinant of healthy and successful aging, with direct causal pathways linking loneliness with physiological pathology [4, 5]. The challenges and frustrations posed by hearing loss often lead to exclusion from and avoidance of social interaction, causing isolation, loneliness and depression, known to be risk factors for dementia and other age-related declines [6]. In view of the high costs of dementia care, and in the absence of a cure, there is a pressing need to address hearing loss as a potential modifiable risk factor in order to help to prevent the onset of dementia and/or slow the rate of its decline [7].

Age-related hearing loss is a complex interaction of changes at all levels of the auditory system which may not be adequately addressed by amplification alone. Treating age-related hearing loss often requires supportive auditory rehabilitation, where audiologists can teach compensatory behavioral communication strategies to improve top-down information processing and help to compensate for auditory deficits [8]. Group rehabilitation programs are known to promote hearing and cognitive health by fostering participation and social interaction while members learn effective communication skills through interacting with their peers. Most hearing rehab groups are offered for a short period of time for new hearing aid users, and they do not address the social isolation felt by many seniors with hearing loss on a long-term basis. The Hard of Hearing Club was created as an innovative solution to social isolation imposed on seniors as a result of severe hearing loss [9].

2 Method

In keeping with the World Health Organization's International Classification of Functioning, Disability and Health (WHO ICF) model [10], the Hard of Hearing Club was designed to provide an opportunity for members to learn to communicate more effectively with their peers through regular social interaction in a supportive, accessible environment. The group was created in response to a psychiatrist's referral of a recently widowed, severely hearing impaired woman who had become depressed as a result of her social isolation.

The goal of the group was to address the psycho-social impact of hearing loss on the whole person, rather than limiting the focus to the hearing impairment. See Table 1 for the key components of the program, including small group size, accessible environment, shared personal factors (hearing loss, age, culture), and the discussion of shared concerns and solutions.

Table 1: Program Components

Accessibility	Educational	Social
- Environment: acoustics, seating, lighting - Small group size - Communication 'rules' followed - Facilitator ensures participation	- Share concerns and solutions - Practice good communication strategies - Assertiveness training - Self and group advocacy	- Long running - Weekly meetings - Shared personal factors - Discussion of common problems - Informal 'tea time'

Weekly meetings began in 2000 and continue to this day, allowing friendships to be developed and maintained over time. While there are many social and self-help groups for seniors, most are not accessible to these clients because of their severe hearing loss. Social fulfillment depends on all members being able to participate in group discussion, requiring the observation of 'rules' that they themselves compiled. A strong facilitator is needed to enforce these rules in order to enable all to participate and have the opportunity to practice and improve communication skills and enhance self-efficacy.

Rehab goals in the activity and participation domains were to optimize clients' communication skills and participation in activities that were challenging for them, thereby improving social interactions and their ability to cope with activities of daily living (Table 2). It was hoped that this would also address goals in the quality of life domain by reducing loneliness and increasing confidence and well-being.

3 Results/Outcomes

Outcomes of qualitative evaluation are shown in Table 3.

3.1 Client outcomes

Responses were grouped according to two main questions, namely 'why do you come to the group?', and 'how does the group help you?' The main emerging themes all address isolation, self-efficacy and coping with ADLs, showing that the group had achieved its objectives of addressing the quality of life issues that were initially identified. Self-efficacy is critical for older adults to put the communication

* mreed@baycrest.org

skills they have learned into practice and its enhancement is an invaluable component of the rehabilitative process [7].

Table 2: Hard of Hearing Club Goals

Activity and	Quality of Life
Reduce social isolation Improve social interactions Increase participation in group activities Develop friendships, social network	**Address loneliness/depression** Acquire sense of belonging to community Increase confidence and self-esteem Improve mood Provide optimism, hope
Coping with ADLs Improve communication skills Self-efficacy; confidence in daily interactions Participation in health care Access to environment and alarms through use of AT	**Independent living/safety** Manage ADLs independently. Enhance safety through use of AT and behavioral strategies. Use available resources effectively to support

3.2 Physician outcomes

The psychiatrist who referred the founding member reported that the group had had a "huge impact" on her depression and isolation and that it had fulfilled her need to connect with others, provided regular social context in a safe and comfortable environment, enabled her to participate in discussion, provided peer support, improved her self-esteem and enhanced her self-efficacy. She had become less reliant on her family and required fewer visits to her psychiatrist as a result. The physician was so impressed by the benefits of the Club that he felt it was important to educate others in psychiatry and psychology about this type of intervention.

Table 3: Outcomes of Qualitative Evaluation; Key Themes

Isolation	Self-Efficacy	Coping with ADLs
Reason to go out Something to look forward to Made friends, feel at home Feel supported, able to share problems Enjoy closeness, warmth	Assertiveness encouraged and practiced 'Can hear and be heard' Collective voice for advocacy Feel valued and respected	Educate about AT Provide and practice communication strategies Help clients manage daily activities "Learn something new every time"

3.3 Family/caregiver outcomes

Responses from members' children were extremely positive, indicating that the group provided a social life for people who understand each other's communication problems, addressed loneliness by providing a social group where members could share and learn about the challenges of life with severe hearing loss, and had given members a routine to look forward to and enjoy the company of others who can understand their feelings.

4 Discussion and Conclusion

The Hard of Hearing Club provides a successful rehabilitation option for seniors with severe hearing loss whose quality of life has been adversely impacted by their inability to participate in social activities.

Addressing social isolation has become a priority for many countries, including Canada, as we become increasingly aware of its negative impact on seniors' health. Because of the effects of hearing loss on communication ability, and in the light of mounting evidence that it may be a risk factor for cognitive and physical declines in the elderly, addressing this issue should be an obvious key component of any such strategy. Audiologists need to design new approaches to hearing health care for older adults in the context of healthy aging, going beyond hearing aid fitting to address their ability to communicate effectively and participate socially.

The Hard of Hearing Club is an example of a social model of hearing rehabilitation that has been shown to promote healthy aging by effectively addressing social isolation while teaching skills to help seniors with severe hearing loss cope with independent living.

References

[1] Lin FR, Yaffe K, Xia J, et al; Health ABC Study Group. Hearing loss and cognitive decline in older adults. JAMA Intern Med 2013; 173(4) :293-299.

[2] Mick P, Kawachi I, Lin FR. (2014). The association between hearing loss and social isolation in older adults. Otolaryngology-Head and Neck Surgery, 150, 378-384

[3] Pronk, M., Deeg, D. J. H., & Kramer, S. (2013). Hearing status in older persons: A significant determinant of depression and loneliness? Results from the Longitudinal Aging Study Amsterdam. AJA, 22, 316–320.

[4] Cacioppo JT, Hawkley LC, Norman GJ, Bernston GG. Social isolation. Ann NY Acad Sci. 2011; 1231;17-22

[5] Tomaka J, Thompson S, Palacios R. The relation of social isolation, loneliness and social support to disease outcomes among the elderly. J Aging Health 2006; 13(3):359-384

[6] Wilson RS, Krueger KR, Arnold SE, et al. Loneliness and risk of Alzheimer disease. Arch Gen Psychiatry.2007; 64:234-240

[7] Pichora-Fuller MK, Mick P, Reed M. Hearing, cognition and healthy aging: Social and public health implications of the links between age-related declines in hearing and cognition. Seminars in Hearing 2015; 36, 3, 122-139.

[8] Kiessling J, Pichora-Fuller MK, Gatehouse S, et al. Candidature for and delivery of audiological services: Special needs of older people. IJA 2003; 42 (Suppl 2):2S92-101.

[9] Reed M. The Hard of Hearing Club: A social framework for audiologic rehabilitation for seniors with severe hearing difficulties. In L. Hickson (Ed), Hearing care for adults: The challenge of aging. Phonak: Stafa, Switzerland; 2009.

[10] World Health Organization (WHO). International Classification of Functioning, Disability and Health (ICF). Geneva, Switzerland: WHO Press; 2001. Available at: http://www.who.int/classifications/icf/en/.

PROMOTING HEARING AND COMMUNICATION HEALTH IN LONG-TERM RESIDENTIAL CARE THROUGH THE USE OF COMMUNICATION APPS

Jeff Small,[*1] Rozanne Wilson, [†2] Alex Mihailidis[‡3] & Diana Cochrane[§1]

[1]School of Audiology and Speech Sciences, University of British Columbia, 2177 Wesbrook Mall, Vancouver BC V6T1Z3

[2]Department of Gerontology, Gerontology Research Centre, Simon Fraser University, Harbour Centre, 2800 - 515 West Hastings Street, Vancouver, BC V6B 5K3

[3]Department of Occupational Science & Occupational Therapy, Institute of Biomaterials & Biomedical Engineering University of Toronto, 500 University Ave., Toronto, ON, M5G 1V7

1 Introduction

Approximately half of seniors currently 65 years old will require long-term residential care (LTRC) at some point in their future, particularly when they have complex, chronic health problems. The majority of seniors living in LTRC experience some degree of sensory, language and/or cognitive impairments associated with various age-related conditions (e.g., dementia, hearing loss, stroke). Moreover, there is often considerable diversity in the language and cultural backgrounds of care staff and residents [1]. These functional challenges can negatively impact the quality of communication between care staff and residents [2], and ultimately quality of life. While attempts to address barriers to effective communication have been made [3], the proposed solutions are often inadequate in scope and impact or are unavailable when care staff need the support. As a result, older residents living with sensory, cognitive and/or language impairments become vulnerable in regard to communicating their needs and preferences, as well as in understanding those in their environment [4]. An emerging approach to supporting communication between care staff and residents is the use of mobile communication technology (e.g., smartphones, tablets, apps). The use of technology as a communication aide in LTRC has to date been explored primarily as a means to stimulate conversation in reminiscence and leisure activities rather than supporting communication during care (e.g., eating, personal hygiene). Consequently, little is known about whether and how LTRC staff have adopted mobile technology in their care practice for the purpose of enhancing communication. This gap was the motivation for the present study, which investigated LTRC staff's usage patterns, needs, and preferences for using mobile technology and communication apps (cApps) with residents in LTRC.

2 Method

2.1 Participants

- 12 Care Staff (11 F); Mean Age = 45 (24-59); 10 = RCA/HCA, 2 = Music/Art/Rec Therapist; 7 of 12 had noticeable accent in English, the language used for care in the facility.
- 6 Residents (4 F), Mean Age = 86 (75-92); Majority had dementia and/or stroke/aphasia; Half spoke a language other than English. Although not formally assessed, most residents presented with hearing and/or vision impairments.

[*]jeffs@audiospeech.ubc.ca
[†]rozanne_wilson@sfu.ca
[‡]alex.mihailidis@utoronto.ca
[§]diana.cochrane@mhlp.ubc.ca

2.2 Design & Procedure

We used a mixed-method design by collecting data from staff via questionnaires, focus groups, and real-time observations of staff-resident interactions. In the first stage, we conducted a comprehensive search of existing cApps in order to identify which ones might be most appropriate in addressing sensory, linguistic, and cognitive barriers in LTRC. We targeted cApps that presented solutions for language-related barriers (language translation apps) and sensory-cognitive-communication barriers (Augmentative and Alternative Communication—AAC—apps). We reviewed each cApp for its features and functionality using a screening and rating process. This process yielded 4 short-listed cApps—2 AAC-type apps, and 2 language translation apps. We then introduced the 4 cApps to LTRC staff to elicit through questionnaires, observations, and focus groups: 1) their perspectives and needs around the adoption of cApps in their daily work, 2) what aspects of current cApp technology do or do not meet their needs, and 3) what usability issues (e.g., display size and quality of images/video and audio) they face when using these cApps with residents who have cognitive-linguistic, sensory, and/or physical challenges.

3 Results

Half of the care staff (N = 6) indicated that they had previously used mobile technology (e.g., smart phones, tablets) with residents, while only one said he had also used cApps with residents. No staff mentioned or were aware of cApps designed specifically for visually or hearing impaired persons (e.g., Jacoti ListenApp®). As for other communication enhancing devices, an FM system was mentioned by two care staff as being available on each floor of the facility. However, most care staff seemed unaware of this technology. Care staff indicated that all 4 cApps were simple to understand, easy to use, and had many desirable features. Staff comments in the pre-/post-Focus Groups and our observations of staff using the cApps with residents highlighted a number of cApp features that were preferred by staff, as well as some challenges to using them:

Preferred cApp Features mentioned by care staff and/or observed while they used cApps with residents:

- Easy navigation that allows combinations of pictures/images/text/braille with speech in order to provide multi-modality input/output to accommodate challenges associated with hearing/vision loss, aphasia, dementia, and/or language differences/accent
- Customizable to support adding resident's personal photos, videos, music etc.—will help facilitate meaningful social interaction, agency, and improve mood. For example, a scene photo of the resident's wardrobe would allow resident to choose what to wear simply by pointing.
- Interfaces with headset/ear buds so that residents with hearing loss can adjust sound level
- Large display/images/font, and strong and clear audio capability (especially for translation)
- Ability to record, store, and play back native spoken language phrases for translation and accent accommodation; translation that can deal with accents and variations in voice quality
- Female (vs. male) voice—softer, more calm, more acceptable to residents; use of familiar voice may appeal to residents [and reduce cognitive load on perception]
- Sensors that allow object or text detection/scanning/tracking/conversion to speech for the visually impaired
- Pain and feelings scales that allow resident to communicate location and severity of pain, or current mood, using touchscreen

Challenges mentioned by care staff and/or observed while they used cApps with residents:

- Residents are often not familiar with mobile technology (e.g., tablets), and therefore, get confused/distracted when presented with the display
- Residents with hearing loss have difficulty with the relatively low audio quality on MT devices, particularly in noisy environments and when confronted with accents (staff, or canned voices)
- Device should be small/portable, yet it needs to allow for large enough display for residents
- Residents' touch of the display was often not detected or precise enough for successful operation of cApp

In the post-observation focus group, staff completed a worksheet that asked them to check off which features they would prioritize as essential for their "dream" cApp. Two of the features listed were related to the importance of accommodating to residents' vision and hearing impairments. The majority of staff (7/8) indicated that a cApp must address vision needs, whereas just over half (5/8) responded similarly for hearing needs.

4 Discussion

Care staff expressed positive perspectives on the potential of using cApps to facilitate communication with residents during daily care activities. By accommodating to the sensory, linguistic, cognitive, and social needs of residents, cApps allow interactions to progress more efficiently and in a more person-supportive manner. As one staff put it, "cApps promote an emotionally closer relationship/friendship, socializing, spending more time with resident, and having fun." Care staff's greater emphasis on vision than hearing issues may suggest a need to educate staff about the prevalence of hearing related challenges to communication in LTRC. However, it may also be an indication of the relatively greater salience of and dependence on visual than auditory functions when using AAC-type apps (e.g., visuals compensate for hearing loss).

5 Conclusion

Future advances in technology should enable integration of the best devices (e.g., FM system, smartphones, Google Glasses®) and cApps (AAC, translation, amplification) in order to provide users with one versatile portable platform that can address residents' sensory, motor, linguistic, and cognitive communication challenges. Wireless technology is now at a point where such integration would enable users to employ the device and accessories (e.g., hearing aids, remote microphone) without attaching cables and in any location [5]. A key factor, however, will continue to be achieving a balance between the need for portability (weight and size of device) and the requirement that features of the device and app support age-related functional changes. Given residents' variable health status and their complex care and communication environment, the usefulness of cApps will necessarily vary from resident to resident, across interactions and tasks, and from day to day. As with any technology designed for enhancing interpersonal interactions, its ultimate effectiveness depends on the users' awareness and appreciation of each other's needs and of how technology is just one ingredient in a tool bag of communication strategies [6].

Acknowledgments

This study was funded by TVN/CFN grant# HTIG2014-20. We are grateful for participants' contributions and insights.

References

[1] Small, J., Chan, S. M., Drance, E., Globerman, J., Hulko, W., O'Connor, D., … Ho, L. (2015). Verbal and nonverbal indicators of quality of communication between care staff and residents in ethnoculturally and linguistically diverse long-term care settings. *J Cross-Cultural Geron*, *30*(3), 285–304.

[2] Ryan, E. B. (2010). Overcoming communication predicaments in later life. In L. Hickson (Ed.), *Hearing care for adults* (pp. 77-86). Stäfa, Switzerland: Phonak.

[3] Wilson, R., Rochon, E., Mihailidis, A., & Leonard, C. (2012). Examining success of communication strategies used by formal caregivers assisting individuals with Alzheimer's disease during an activity of daily living. *J Speech Lang Hear Res*, *55*, 328–342.

[4] Pichora-Fuller, M. K. (2015). Cognitive decline and hearing health care for older adults. *Amer J Audiology, 24*(2), 108-111

[5] Medwetsky, L. (2015). Mobile device apps for people with hearing loss: Expanding the horizons of hearing access. *Hearing Loss Magazine* (Nov/Dec), 26-32.

[6] Gonsalves, C., & Pichora-Fuller, M. K. (2008). The effect of hearing loss and hearing aids on the use of information and communication technologies by community-living older adults. *Canadian J Aging, 27*(02), 145-157.



An information system with academic CV management, expertise inventory and networking capabilities for research institutions and associations.

Un système d'information avec gestion de CV académique, un inventaire de l'expertise interne et des capacités de réseautage pour des organismes de recherche.

With UNIWeb, researchers can:

Streamline
funding applications with Canadian Common CV integration

Reuse
CCV data to generate academic CVs and progress reports

Mobilize
knowledge by creating engaging webpages for research projects

Avec Uniweb, les chercheurs peuvent:

Simplifier
les demandes de financement grâce à l'intégration au CV commun canadien

Réutiliser
les données du CVC pour générer des CV académiques et des rapports de progrès

Mobiliser
les connaissances en créant des pages Web attrayantes pour les projets de recherche

http://uniweb.network

INDIGENOUS SPEECH COMMUNICATION - COMMUNICATION VERBALE INDIGÈNE

The Realisation Of Coronal Obstruents Across Hul'q'umi'num' Speakers
Sonya Bird, Donna Gerdts, Janet Leonard 134

Ultrasound Overlay Videos And Their Application In Indigenous Language Learning And Revitalization
Heather Bliss, Strang Burton, Bryan Gick 136

Shifty Vowels: Variation In Dialectal Lowering In Gitksan
Kyra Fortier 138

Acoustic Characteristics Of Obstruents In Huehuetla Tepehua
Rebekka Puderbaugh 140

Acoustic Characteristics Of Preaspirated Stops In Plains Cree
Katherine Schmirler 142

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 144

THE REALISATION OF OBSTRUENTS ACROSS SPEAKERS OF HUL'Q'UMI'NUM'

Sonya Bird [*1], Donna Gerdts [†2] and Janet Leonard [‡2]
[1]Dept. of Linguistics, University of Victoria, Victoria, BC, Canada
[2]Simon Fraser University, Burnaby, BC, Canada

1 Introduction

The language revitalization movement is growing in Canada, with initiatives currently underway in many communities to help language teachers to become fluent and to deliver various kinds of immersion programs: mentor-apprentice teams, language nests, and immersion camps and classes. Although all of these programs emphasize spoken language, virtually no research has been done on pronunciation in the context of language revitalization, e.g. on how pronunciation varies across speakers and on what challenges learners face in becoming proficient speakers. As a step towards filling this gap, this paper provides a preliminary overview of the pronunciation of a particularly complex set of sounds—coronal obstruents—among learners of Hul'q'umi'num', the Island dialect of Halkomelem Salish (ISO 639-3 hur).

1.1 Background

Hul'q'umi'num' is spoken by around forty first-language speakers living on southern Vancouver Island and neighboring islands. Although it is considered a highly endangered language because so few first-language speakers remain, the language remains strong due to its use in the ceremonial life of the Hul'q'umi'num' people. Many adults speak Hul'q'umi'num' to some degree and many hundreds understand it. People of all generations have become semi-fluent in Hul'q'umi'num' and help to ensure the survival of the language by teaching it to the younger generation.

Since the intergenerational transmission of the language has been interrupted due to decline in number of first-language speakers, some of the burden of teaching the language has shifted over to post-secondary language programs. The opportunity for this research arose when Gerdts helped teach a Simon Fraser University course in Hul'q'umi'num' practical phonetics (four weeks; forty hours total) in July 2016 in Duncan, British Columbia. Various audio and video recordings were made of teachers and students using Hul'q'umi'num' and these materials are being used to help design better teaching materials for learners.

1.2 Research questions

The goal of our project is to determine how learners of different fluency levels realize coronal sounds; in particular, do they maintain the contrasts among the stops /t ṫ/, the fricatives /θ ɬ s š/, and the affricates /t$^{\theta}$ ṫ$^{\theta}$ ƛ c ċ č č̓/? (Americanist phonetic symbols are used in this paper)? The coronals constitute one-third of the Hul'q'umi'num' consonantal phonemic inventory. Also present in the language are /p ṗ k k^{w} k̓^{w} q q q^{w} q̓^{w} ʔ x^{w} x̌ x̌^{w} h m ṁ n ṅ l l̓ y ẏ w ẇ/.

2 Method

During week one of the four-week practical phonetics course, students were given an assessment test, which was used to pinpoint sounds that were proving problematic for learners. For the assessment test, one of the teachers (a fluent first-language speaker and linguist in her eighties) read a list of 48 words to fifteen second-language speakers and learners, one or two students at a time. The list was designed to test the pronunciation of coronals in various environments. The teacher read the words from a list that she could see but the students could not. She said each word once and then the student attempted to repeat the word. Some of the words were familiar to the students and some were not. We recorded the tests and then analyzed them auditorily (through transcription), as a first pass at determining to what extent the second-language speakers' pronunciations deviated from the first-language speaker's.

3 Results

We started from the hypothesis that sounds that are present in the first language of the learners (English) are easier for the Hul'q'umi'num' students to pronounce (Flege, 2003). Indeed, we found that the coronal sounds /t, s, š, č/ were non-problematic: all students were able to repeat these sounds without errors. For other coronals, at least some of the students made errors in their pronunciation in some words.

Table I. Number and percentage of errors, by target sound.

sound	errors/tokens	%
t	4/180	0.2%
ṫ	13/150	8.6%
c	9/75	12%
ċ	14/90	15.5%
ɬ	42/210	20%
θ	28/105	26.6%
ƛ̓	16/75	34.6%
ṫ$^{\theta}$	47/120	39.1%

Table 1 presents the proportion (errors/total tokens) and percentage of words that contained errors for each target

[*] sbird@uvic.ca
[†] gerdts@sfu.ca
[‡] jleonard@uvic.ca

sound (total token number = number of the words on the list x fifteen speakers); the targets sounds are presented from those with the least errors to those with the most.

4 Discussion

4.1 Glottalized sounds

Given the lack of glottalized stops and affricates in English, our hypothesis was that the glottalized coronals would be more difficult to pronounce than their plain corresponding sounds. In the case of alveolar stops /t/ versus /ṫ/, this was the case. The only error made with /t/ was its omission in the coda of the first syllable of the word *ci:tməx*^{w} 'great-horned owl' by four speakers. In contrast, 13 errors were made in words with /ṫ/ (9 /t/ substitutions, 2 /t$^{\theta}$/, 1 /λ̇/ and 1 /l/). However, we found that speakers made more errors in pronouncing the plain alveolar affricate /c/ (11 /ċ/ substitutions, 2 /ṫ/ and 1 /š/) than its glottalized counterpart /ċ/ (5 /c/ substitutions, 1 /ṫ/ and 3 /s/). Eight of the fifteen students, all except the ones that were the most fluent, strengthened /c/ to /ċ/ in word initial position in at least one word, and one student of intermediate fluency used this strategy for all /c/s in word-initial position.

The two typologically rare glottalized affricates, the lateral /λ̇/ and the dental /ṫ$^{\theta}$/ also proved challenging for the students. Of the 25 errors in pronouncing /λ̇/, 15 involved deglottalizing and unpacking the affricate to a sequence of /t/ and /l/. (This pronunciation was judged by the native speaker linguist to not be an impediment to understanding, presumably because there is no non-glottalized counterpart to this sound, and sequences of /t/ and /l/ are quite rare.) Other errors were to pronounce it as /t/ (4), /ṫ/ (4), /ɬ/ (1), and /ṫ$^{\theta}$/ (1). Twelve of the students made an error in at least one word with /λ̇/, and four of the students made errors in 3 or 4 of the 5 target words.

The glottalized dental affricate /ṫ$^{\theta}$/ was the coronal sound that was most often mispronounced in our study. All the students except the most fluent student in the class made errors with this sound. The dental affricate is quite close in pronunciation to the glottalized alveolar affricate /ċ/: 19 of the 45 errors involved substituting /ċ/. Other errors were substituting a glottalized coronal (/ṫ/ (3) and /λ̇/ (3)) or a plain affricate or fricative (/t$^{\theta}$/ (1), / c/ (5), /č/ (1), /θ/ (2), /ɬ/ (1)), or a sequence of sounds (/sṫ$^{\theta}$/, /θs/, /st/). Half of the students consistently used one sound when substituting for /ṫ$^{\theta}$/, while others made a variety of errors, and this did not correlate with the degree of fluency of the student.

4.2 Fricatives

We were surprised by the degree to which the lateral fricative /ɬ/ than the dental fricative /θ/ were confused, especially given that /θ/ (but not /ɬ/) exists in English: 67% of the errors with /ɬ/ involved substituting /θ/, and conversely 73% of the errors with /θ/ involved substituting /ɬ/. Other miscellaneous substitutions for /ɬ/ were /s/, /l/, lɬ/, /x^{w}/, and /x̌^{w}/, and for /θ/ the coronals /s/, /t/, /λ̇/, and /c/ were substituted. Other than in one lexical item (*sqəqəwəθ* 'rabbit'), advanced learners had no difficulty differentiating /ɬ/ and /θ/. Some of the students of intermediate fluency merged the two sounds to /θ/. One intermediate student substituted half of the occurrences of /θ/ with /ɬ/ and half the occurrences of /ɬ/ with /θ/. One lower intermediate student used a variety of substitutions for /ɬ/, but pronounced /θ/ in all examples except one. One beginning student used a variety of substitutions for both /ɬ/ and θ/.

More research is required on the pronunciation of /θ/ in English vs. Hul'q'umi'num', by Hul'q'umi'num' speakers and learners. One possible cause of confusion is that Hul'q'umi'num' learners are replacing a sound that exists in their first language (/θ/) with one that exists only in their second language (/ɬ/). This type of pattern has been observed anecdotally elsewhere; it seems to be a kind of hypercorrection or overcompensation, which results from a certain degree of insecurity on the part of the speaker about the difference between the two sounds (Eckman et al. 2013) and from a desire to sound pronounce in a distinctively non-English way.

5 Conclusion

While preliminary, this study helps delineate areas for further research. Testing perception and production of first and second Hul'q'umi'num' language speakers may lead to strategies for improving pronunciation of /c/, /ṫ$^{\theta}$/, and /ṫ$^{\theta}$/ for all students and /θ/ versus /ɬ/ for beginning students. In the interim, we shared the assessment test with the students, noting problem areas, and then we designed lectures, pronunciation exercises, and tasks (such as memorizing short poems and practicing tongue twisters) to target the problematic sounds. Some improvement was noted in beginning students, even during the short time frame of the course. This points to the benefit of structuring the practical phonetics course around initial student assessments; contributing factors to improvement may be receiving corrections from teachers, improved reading skills, or simply the opportunity to use the language.

Acknowledgments

Thank you to the FLNG 130 students and teachers for participating in this study. Thank you to Ruby Peter, Sti'tum'at, for administering the test and to Heather Harris for helping in recording. Thank you to Vancouver Island University for hosting the course and to SSHRC for supporting this research. Thank you to Murray Munro for helpful advice and to Charles Ulrich for corrections.

References

[1] F. Eckman, G. Iverson, and J. Y. Song. The role of hypercorrection in the acquisition of L2 phonemic contrasts. Second Language Research 29: 257–283, 2013.

[2] J. E. Flege. Assessing constraints on second-language segmental production and perception. In N.O. Schiller & A.S. Meyer (eds), Phonetics and Phonology in Language Comprehension and Production. Berlin: De Gruyter, 2003.

ULTRASOUND OVERLAY VIDEOS AND THEIR APPLICATION IN INDIGENOUS LANGUAGE LEARNING AND REVITALIZATION

Heather Bliss [*1], Strang Burton[†1] and Bryan Gick[‡1,2]
[1]Dept. of Linguistics, University of British Columbia, 2613 West Mall, Vancouver, BC, Canada, V6T1Z4
[2]Haskins Laboratories, 900 George St., New Haven, CT 06511, USA

1 Introduction

In second language (L2) acquisition, learners make use of not only auditory but also visual information to acquire the speech sounds and patterns in a new language. With respect to the latter, ultrasound has been demonstrated to be an effective tool in L2 pronunciation learning for facilitating visualization of the articulatory processes involved in speech production [8, 10, 11, 12]. Safe, non-invasive, and versatile, ultrasound allows learners to directly observe their articulator movements (and compare them with those of an instructor or native speaker), allowing them to make articulatory adjustments to improve their pronunciation.

However, ultrasound-based pronunciation instruction faces challenges as well. With current technology, it is best used in one-on-one contexts or with small groups [8], and interpreting the displays requires specialized knowledge in both articulatory phonetics and ultrasound imagery.

To make ultrasound visualization accessible and interpretable to a broader audience, we developed a series of ultrasound overlay videos which combine ultrasound images of tongue movement in speech with external profile views of a speaker's head [1]. Initially, the video library consisted of 91 videos, corresponding to each sound in the International Phonetic Alphabet (see enunciate.arts.ubc.ca), and it was intended for use primarily by university-level linguistics students and learners of widely studied languages such as Japanese. However, an unexpected outcome of this project has been a broad interest from First Nations communities and allies, who recognized the potential benefits of using this technology in Indigenous language learning and revitalization. In this paper, we report on our work with four First Nations groups to develop customized ultrasound overlay videos for their languages.

2 Methods

2.1 Ultrasound Overlay Video Library

The procedure for developing the original 91 videos was time-intensive. After recording and time-aligning ultrasound and exterior videos, the process of generating the overlay from the raw footage consisted of three steps completed frame-by-frame for each video: erasing all but the tongue image from the ultrasound video, brightening and colouring pink the tongue image, and scaling and overlaying the tongue image on the facial image [1].

[*] hbliss@uvic.ca
[†] strangb1@telus.net
[‡] gick@mail.ubc.ca

2.2 Customization

In response to the interest in customizing ultrasound overlay videos for different linguistic communities, we developed a software package to automate the overlay process. Using time-aligned ultrasound and exterior video clips, the software generates a tongue trace that is scaled, shaded, and superimposed on the face. The image can be manually adjusted frame-by-frame; even with such adjustments, developing overlay videos is much faster and easier with the software than with the previous manual methods.

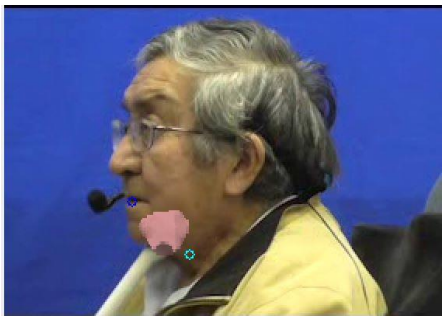


Figure 1: Screenshot from customized ultrasound overlay video of W̱SÁNEĆ elder and SENĆOŦEN speaker Lou Claxton

3 Case Studies

3.1 Halq'eméylem (Coast Salish)

Critically endangered, Halq'eméylem has only one fluent first language (L1) speaker, but a growing number of L2 speakers and learners. Halq'eméylem has a rich consonant inventory, including plain, ejective, and labialized obstruents at various places of articulation. This yields a number of contrasts that are challenging for learners, most of whom are L1 English speakers. These contrasts were the focus of this case study; we recorded the one L1 speaker producing near-minimal pairs of Halq'eméylem words highlighting challenging contrasts; examples are given in Table 1.

Table 1: Examples of Halq'emeylem words recorded

Contrast	Word 1	Word 2
Plain/Ejective	tá:l qwélqwel	t'á:l qw'él
Alveolar/Dental	tale	th'ále
Velar/Uvular	k'ák'elha	q'áq'el

The intention is to use the resulting ultrasound overlay videos in the production of digital storybooks that are used for language teaching purposes.

3.2 Secwepemc (Interior Salish)

Like Halq'eméylem, Secwepemc is critically endangered, has a rich phonetic inventory that poses challenges for learners. The Splatsin Tsm7aksaltn has produced a 'Sounds of Secwepemc' resource that exemplifies the sounds of the language with words from the Eastern dialect [5]. We are developing ultrasound overlay videos to supplement the audio clips for each of these words with one of the L1 speakers in the Splatsin community.

3.3 SENĆOŦEN (Northern Straits Salish)

Although there are only two L1 SENĆOŦEN speakers, there is a vibrant language revitalization program in the W̱SÁNEĆ community, including a language nest and immersion school. Community members regard pronunciation as important, stressing the need to develop "SENĆOŦEN muscles" [3]. This project's aim was to train students in the W̱SÁNEĆ community to create ultrasound overlay videos, thereby building an educational resource for other learners as well as building capacity and expertise for language resource development within the community. Focusing on words that illustrated challenging sounds and clusters, as well as early vocabulary such as numerals, we recorded productions of one L1 speaker, four advanced adult learners, and two children in the immersion program.

3.4 Blackfoot (Plains Algonquian)

This project focused on a particular feature of Blackfoot phonology that varies across dialects: word-final vowels. These vowels are seen as emblematic of language loss, assumed to be absent in the grammars of younger speakers and/or disappearing from certain dialects [4, 6]. For some speakers, these vowels are soundless: articulated but inaudible [7], and for others they are not articulated at all, but are nevertheless phonologically active. As such, the final vowels are challenging for learners. We developed ultrasound overlay videos with speakers of three dialects in order to help learners understand the range of variation in the pronunciation of final vowels.

4 Discussion

The challenges of pronunciation learning can be compounded for Indigenous languages by both a scarcity of resources (including small numbers of speakers) and the pressures faced by heritage learners to preserve their ancestral language in an authentic way [9]. Some community members may be latent speakers, inhibited due to perceived concerns with their pronunciation, particularly in the presence of elders [2]. The development and distribution of customized ultrasound overlay videos for Indigenous languages may help to address these challenges by giving learners a new way to learn pronunciation, and by creating documentation for future generations. The case studies highlight different ways in which ultrasound overlay videos may contribute to Indigenous language revitalization.

5 Future Directions

The next step for these projects is to develop and implement procedures for evaluating the effectiveness of ultrasound overlay videos in Indigenous language learning. As for developing videos for other languages, there is interest from Indigenous communities and allies for languages including Heiltsuk, Hul'q'umi'num', Kwak'wala, and Skwxwu7mesh.

Acknowledgments

Sincere thanks to Fred Breaker, Noreen Breaker, Lou Claxton, Ashlee Cooper, Shirlee Crowshoe, Emmeline Felix, Nick Henry, Tiffany Joseph, Katia Olsen, Elizabeth Phillips for sharing their languages with us. Thank you to Jennifer Abel, Saurabh Garg, Matthew Law, Masaki Noguchi, Noriko Yamane and other researchers in the ISRL for their contributions towards developing the ultrasound overlay procedure, and to Sonya Bird, Inge Genee, Christine Schreyer, Tye Swallow, and Rosalind Williams for helping facilitate community partnerships. This work is supported by an NIH Grant (DC-02717) to Haskins Laboratories, and the UBC Teaching and Learning Enhancement Fund.

References

[1] J. Abel, B. Allen, S. Burton, M. Kazama, M. Noguchi, A. Tsuda, N. Yamane, and B. Gick. Ultrasound-Enhanced Multimodal Approaches to Pronunciation Teaching and Learning. *Canadian Acoustics*, 43(3). 124-125, 2015.

[2] C. Basham and A. Fathman. The latent speaker: Attaining adult fluency in an endangered language. *Intl Journal of Bilingual Education and Bilingualism*, 11: 577-97, 2008.

[3] S. Bird and S. Kell. Pronunciation in the context of language revitalization. Presented at ICLDC 4, 2015.

[4] A. Chatsis, M. Miyashita, and D. Cole. A Documentary Ethnography of a Blackfoot Language Course. *The Persistence of Language*. Benjamins, 257-90, 2013.

[5] FirstVoices. www.firstvoices.com. 2000-13.

[6] D. Frantz. *Blackfoot Grammar*, 2nd ed. U Toronto Press, 2009.

[7] B. Gick, H. Bliss, K. Michelson and B. Radanov. 2012. Articulation without Acoustics: "Soundless" Vowels in Oneida and Blackfoot. *Journal of Phonetics* 40(1):46-53.

[8] B. Gick, B. Bernhardt, P. Bacsfalvi, and I. Wilson. Ultrasound imaging applications in second language acquisition. *Phonology and second language acquisition*. Benjamins: 309-22, 2008.

[9] L Hinton and J. Ahlers. The issue of "authenticity" in California language restoration. *Anthropology & Education Quarterly*, 30: 56-67, 1999.

[10] C. Pillot-Loiseau, T. Kamiyama, and T. Kocjančič Antolík. French /y/-/u/ contrast in Japanese learners with/without ultrasound feedback: vowels, non-words and words. Paper presented ICPS, 2015.

[11] M. Tateishi and S. Winters. Does ultrasound training lead to improved perception of a non-native sound contrast? Paper presented at CLA, 2013.

SHIFTY VOWELS: VARIATION IN DIALECTAL LOWERING IN GITKSAN

Kyra Ann Fortier[1]
Universtity of Victoria

1 Introduction

Gitksan is a Tsimshianic language spoken by the peoples living along the upriver areas of the Skeena river, British Columbia (Rigsby, 1986; Tarpent, 1987). The current body of research has largely identified two broad dialectal categories: 'East' and 'West' (Rigsby, 1986). There are socio-culturally distinct and linguistically variable groups within these categories, including similarity with Nisgha (Tarpent, 1987), therefore this paper will argue is a dialectal continuum.

Gitksan has a set of 5 vowels, {a, e, o, u, i}, further differentiated by a phonemic contrast in length (Rigsby, 1986). Additionally, allophonic alternations between [a] and [ɛ] have been observed such that in certain linguistic environments, where [a] is seen in the Eastern dialect, the Western dialect produces [ɛ] (Rigsby, 1986; Tarpent, 1987). Gitksan uses a rich consonant inventory including post-velar articulations. Consequently, uvular lowering effects have been observed, particularly as it pertains to [a, ɑ] in the Eastern dialect, where the vowel precedes a uvular articulation (Rigsby, 1986). The interaction of these two processes ([a, ɛ] alternation and uvular lowering) creates dialectal differences in the effects of uvular lowering, which is the topic of this paper.

The physiological characteristics of uvular articulation have been shown cross-linguistically to create lowering systems through uvular assimilation, such as this paper observes in Gitksan (Bessell, 1992; Walker & Rose, 2015). Gitksan has three uvular consonants: the voiceless stop [q] (written in the practical orthography as k̲), the voiced stop [G] (written as g̲), and the voiceless fricative [χ] (written as x̲). The effects of uvular lowering on vowels have been measured using spectrograms and formants: F1 (indicative of vowel height) and F2 (indicative of vowel backness) (Ladefoged & Johnson, 2010). Due to the shape of the vocal tract and articulation of low back vowels like [ɑ], both of these measurements are important when discussing this lowering process (Gick, Wilson, & Derrick, 2012).

2 Méthode/Method

2.1 Elicitation

Phase 1

This study has 2 phases. We conducted Phase 1 in the field with 6 participants. 4 participants (3 female, 1 male) identified as Eastern (Gigeenix) speakers, and 2 participants (2 male) identified as Western (Gyeets) speakers. All speakers in both phases were L1. Phase 1 focused exclusively on identifying and characterising patterns of uvular lowering within these dialects. The word list in this phase used 56 tokens, containing several examples of phonologically contrastive vowels (in shape and length) representing the extensive variety of the Gitksan vowel system. We focused most closely on short [a], and long {i:, e:, o:}. Each token was elicited and recorded 3 times. After elicitation this word list was narrowed to a smaller set of minimal and near-minimal pairs that best illustrated patterns of uvular lowering consistent with existing research into this process within neighbouring language families (Bessell, 1992; Walker & Rose, 2015).

Phase 2

The second phase of this study was smaller in scale and more narrow in focus. 3 participants were consulted. Phase 2 was conducted in a lab setting, using the same elicitation techniques as Phase 1. This phase consisted of 1 male participant who identified as a Western (Gyanimx̲) speaker, 1 male participant who identified as a Western (Gyeets) speaker, and one female participant who identified as an Eastern (Gigeenix) speaker. The word list in phase 2 used 36 tokens, demonstrating short [a, ɛ]. The purpose of this phase was to collect data that informed the phonetic components of the [a, ɛ] phonemes of the language as represented in the current body of research, and to determine whether there was sufficient evidence to suggest a systemic vowel shift along a dialectal continuum.

2.2 Phonetic Analysis

For both phases, we measured the tokens in Praat, using spectrograms. We measured the F1 and F2 of all vowel tokens, and in Phase 1 the Eastern and Western averages for both were compared to ascertain dialectal patterns. Where long vowels were present, start and end points were measured and compared to determine slope. Phase 2 measurements were used to plot each speakers vowel space to allow phonetic classification of tokens and interspeaker comparison

3 Résultats/Results

3.1 Phase 1

Phase 1 results are illustrated with the folowing minimal and near minimal pairs:

Table 1 : Phase 1 Token Pairs

Token	Vowel, Environment	Western Formant Average	Eastern Formant Average
ban (belly)	[a], __velar	F1: 1148 F2: 1858	F1: 792 F2: 1561
bax̲ (to run)	[a], __uvular	F1: 657 F2: 1607	F1: 813 F2: 1346

[1] kyrabw93@gmail.com

Token	Vowel, Environment	Western Formant Slope Average	Eastern Formant Slope Average
ts'eex (to be full)	[eː], __velar	F1: -62 F2: 76	F1: 53 F2: 125
ts'eek̲ (to eat fast and a lot)	[eː], __uvular	F1: 95 F2: 18	F1: 538 F2: -317
ha'niigilbilsa (Tuesday)	[iː], __velar	F1: 24 F2: 253	F1: 41 F2: 200
ha'niigoot (to think)	[iː], __uvular	F1: 110 F2: -339	F1: 18 F2: 532
ha'niigoot (to think)	[oː], __velar	F1: 90 F2: 16	F1: 7 F2: 275
ha'niit'ook̲xw	[oː], __uvular	F1: 6 F2: 457	F1: 196 F2: 42

3.2 Phase 2

We first recorded the formant range of each speaker to plot approximate phonetic boundaries within their vowel spaces, and identified phonetically significant tokens :

Table 2: Speaker Formant Ranges

BS, Gigeenix, East, Female	HH, Gyeets, West, Male	VG, Gyanim̲x̲, West, Male
F1: 393 – 887	F1: 231 – 786	F1: 148 – 1247
F2: 1097 – 2786	F2: 1128 – 2291	F2: 702 – 2234

Table 3: Phase 2 Tokens

Token	Vowel, environmen	Speaker Formant Average, Phonetic Classification		
		BS	HH	VG
amksiwaa (driftwood; white	[aa], word final, stressed	F1: 752 F2: 1183 [ɑ]	F1: 617 F2:1191 [a]	F1: 540 F2:702 [æ]
asgi (ugly)	[i], word final, stressed	F1: 393 F2:2786 [i]	F1: 300 F2: 2291 [ɪ]	F1: 391 F2: 2067 [ɪ]
gyat (man)	[a], single closed syllable	F1: 768 F2: 1982 [æ]	F1: 454 F2: 2143 [ɛ]	F1: 409 F2:1806 [ɛ]
hat' (martin)	[a], single closed syllable	F1: 880 F2: 1604 [a]	F1: 757 F2: 2136 [ɛ]	F1: 821 F2: 1767 [ʌ]
hlap (crooked)	[a], single closed syllable	F1: 781 F2: 1502 [a]	F1: 558 F2: 2076 [ɛ]	F1: 601 F2: 1688 [ɛ]
jahl (to lose)	[a], single closed syllable	F1: 791 F2: 1425 [a]	F1: 456 F2: 1982 [ɛ]	F1: 508 F2: 1682 [ɛ]
ya' (spring salmon)	[a], single closed syllable	F1: 851 F2: 1642 [ɑ]	F1: 527 F2: 2072 [ɛ]	F1: 406 F2: 2183 [ɪ]
ye'	[ɛ], single closed syllable	F1: 571 F2: 2537 [ɛ]	F1: 567 F2: 2106 [ɛ]	F1: 399 F2: 2334 [ɪ]

4 Discussion

Previous research has asserted that the underlying vowel of the {a, ɛ} vowel set is [a] for both East and West dialects (Rigsby, 1986; Tarpent, 1987). The observed lowering in Phase 1 from [ɛ] to [a] (Western) and [a] to [ɑ] calls this into question as it has not been accounted for.

Inconsistent lowering effects are observed for the dialectal categories in the long vowels. This suggests there is a complex phonetic interaction with uvular assimilation. This may be influenced by an overall shift in the vowel space between the two dialects, causing unique positioning of the tongue root during uvular articulations, differently affecting lowering patterns. Though not attested with phonemic length contrasts, similar effects lowering effects and vowel shifts have been observed in English dialectology in pre-velar environments (Riebold, 2015).

Evidence for a systemic vowel shift is suggested by the Phase 2 results. Between our 3 speakers, BS has the lowest and most back vowels, HH has the most mid vowels, and VG has the highest vowels. With BS being the Eastern most speaker, HH falling geographically in the middle, and VG being the Western most speaker, we see a correlating geographic pattern. Within the vowel space, I suggest a continuous dialectal shift, moving forward and upward from the low back vowel, as dialectal location correlates move from East to West. I am not asserting directionality or chronology. As with our Phase 1 results, these results have parallels to current views on English dialectology (Prichard, 2015), which also presents a systemic vowel shift in low vowels along a dialectal and geographic continuum.

5 Conclusion

This study is limited most heavily by sample size. A wider range of participants in terms of sex and dialect is ideal. Future research should aim to account for the vowel system holistically, including high short vowels and more long vowel tokens. Additionally there is evidence to suggest a relationship between stress and vowel quality found in the Phase 2 results. A detailed analysis of these components may reveal additional evidence for a systemic vowel shift as suggested by this paper.

Remerciements/Acknowledgments

Thank you to the Gitksan Community for sharing and teaching their language. We thank Barbara Sennott, Vincent Gogag, Hector Hill, Louise Wilson, Fern Wiget, Art Sampson, Art Matthews, and Ray Jones. We thank the UBC Gitlab, Henry Davis, Lisa Matthewson, Michael Schwan, Jason Brown, as well as Sonya Bird. *Ha'miyaa!*

Thank you to the Musqueam people for the use of unceded ancestral territories during our work at UBC.

References

Bessell, N. J. (1992). *TOWARDS A PHONETIC AND PHONOLOGICAL TYPOLOGY OF POST-VELAR ARTICULATION*. University of British Columbia.

Brown, J. (2010). *Gitksan phonotactics*. LINCOM Europa.

Gick, B., Wilson, I., & Derrick, D. (2012). *Articulatory Phonetics*. Wiley.

Ladefoged, P., & Johnson, K. (2010). *A Course in Phonetics*. Cengage Learning.

Prichard, H. (2015). The Great Vowel Shift in the North of England. In R. Hickey (Ed.), *Researching Northern English* (1st ed., pp. 51–70). Amsterdam: John Benjamins Publishing Company. http://doi.org/10.075/veaw.55

Riebold, J. M. (2015). *The Social Distribution of a Regional Change: /æg, ɛg, eg/ in Washington State*. University of Washington.

Rigsby, B. (1986). *Gitksan Grammar*. University of Queensland.

Tarpent, M.-L. (1987). *A Grammar of the Nisgha Language*. Cornell University.

Walker, R., & Rose, S. (2015). Guttural Semi-Transparency. In *Annual Meeting of Phonology* (pp. 1–16). Vancouver, BC.

Yamane-Tanaka, N. (2006). Transguttural Harmony in Gitksan: Its Development and Typological Implications *. In *Proceedings of the 21st NorthWest Linguistics Conference* (pp. 136–152). Vancouver, BC: The University of British Columbia Working Papers in Linguistics.

ACOUSTIC CHARACTERISTICS OF OBSTRUENTS IN HUEHUETLA TEPEHUA

Rebekka Puderbaugh
Department of Linguistics, University of Alberta, Edmonton, Canada T6G 2E7

1 Introduction

The Totonacan languages of Mexico are spoken by approximately 244,000 people [2] in a region comprised of portions of the states of Veracruz, Puebla, and Hidalgo. Much of the documentation on Totonacan focuses on the complex morphosyntax of the language family. Phonological descriptions of Totonacan languages note the prevalence of glottal stop and contrastive laryngealization of vowels and consonants, but little is known about the phonetic realizations of these segments.

A major division has been established in the Totonacan family based largely on morphosyntactic and lexical differences between Totonac to the south and Tepehua to the north. An additional distinction between the two language groups is the incorporation of laryngealization: in Totonac languages laryngealization contrasts are typically realized on vowels, while in Tepehua languages laryngealization contrasts are expressed in the production of the preceding obstruent. The language spoken in Huehuetla, Hidalgo (hereafter HT) belongs to the Tepehua branch of the Totonacan family and exhibits a laryngealization contrast in stops at three places of articulation: /p pʾ t tʾ k kʾ/, and in affricates at two places of articulation:/ts tsʾ tʃ tʃʾ/. This laryngealization contrast (also referred to as "glottalization") has been transcribed phonemically with the notation of the International Phonetic Alphabet indicating sounds produced with an egressive glottalic airstream, commonly referred to as ejectives. However, impressionistic descriptions of these segments indicate that only the velar segment /kʾ/ seems to fit this description. The sounds transcribed as /pʾ/ and /tʾ/ have instead been described as "phonetically closer to the implosives [ɓ] and [ɗ], respectively" [3]. Traditional phonetic definitions of implosives imply some period of voicing due to incomplete closure of the vocal folds while the larynx is being lowered, but both the pulmonic and laryngealized stop series in HT are described as phonemically voiceless. In contrast to the alveolar stop /tʾ/, both /tsʾ/ and /tʃʾ/ are described as ejectives produced with an egressive airstream.

The present paper will provide some preliminary impressions of closure duration and voice onset time of stops and affricates in HT. Duration has long been known to be a salient cue to place of articulation, manner, and voicing in many speech segments. This study is part of a larger project investigating the acoustic realization of laryngealization in Totonacan, a characteristic that has not yet received satisfactory analysis within Totonacan phonology. This pilot study contributes to a basic phonetic description of obstruents in HT, which will allow for broader crosslinguistic comparison.

puderbau@ualberta.ca

2 Methods

2.1 Speakers

Acoustic data from three speakers, two women and one man, are reported here. The speakers were all between the ages of 50 and 60 years old lifelong speakers of HT.

2.2 Materials and Procedures

The wordlist consisted of 116 words previously documented in a formal grammar of HT [3]. The list was designed to highlight contrasts between laryngeal configuration or airstream (pulmonic/glottalic), place (bilabial, alveolar, velar), and manner (stop, affricate), as indicated in the practical orthography. Target segments were elicited within a frame sentence, such as *juu chawai* ____ *iknajun* /huː tʃawai ____ iknahun/ 'Now I say ____'. Each target word was repeated in the frame three times. Due to the unfamiliar nature of the task, speakers sometimes altered the frame to place the target at the end instead of the middle of the sentence. Because HT is not commonly written, speakers were prompted orally in English and HT with the assistance of one of the speakers, who sometimes offered oral explanation in HT of which term was intended.

Digital recordings were made with a head mounted microphone and a Marantz portable digital recorder (PMD 660) at a sample rate of 44 kHz. The recordings were segmented by hand in Praat [1]. Praat scripts were used to extract acoustic information from the annotated recordings. Graphical analyses were undertaken in R [4].

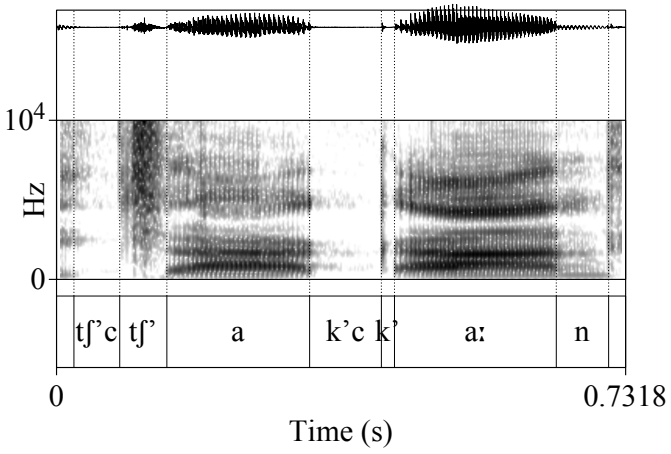


Figure 1. Example spectrogram from female speaker J60 with segmental annotation, *chʾakʾaan* 'armadillo'.

2.3 Acoustic segmentation

Closure duration and voice onset time (VOT) were taken of both stop and affricate segments. Figure 1 illustrates the placement of segment boundaries. The beginnings of closure periods were taken as the end of periodic vibration in the preceding vowel (or nasal in cases where the frame sentence was modified by the speaker). The closure period ended with the stop release, which was concurrent with the beginning of the VOT segment. In the case of affricates, this post-release segment includes both the stop burst and the following frication. This segment ended with the resumption of periodic vibration in the following vowel. No instances of prevoicing during the closure period were observed, supporting previous descriptions of HT stops as lacking a voicing distinction, but in some cases there was no obvious burst upon release of the closure. In such cases, a VOT value of zero was recorded. Duration values were log normalized before undergoing graphical analysis.

3 Results

Stop duration

Figure 2 summarizes the log normalized closure and VOT duration data across all three places of articulation and both laryngeal categories. Results of a cursory linear model with place of articulation and laryngeal setting as predictors show significant effects of place of articulation and laryngeal setting. An interaction between place and laryngeal setting was not found to improve the model. Bilabial closures tended to be longer than either alveolar or velar closures across laryngeal settings. Pulmonic stops were overall slightly shorter than laryngealized stops.

Affricate duration

Figure 3 summarizes the log normalized closure and release portions of affricates produced in both laryngeal settings. Unlike the stops, affricates have somewhat shorter closure periods, followed by a longer release portion. This is to be expected because of the sustained nature of fricative noise. A rather striking aspect of the duration data, however, is the stark contrast between pulmonic and laryngealized affricate release periods. Pulmonic affricates have release durations of roughly 80-90ms, while laryngealized affricate release periods last only about 55ms.

4 Discussion and Conclusion

Duration has long been established as a salient cue to stop categorization across languages. The Tepehua data aligns with previous reports that the further back the place of articulation, the longer VOT duration will be. Likewise, it was also found that stops articulated at more front places of articulation have longer closure durations. However, there does not seem to be a strong effect of laryngealization in word initial stop segments. On the other hand, the duration differences between pulmonic and laryngealized affricates are striking.

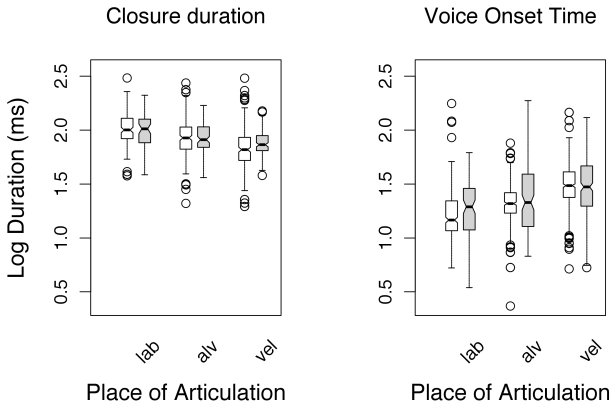


Figure 2. Closure duration and VOT of pulmonic (white) and laryngealized (gray) stops, reported in log milliseconds.

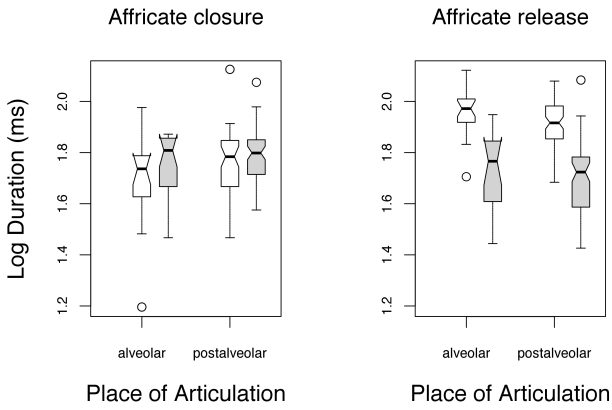


Figure 3: Closure duration and VOT of pulmonic (white) and laryngealized (gray) affricates, reported in log milliseconds.

Acknowledgments

The author thanks the speakers of Huehuetla for their generosity, hospitality, and the invaluable contribution of their voices. This research was supported by a grant from the Jacobs Research Funds at the Whatcom Museum in Bellingham, WA.

References

[1] Boersma, Paul & Weenink, David (2016). Praat: doing phonetics by computer [Computer program]. Version 6.0.19, retrieved 13 June 2016 from http://www.praat.org/

[2] Instituto Nacional de Estadística, Geografía e Informática (INEGI). 2010 Censo de Población y Vivienda 2010. Veracruz, México.

[3] Kung, Susan Smythe. (2009). A Descriptive Grammar of Huehuetla Tepehua.

[4] R Core Team (2016). R: A language and environment for statistical computing. R Foundation for Statistical Computing, Vienna, Austria. URL https://www.R-project.org/.

ACOUSTIC CHARACTERISTICS OF PREASPIRATED STOPS IN PLAINS CREE

Katherine Schmirler *
University of Alberta

1 Introduction

In this paper, I investigate the acoustic characteristics of preaspirated stops in Plains Cree, an Algonquian language of Canada. Plains Cree is the westernmost member of the Cree-Montagnais-Naskapi dialect continuum that stretches from Alberta to Quebec. Existing descriptions of Plains Cree focus on standard and often archaic phonetic variants and rarely contain phonetic analysis. Similarly, teaching materials put little emphasis on Cree pronunciation beyond individual characters. Here, I focus on a small set of data drawn from Plains Cree spoken in Maskwacîs, Alberta and describe the occurrence of preaspiration, its durational properties, and durational properties of preceding vowels.

1.1 Plains Cree sound system

The consonants and vowels of Plains Cree are given in Table 1; the Standard Roman Orthography is used herein. Stops are unaspirated and *c* is an affricate which can range from [t͡s] to [t͡ʃ]. All consonants occur in onsets and codas. In onsets, stop/nasal-*w* clusters occur. For vowels, a length distinction is conventionally described; long vowels are tense while short vowels are lax, e.g., *î* [iː] and *i* [ɪ]; *ô* [oː (~ uː)] and *o* [ʊ] [1], [2]. In *VhC* sequences these distinctions may be lost (e.g., both *īhc* and *ihc* occur as [ih t͡s]) [1], [3].

Table 1: The sounds of Plains Cree

Consonants						Vowels					
p	t			k							
		c				î	i				
		s			h	ê				ô	o
m	n							â	a		
w			y								

Preaspiration in Cree

In Cree, *h* occurs in onsets and codas and in codas before any other consonant, though *hs* and *h*-nasal sequences occur only across morpheme boundaries. "Preaspirated" is used to refer to sounds which *h* can precede word-finally: *p*, *t*, *k*, and *c*. The affricate *c* is included as a stop due to this patterning [1]. Changes in preaspiration in Plains Cree are not well-documented, though ongoing changes in Woods Cree in northern Saskatchewan have been investigated. Woods Cree speakers tend to simplify *hk* and *hp* to *h* word-finally but not word-internally, with little change in *ht* and *hc* clusters [4]. In Plains Cree, word-final *hk* may simplify to *h* in some areas (A. Wolvengrey, pers. comm.).

1.2 Cross-linguistic preaspiration

Preaspiration is rare among the world's languages and may be used in different ways [5]. For example, Scottish Gaelic uses non-phonemic preaspiration to indicate voiceless rather than voiced stops. This has influenced nearby English dialects, where voiceless consonants are preaspirated [6].

Icelandic distinguishes between plain, postaspirated, and preaspirated stops. Experiments have found that preaspiration can be cued by voice offset time (VOffT), a syllable-final counterpart of VOT. Unlike Cree, Icelandic preaspiration only follows short vowels; longer vowel durations require longer VOffT [7]. VOffT, vowel duration, and vowel length considered in the following analyses.

2 Method

Data used herein was drawn from recordings collected in an ongoing collaboration between the Alberta Language Technology Lab at University of Alberta and Miyo Wahkohtowin Education in Maskwacîs, Alberta.

2.1 Recordings

Careful pronunciations of Cree words were collected using Countryman E6 unidirectional head-mounted microphones and a Roland OCTA-CAPTURE external sound card at a sampling rate of 44.1kHz and bit depth of 16 onto a computer using either Audacity or Adobe Audition. The files were stored in an uncompressed .wav format. Data from 8 native speakers Plains Cree (6 female), aged 50 to 80 years, were used in this study. Fluency levels (1 to 3) were assessed by the author on the basis of fieldwork sessions and speaker self-assessments.

2.2 Data collection

Words containing phonemic preaspirated stops, determined on the basis of the Standard Roman Orthography [3], were extracted for each of the stops (*p*, *t*, *k*, *c*) preceded by each vowel, as available. For each word, recordings from three speakers were included. Word-internal preaspirated stops were the focus of this study, though some word-final instances were included due to a small data set. Stress, speech rate, and position were not analysed at this time. Table 2 gives counts for available sequences ($N = 144$).

Table 2: Counts for *VhC* sequences

	â	a	ê	î	i	ô	o
hp	3	0	0	0	6	0	0
ht	9	6	6	6	6	0	3
hk	15	12	6	0	12	6	12
hc	0	12	6	0	12	0	6

2.3 Measurements

Praat was used for acoustic measurements [8]. Preaspiration was measured as VOffT (ms), a devoiced period of the

* schmirle@ualberta.ca

vowel before the stop closure [7]. The consonant and the non-occurrence of phonemic preaspiration were included, for comparison to ongoing changes in Woods Cree [4]. Vowel duration and phonemic length were also included.

3 Results and discussion

Due to the small data set, the regression models were kept simple. Further research will allow for more in-depth analyses. For logistic regression models, *īhC* and *Vhp* sequences were removed as they occurred categorically with expected preaspiration. Analyses were performed in R using the package lme4 [9], [10].

3.1 Preaspiration

Preaspiration, measured by the presence of VOffT, occurred in 87% of tokens. All instances of *hp* (n = 9) occurred with VOffT, consistent with Woods Cree findings [4]. For *ht*, *hk*, and *hc*, there were no reliable predictors for the occurrence of VOffT. Fixed effects included in logistic regression models were consonant, vowel, vowel length, and fluency, with word and speaker as random effects. No significant effects were found. However, the 87% occurrence rate of preaspiration suggests a higher degree of retention in this Plains dialect compared to Woods Cree [4]. This 87% of the data was used for the following analyses of VOffT.

The mean duration of VOffT was 74.82 ms (SD = 26.56). Table 3 gives the descriptive statistics for the VOffT (ms) by phonemic vowel length.

Table 3: Preaspiration duration by phonemic vowel length

Vowel Length	Mean	SD	Min	Max
long	67.55	26.86	24.41	136.82
short	79.35	25.50	33.34	147.32

A linear mixed effects regression model was fit with VOffT as the response variable, word as a random effect, and vowel duration and length as fixed effects, with an interaction between them. Vowel length was a significant predictor (b = 43.07, t(118) = 2.41, p = .0177). Long vowels occur with shorter VOffTs and short vowels with longer VOffTs. The interaction was also significant (b = -.298, t(121) = -2.03, p = .0449); as long vowels became longer, the VOffT period became longer, while as short vowels became longer, the VOffT period became shorter. This may suggest a preferred *Vh* duration relative to vowel length, perhaps to maintain the length distinction. This is similar to Icelandic findings, where longer *Vh* durations required longer VOffTs for preaspiration to be perceived [7]. However, Plains Cree allows preaspiration after long and short vowels which is further reflected in the significant vowel duration/length interaction. An analysis of the *h*/*Vh* ratio may clarify this, following Pind [7] for Icelandic. Comparisons to other *hs* and *h*-sonorant clusters will also allow for more detailed description.

3.2 Vowel duration

A linear mixed effects regression model was fit with vowel duration as the response variable, word and speaker as random effects, and the occurrence of VOffT and vowel length as fixed effects. Phonemic vowel length was a significant predictor (b = -36.64, t(69) = -4.62, p < .001); long vowels occurred with a mean duration of 136.60 ms (SD = 46.17) and short vowels with a mean duration of 103.49 ms (SD = 28.43). These results align with previous phonological and phonetic descriptions of Cree, e.g. [1], [2]. However, there was no significant effect for the occurrence of preaspiration. Thus, the previously-described vowel length neutralisation in *VhC* sequence is not seen in these Maskwacîs Cree data, though further data collection and analysis is necessary to confirm this. Detailed comparison with vowels followed by plain consonants, and between different vowel qualities, may also offer insight.

4 Conclusion

The Plains Cree speakers in this pilot study have shown relatively little loss of preaspiration compared to Woods Cree findings. The data analysed here have suggested some important predictors for VOffT and the duration of the preceding vowel in Plains Cree, though the data for more in-depth analyses are not yet available.

Additional data collection in different regions and communities, and the inclusion of speech rate, stress, and medial/final position in analyses, will allow for fuller descriptions of Plains Cree preaspiration and variation across dialects. Detailed descriptions of modern Cree use may benefit researchers, teachers, and students alike, and in turn facilitate the development of pedagogical materials.

Acknowledgments

I am indebted to my consultants in Maskwacîs and my colleagues at the Alberta Language Technology Lab for their work in elicitation, transcription, and data processing.

References

[1] H. Christoph Wolfart. Sketch of Cree, an Algonquian Language, in *Handbook of American Indians. Volume 17: Languages*. Washington: Smithsonian Institute: 390, 1996.

[2] Jeffrey Muehlbauer. Vowel spaces in Plains Cree. *J. Int. Phon. Assoc.*, 42(1):91, 2012.

[3] Arok Wolvengrey. *Cree: Words*. Bilingual edition. Regina: University of Regina Press, 2011.

[4] D. Starks and E. Ballard. A short note on changes in pre-aspiration in Woods Cree. *Linguistica Atlantica*, 24:1, 2003.

[5] Daniel Silverman. On the rarity of pre-aspirated stops. *Journal of Linguistics*, 39(3):575, 2003.

[6] Patrick C. Douaud. Pre-Consonantal Aspiration in a Celtic and an Algonquian Language. *Anthropos*, 86(1):185, 1991.

[7] Jörgen Pind. Constancy and normalization in the perception of Voice Offset Time as a cue for preaspiration. *Acta Psychologica*, 89(1):53, 1995.

[8] P. Boersma and D. Weenink. Praat: doing phonetics by computer [Computer program]. Version 6.0.14, retrieved 24 February 2016 from *http://www.praat.org*, 2016.

[9] R Core Team. *R: A language and environment for statistical computing*. Vienna, Austria: R Foundation for Statistical Computing. Retrieved from *https://www.R-project.org*, 2015.

[10] D. Bates, M. Mächler, B. Bolker, and S. Walker. Fitting Linear Mixed-Effects Models Using lme4. *Journal of Statistical Software*, 67(1):1, 2015.

Articulatory Conflict Resolution Strategies Among L1 And L2 Sencoten Speakers

Sonya Bird

This paper documents variation in the strategies used by SENCOTEN speakers of different generations and fluency levels to overcome articulatory conflicts, i.e., sequences of sounds that are difficult to pronounce because they require conflicting configurations of the articulators (Gick & Wilson, 2006). Previous work on SENCOTEN /qi/ and /iq/ sequences (Bird & Leonard, 2006; Bird, 2012) is based on two fluent L1 speakers, and shows that three main strategies are used: 1) compromise of the vowel (/iq/ > [Iq]); 2) insertion of a transitional element, often a fricative (/iq/ > [ixq]); and 3) dynamic tongue movement during the /q/ closure (/iq/ > [ikq]). Crucially, all of these strategies maintain some acoustic evidence of the uvular /q/ closure. One of the concerns among the SENCOTEN-speaking community is that the velar uvular contrast is being lost in younger (L2) speakers (Bird & Kell, 2015). If this is the case, it is likely that articulatory conflict resolution strategies would reflect this, for example /iq/ > [ik]. To explore this possibility, a phonetic study was conducted on two specific sequences: /iq/ and /sq/; both of these require moving quickly between a high, advanced tongue position and a (relatively low) retracted position. Target words containing these sequences (e.g. /sqaxe7 'dog'; /st'iqel/ 'bog') were recorded by 12 speakers varying in generation and fluency level. Preliminary results show that (younger) L2 speakers do indeed tend to pronounce /q/ as [k] in these sequences, whereas their elders (L1 speakers) use a variety of strategies that, for the most part, maintain /q/. Findings offer insight into the strategies used to ease pronunciation among L2 learners, and also give us valuable direction in terms of teaching and learning SENCOTEN.

MULTISENSORY AND SENSORIMOTOR INTEGRATION - INTÉGRATION MULTISENSORIELLE

Sensorimotor Integration Of Vestibular, Proprioceptive, And Visual Information About Head Motion In The Interpretation Of Dynamic Auditory Cues For Front/back Sound Localization
Ewan A Macpherson, Janet Kim 146

Using Spatialized Sound To Enhance Self-Motion Perception In Virtual Environments And Beyond: Auditory And Multi-Modal Contributions
Bernhard E. Riecke 148

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 150

SENSORIMOTOR INTEGRATION OF VESTIBULAR, PROPRIOCEPTIVE, AND VISUAL INFORMATION ABOUT HEAD MOTION IN THE INTERPRETATION OF DYNAMIC AUDITORY CUES FOR FRONT/BACK SOUND LOCALIZATION

Ewan A. Macpherson [*1,2] **and Janet Kim** [†1,2]
[1] School of Communication Sciences and Disorders, Western University, 1201 Western Rd, London, ON N6G 1H1
[2] National Centre for Audiology, Western University, 1201 Western Rd, London, ON N6G 1H1

1 Introduction

Accurate perception of the location of a sound source is greatly facilitated by listener head rotation, which generates dynamic interaural time- and level-difference cues that differ between front- and rear-hemisphere sources. Those cues, however, specify front/back location only when coupled with information about the head motion that produced them. Potential sources of head-motion information for the auditory system are the vestibular, proprioceptive, and visual systems. To investigate the influence of these extra-auditory modalities on the interpretation of dynamic acoustic cues, we have examined dynamic localization performance using an oscillating chair apparatus that permits dissociation of head-on-body and head-in-space motion in combination with a pair of left/right-reversing prism glasses that provide incongruent visual motion information.

2 Methods

2.1 Study design and participants

The data reported here were obtained under various static and dynamic listening conditions. In each, in a within-subjects design, we measured listeners' ability to discriminate between front- and rear-hemisphere low-frequency noise targets while individual or multiple sources of head-movement information were minimized or eliminated and while vestibular and proprioceptive or vestibular and visual head-motion cues were placed in opposition. In total, 12 normally hearing adult listeners participated (6 females, 6 males, age = 23-49).

2.2 Stimuli

Sound stimuli were bursts of low-frequency noise (0.5-1 kHz) with 5-ms raised cosine onset and offset ramps. Low-frequency stimuli were used, as it has been demonstrated that dynamic cues are more salient for low-frequency stimuli, which typically cannot be localized accurately without head motion [1,2]. Bursts were presented at a mean level of 70 dB SPL and trial-by-trial level roving (± 5 dB) was applied. They were presented over attenuating insert earphones in virtual auditory space using individually measured head-related transfer functions (HRTFs). In dynamic conditions, HRTF filters were updated and interpolated (using a TDT RX6 processor) at 60 Hz based on tracking of head and body orientation, as described below. On each trial, a stimulus was presented from one of three front (azimuth 0° or ±22.5°) or three rear (azimuth 180° or ±157.5°) locations in the horizontal plane. The bursts were gated by source, head-in-space, or head-on-body position as described below.

2.3 General procedure

In most conditions, participants were seated in an open-fronted wooden box/chair mounted on a motorized apparatus that, when activated, continuously oscillated the box ±45°about its vertical axis with a peak velocity of 50°/s. In some conditions, the participant's legs and body were immobilized with straps and foam with the head free to move, whereas in others the head was also immobilized. In all but the vestibular/visual opposition condition, the participants were blindfolded to remove visual motion cues. Circumaural hearing protectors were placed over the insert headphones to further reduce the sensation level of the motor noise.

Two sensors from a Polhemus FASTRAK system were used to measure head-in-space and body-in-space orientation, and the difference between those measurements was used to derive a head-on-body angle when required.

On each trial in dynamic conditions, as the listener oscillated their head or were oscillated in the box, stimuli were gated on and off as the appropriate position angle (head-in-space, or head-on-body) passed through a *spatial window* centred on 0° and with a width of 2.6, 5, 10, 20, or 40°. Wider windows provided greater access to dynamic localization cues. After hearing the stimulus, the participant indicated whether the source appeared to be in the front or the rear hemisphere by using a hand-held button box. In each motion condition, multiple trials were presented in pseudo-random order for each combination of source position, motion direction (leftwards or rightwards), and spatial window width. The proportion of correct front/rear judgements was computed for each spatial window width.

2.4 Listening conditions

The front/rear discrimination task was performed under the following listening conditions:

Active head rotation. With the motor deactivated and the head free, the participant actively oscillated the head from side-to-side. Trials in which the peak head velocity deviated by more than 15°/s from 50°/s were repeated. Stimuli were gated by head-in-space angle. In this condition the participant

[*] ewan.macpherson@nca.uwo.ca
[†] janetkk89@gmail.com

has access to vestibular and proprioceptive information about head movement.

Passive rotation. With the motor activated and the head immobilized, the participant was passively oscillated from side-to-side. Stimuli were gated by head-in-space angle. In this condition proprioceptive information was minimized and the participant had access primarily to vestibular information about head movement.

Counter-rotation. With the motor activated and the head immobilized, the participant's body was passively oscillated from side-to-side while they used their neck muscles to keep the head touching two fixed reference points. Stimuli were gated, and dynamic HRTF synthesis was driven, by head-on-body angle. In this condition vestibular information was minimized and the dynamic cues were consistent with proprioceptive information about head movement.

Static listening. 200-ms stimuli were presented without head, body, or source motion. This condition was included to demonstrate the difficulty of front/rear localization of the low-frequency noise stimuli in the absence of dynamic cues and to provide a baseline for assessing head-motion benefit.

"Dynastatic" listening. The participant sat without motion in a normal chair while the stimuli were gated, and dynamic HRTF synthesis was driven, by the orientation of a sensor placed on the activated (but empty) oscillating chair. This condition was included to demonstrate the difficulty of front/rear localization of the low-frequency noise stimuli in the presence of dynamic cues but in the absence of information about head motion.

Hemi-counter-rotation. To assess the relative dominance of vestibular and proprioceptive head-motion cues, they were placed in opposition. This condition was similar to the *counter-rotation* condition, but the support for the head-reference points was geared such that it rotated in the same direction as the chair but at ~ ½ the amplitude. Head-in-space and head-on-body angles were therefore opposite and approximately equal throughout the oscillation with peak velocities of 25°/s. HRTF synthesis and stimulus gating were driven by head-on-body angle.

Left/right reversed vision. To assess the relative dominance of vestibular and visual head-motion cues, the *passive* condition was repeated with the room lit and the blindfold removed. This was done once with normal visual input (congruent vestibular and visual information) and once while the participant wore left/right reversing prism goggles (opposed vestibular and visual information). Spatial windows of 10° and 20° were used.

3 Results

In Figure 1, performance in the *static* condition was near chance, but improved similarly with increasing spatial window width in the *active* and *passive* rotation conditions, in which vestibular information was available. Source motion (*dynastatic*) or isolated head-on-body motion (*counter-rotation*) did not provide a benefit. In the cue-opposition conditions (Figure 2), dynamic cues were typically interpreted based on the vestibular head-in-space motion information.

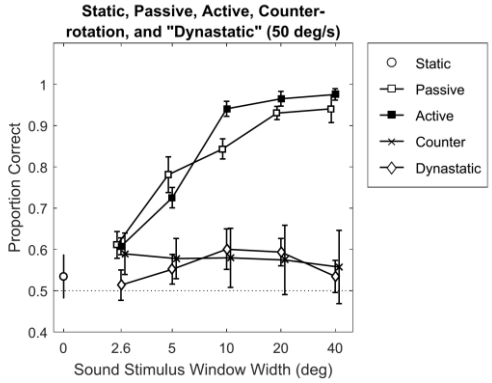


Figure 1: Mean front/rear performance for six participants in the motion-cue deletion conditions. Bars show standard errors.

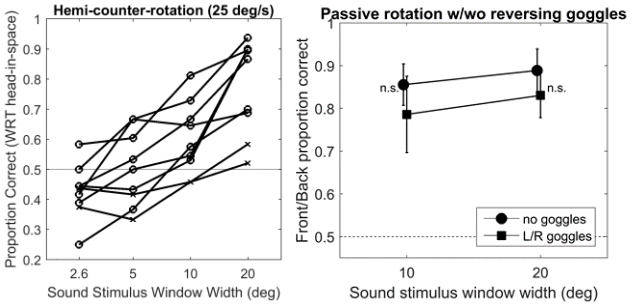


Figure 2: Left: Front/rear discrimination for nine participants in the vestibular/proprioceptive opposition condition. **Right:** Mean performance for five participants in the vestibular/visual opposition condition; no significant effect of reversed vision was observed.

4 Summary and conclusion

The results indicate that only vestibular head-motion information is necessary, sufficient and dominant in the interpretation of dynamic auditory cues for front/back sound localization, as summarized in Table 1.

Table 1:

Head-motion cue	Necessary	Sufficient	Dominant
Vestibular	✓	✓	✓
Proprioceptive	×	×	×
Visual	×	?	×

Acknowledgments

This work was supported by funding from the National Science Foundation (USA) and NSERC (Canada).

References

[1] S. Perrett and W. Noble. The contribution of head motion cues to localization of low-pass noise, *Percept. Psychophys*., 59:1018-1026, 1997.

[2] E.A. Macpherson. Head motion, spectral cues, and Wallach's 'Principle of least displacement' in sound localization. In Y. Suzuki, D. Brungart, et al. (Eds.), *Principles and applications of spatial hearing* (pp. 103-120). World Scientific Publishing, 2011.

USING SPATIALIZED SOUND TO ENHANCE SELF-MOTION PERCEPTION IN VIRTUAL ENVIRONMENTS AND BEYOND: AUDITORY AND MULTI-MODAL CONTRIBUTIONS

Bernhard E. Riecke[*1]
[1]iSpace Lab, School of Interactive Arts and Technology (SIAT), Simon Fraser University, Surrey, BC, Canada

1 Introduction

Embodied self-motion illusions ("vection") have long fascinated both researchers and laypeople. With the increasing quality and affordability of immersive virtual reality (VR) and tele-operation/tele-robotics interfaces, there is also increasing interest in providing compelling sensations of self-motions to create more life-like and convincing experiences [4, 13, 21]. Whereas most research on self-motion perception focuses on visual and vestibular contributions, auditory input can also play a relevant role. Here, we will provide an overview of research indicating how spatialized sound (moving sound fields) can both induce self-motion illusions in blindfolded listeners and enhance self-motion illusions induced by other modalities.

2 Auditory vection

Although vection research has traditionally focused on self-motion illusions induced by moving visual cues, it has been known for more than a century that blindfolded stationary listeners can also experience illusory sensations of self-motion (auditory vection) when listening to moving sound sources [23, 24]. Visually-induced vection can be experienced by virtually everybody and can be compelling to a point where observers reportedly cannot distinguish it from physical self-motion [1, 2, 11]. Yet, auditory vection tends to be much weaker and is only reported in 20-80% of blindfolded listeners [25], which might explain why it has received less attention in research and applications. Similar to visually-induced vection, auditory vection can be perceived for both rotational self-motion (e.g., circular vection around the earth-vertical axis), and for translational self-motions (linear vection) such as forward/backward motions.

Various stimulus characteristics have been shown to enhance auditory vection, resulting in earlier vection onset and increased vection intensity or convincingness ratings. Such vection-facilitating factors include increasing the number of moving sound sources [19, 27] and increasing the velocity of moving sound sources [7, 19, 27], although there seems to be an optimal stimulus velocity (around 60°/s for yaw circular vection) beyond which faster stimulus motion does not further enhance vection [5, 7]. While a minimum quality of sound spatialization is necessary to induce auditory vection, increasing sound spatialization or rendering quality further by using individualized (as compared to generic) head-related transfer functions or binaural recordings does not necessarily enhance vection further [12, 27].

* ber1@sfu.ca

Besides physical stimulus characteristics, the meaning and interpretation of the moving sound sources can also matter. For example, sound sources associated with stationary "acoustic landmarks" such as church bells or fountain sounds were shown to be more effective in inducing auditory vection than sound sources associated with moving objects (e.g., vehicle sounds) or neutral sounds such as pink noise [8, 19, 26, 28]. Even a non-spatialized sound can induce auditory vection if it is perceived to have an inherent motion direction: Blindfolded listening to an ascending Shepard-Risset Glissando (i.e., a continuously rising pitch) resulted predominately in upward (elevator) vection, while descending glissandos elicited downwards vection [9]. One's own belief and perception of whether or not actual self-motion is possible can also modulate auditory vection [12]; it is likely one of the reasons why auditory vection researchers often seat listeners on rotation platforms, movement carts, or other platforms, suggesting that actual self-motion might be possible.

3 Multi-modal contributions of moving spatialized sound sources

Despite the fact that visual cues are typically much more effective in inducing vection than auditory cues, spatialized auditory cues that move in sync with a rotating visual stimulus can nevertheless be used to significantly enhance visually-induced circular vection [5, 6, 16, 20]. Similarly, adding rotating sound sources can also enhance "biomechanical" circular vection induced by blindfolded listeners stepping along a rotating floor platform above which they are seated stationary [17, 18]. Conversely, listening to stationary spatialized sound can reduce biomechanical vection [18]. That is, spatialized sound can both enhance circular vection induced by another modality (if consistent with that modality) and interfere with it (if inconsistent).

4 Multi-modal contributions of non-spatialized sounds

Even if auditory cues are not spatialized, they can under certain conditions enhance visually-induced vection if they metaphorically match the visually-presented self-motion. For example, non-spatialized sounds increasing or decreasing in pitch facilitated visually-induced upward and downward (elevator) vection, respectively, but did not affect visually-induced vection in other motion directions [22].

However, sound decreasing in volume (as if moving away) showed no clear effect on visually-induced vection, even though sound increasing in volume (as if moving closer) facilitated visually-induced forward vection [22]. Note that

such metaphorical or higher-level/cognitive contributions might be prone to experimental demand characteristics as discussed in more detail in [10, 15].

5 Conclusions

Although auditory cues alone provide a much less compelling self-motion sensation than visual cues or biomechanical cues (from walking on a circular treadmill), they can significantly enhance vection induced by other modalities as well as enhance presence and immersion in virtual environments [3, 14, 16, 27]. Furthermore, they can provide omnidirectional cues beyond the limitation of the visual field of view or visual occlusion. This makes spatial sound a promising candidate for further enhancing numerous self-motion simulation applications ranging from immersive virtual reality to entertainment, movies, and tele-presence/tele-robotics; All the more so because high quality spatialized sound can be provided at relatively low cost, using either multi-speaker setups or headphones with HRTF convolution or binaural recordings. In fact, we posit that auditory virtual reality might be the best (and only true) virtual reality we can currently provide. High-quality sound renderings or recordings can not only be indistinguishable from real-world stimuli, but can also be provided by a "transparent interface". That is, while we tend to be aware we are looking at a display presenting visual VR cues, we can easily be unaware of the sound-producing device. In this sense, the ultimate VR system might not be visual, but auditory.

References

[1] Berthoz, A. et al. 1975. Perception of linear horizontal self-motion induced by peripheral vision (linearvection) - basic characteristics and visual-vestibular interactions. *Experimental Brain Research*. 23, 5 (1975), 471–489.

[2] Brandt, T. et al. 1971. Optisch induzierte Pseudocoriolis-Effekte und Circularvektion. *Archiv für Psychiatrie und Nervenkrankheiten*. 214, 4 (Dec. 1971), 365–389.

[3] Hendrix, C. and Barfield, W. 1996. The sense of presence within auditory virtual environments. *Presence - Teleoperators and Virtual Environments*. 5, 3 (1996), 290–301.

[4] Hettinger, L.J. et al. 2014. Illusory Self-motion in Virtual Environments. *Handbook of Virtual Environments*. K.S. Hale and K.M. Stanney, eds. CRC Press. 435 – 466.

[5] Keshavarz, B. et al. 2013. Combined effects of auditory and visual cues on the perception of vection. *Experimental Brain Research*. (2013).

[6] Keshavarz, B. et al. 2014. Demonstrating the Potential for Dynamic Auditory Stimulation to Contribute to Motion Sickness. *PLoS ONE*. 9, 7 (Jul. 2014), e101016.

[7] Lackner, J.R. 1977. Induction of Illusory Self-Rotation and Nystagmus by a Rotating Sound-Field. *Aviation Space and Environmental Medicine*. 48, 2 (1977), 129–131.

[8] Larsson, P. et al. 2004. Perception of Self-motion and Presence in Auditory Virtual Environments. *7th Workshop of Presence* (Valencia, Spain, 2004), 252–258.

[9] Palmisano, S. et al. (in preparation). Auditory vection induced by Shepard-Risset-Glissando stimuli.

[10] Palmisano, S. et al. 2015. Future Challenges for Vection Research: Definitions, Functional Significance, Measures and Neural Bases. *Frontiers in Psychology*. 6, 193 (2015).

[11] Palmisano, S. and Gillam, B. 1998. Stimulus eccentricity and spatial frequency interact to determine circular vection. *Perception*. 27, 9 (1998), 1067–1077.

[12] Riecke, B.E. et al. 2009. Auditory self-motion simulation is facilitated by haptic and vibrational cues suggesting the possibility of actual motion. *ACM Trans. Appl. Percept.* 6, 3 (2009), 1–22.

[13] Riecke, B.E. 2011. Compelling Self-Motion Through Virtual Environments Without Actual Self-Motion – Using Self-Motion Illusions ("Vection") to Improve User Experience in VR. In J. Kim (Ed.). *Virtual Reality*. InTech. 149–176.

[14] Riecke, B.E. et al. 2005. Influence of Auditory Cues on the visually-induced Self-Motion Illusion (Circular Vection) in Virtual Reality. *Proceedings of 8th Annual Workshop Presence* (2005), 49–57.

[15] Riecke, B.E. et al. 2015. More than a Cool Illusion? Functional Significance of Self-Motion Illusion (Circular Vection) for Perspective Switches. *Frontiers in Psychology*. 6, 1174 (2015).

[16] Riecke, B.E. et al. 2009. Moving sounds enhance the visually-induced self-motion illusion (circular vection) in virtual reality. *ACM Transactions on Applied Perception (TAP)*. 6, (2009), 7:1–7:27.

[17] Riecke, B.E. et al. 2011. Spatialized sound enhances biomechanically-induced self-motion illusion (vection). *Proceedings of ACM-CHI* (Vancouver, Canada, 2011), 2799–2802.

[18] Riecke, B.E. et al. 2010. Spatialized sound influences biomechanical self-motion illusion ("vection"). *Proceedings of the 7th Symposium on Applied Perception in Graphics and Visualization* (2010), 158–158.

[19] Riecke, B.E. et al. 2005. Top-Down and Multi-Modal Influences on Self-Motion Perception in Virtual Reality. *Proceedings of HCI international 2005* (Las Vegas, NV, USA, 2005), 1–10.

[20] Riecke, B.E. et al. 2005. Towards Lean and Elegant Self-Motion Simulation in Virtual Reality. *Proceedings of the IEEE Conference on Virtual Reality* (2005), 131–138.

[21] Riecke, B.E. and Schulte-Pelkum, J. 2013. Perceptual and cognitive factors for self-motion simulation in virtual environments. *Human Walking in Virtual Environments*. F. Steinicke et al., eds. Springer.

[22] Seno, T. et al. 2012. Perceptually plausible sounds facilitate visually induced self-motion perception (vection). *Perception*. 41, 5 (2012), 577–593.

[23] Stein, S. von 1910. *Schwindel (Autokinesis externa et interna)*. O. Lessier.

[24] Urbantschitsch, V. 1897. Über Störungen des Gleichgewichtes und Scheinbewegungen. *Z. Ohrenheilk.* 31, (1897), 234–294.

[25] Väljamäe, A. 2009. Auditorily-induced illusory self-motion: A review. *Brain Research Reviews*. 61, 2 (2009), 240–255.

[26] Väljamäe, A. et al. 2009. Auditory Landmarks Enhance Circular Vection in Multimodal Virtual Reality. *Journal of the Audio Engineering Society*. 57, 3 (Mar. 2009), 111–120.

[27] Väljamäe, A. et al. 2004. Auditory Presence, Individualized Head-Related Transfer Functions, and Illusory Ego-Motion in Virtual Environments. *Proceedings of 7th Annual Workshop of Presence* (Valencia, Spain, 2004), 141–147.

[28] Väljamäe, A. and Sell, S. 2014. The influence of imagery vividness on cognitive and perceptual cues in circular auditorily-induced vection. *Perception Science*. 5, (2014), 1362.

Quantifying The Time-Varying Relation Between Speech Production And Postural Control At Different Levels Of Vocal Effort

Robert Fuhrman, Adriano Vilela-Barbosa, Eric Vatikiotis-Bateson

Speech production affects the behavior of physiological systems beyond the vocal tract. In this study, we investigate how increasing the loudness of speech production (vocal effort) affects the coupling of kinematic motions the head, body, and speech acoustics, and the relation of these coupling patterns to postural control. We predicted that increasing vocal effort would result in stronger couplings and increasingly regular kinematic and acoustic signals, reflecting an entrainment effect related to the changes in respiratory patterning associated with producing speech at high levels of vocal effort. We also predicted that postural instability would accompany this effect, since the postural control system is biomechanically coupled to the speech system via the musculature of rib cage, which takes on a more active role in respiration in high vocal effort speech. Results from six talkers who produced spontaneous speech at multiple levels of vocal effort confirmed both of these hypotheses. Stronger within-talker coupling, and the diminished behavioral variability that accompanies it, is associated with weaker coupling between the individual feet—the interface of the body with the external environment. We demonstrate this using a combination of time-series analysis methods assessing signal fluctuations (detrended fluctuation analysis) and system dimensionality (principal component analysis (PCA), correlation map analysis (CMA)). In future work, we plan to include measures of respiration in this analysis.

Preliminary Findings From Investigations Of The Relationship Between Hearing And Segmental Orientation During Walking

Sin-Tung Lau, Michael Cinelli, Karl Zabjek, Nicolas Nasri Ellaham, Jennifer Campos

Several longitudinal studies over the past seven years have found associations between hearing loss and increased falls risk1,2, poor postural control while standing2, increased likelihood of having difficulties with walking3, slower walking speed4 and frailty5. The present studies investigate one possible explanation of the link between hearing, balance, and mobility, being that listeners may change head and trunk orientation when performing a listening task in which potential sound sources are spatially separated. In this presentation, preliminary findings with respect to head and trunk positioning (i.e., segmental orientation) from two independent but related studies will be shared. In the first study, younger (18-30 years old) and older (60 years old and older) adults with self-reported normal hearing performed a listening task while walking. Participants completed listening tasks in three conditions. In the first condition, participants walked along a 9m path, when they were 5m from the goal were presented with a 1 kHz pure tone from either the left or right speakers and were instructed to orient their head towards the sound source while maintaining a straight walking directory. In the second and third experimental conditions, participants performed a series of sentence recognition tasks while walking in which they were and were not instructed to orient their head position towards the sound source, respectively. The second study included younger and older adults with self-reported normal hearing, and older adults with poor hearing and who do not use hearing aids. Participants performed a sentence-in-babble recognition task during three experimental conditions: (1) while sitting on a bench, (2) while standing, and (3) while walking along a pathway similar to that of Study 1. In both studies, motion capture technology was used to collect positional information about different parts of the body. References: 1Lin, F. & Ferrucci, L. (2012). Hearing loss and falls among older adults in the United States. Arch Intern Med, 172(4), 369-371. 2Viljanen, A., Kaprio, J., Pyykko I., Sorri, M., Pajala, S., . . . & Rantanen, T. (2009). Hearing as a predictor of falls and postural

balance in older female twins. J Gerontol A Biol Sci Med Sci, 64A(2), 312-317. 3Viljanen, A., Kaprio, J., Pyykko, I., Sorri, M., Koskenvuo, M., & Rantanen, T. (2009). Hearing acuity as a predictor of walking difficulties in older women. J Am Geriatr Soc, 57, 2282-2286. 4Li, L., Simonsick, E., Ferrucci, L., & Lin, F. (2013). Hearing loss and gait speed among older adults in the United States. Gait Posture, 38(1), 25-29. 5Kamil, R., Li, L., & Lin, F. (2014). Association of hearing impairment and frailty in older adults. J Am Geriatr Soc, 62(6), 1186-1188.

Auditory, Tactile And Bimodal Synchronization To The Beat: A Test Of The Auditory-Motor Enhancement Hypothesis

Frank A. Russo, Paolo Ammirante, Aniruddh D. Patel

Previous tapping studies have suggested that auditory-motor coupling is more robust than visual-motor coupling in the context of rhythmic processing. Does the auditory advantage persist for other modalities? In the current study, nonmusicians were asked to tap their finger to the beat of auditory, tactile, and bimodal metronomes. Stimulus magnitude and rhythmic complexity were also manipulated. In conditions involving a large area of stimulation and simple rhythmic sequences, tactile synchronization closely matched auditory. Although this finding shows a limitation to the hypothesis of enhanced auditory-motor coupling for rhythmic processing, other findings clearly support it. First, there was a robust advantage with auditory information for synchronization with complex rhythm sequences; moreover, in complex sequences a measure of error correction was found only when auditory information was present. Second, higher order grouping was evident only when auditory information was present.

Rethinking Multimodal Integration: The Importance Of Event Unity

Michael Schutz

Our perceptual system continuously processes information from multiple modalities. Some of these derive from common sources – such hearing a speaker's voice while watching their lip movements. Others arrive independently – for example watching the same speaker's lips when a passing train masks their voice. Deciding which sights and sounds "go together" is essential in reconstructing an accurate internal representation of our environment. The challenges of this process are illustrated vividly whenever clever ventriloquists trick audiences into "hearing" puppets speak. This "ventriloquist effect" is explored in laboratories through tasks involving localization of disembodied tone beeps and light flashes. Divorced from real world referents, these stimuli provide context-free tools thought to illuminate generalized principals of perceptual processing. A large body of research suggests a pattern of "optimal integration" in which the highest quality information is weighted most strongly, accurately describing both audio-visual (Alais & Burr, 2004) and visual-haptic tasks (Ernst & Banks 2002). However, the degree to which these models generalize beyond artificial stimuli is a matter of some debate. Here I will discuss new findings from my group arguing for a broader understanding of audio-visual integration using naturalistic stimuli. Broadly construed, this approach explores trade-offs between three parameters. The first two, coincidence in space and time, are well understood by virtue of previous work. However the third– event congruity – also plays a crucial albeit overlooked role. Although more difficult to quantify than spatial agreement or temporal alignment, event congruity plays an important role in the audio-visual binding of speech (Vatakis & Spence, 2007) impact events (Vatakis & Spence, 2008), and musical notes (Chuen & Schutz, in press; Schutz & Kubovy, 2009). Together this research raises intriguing questions about the best practices for exploring the perceptual system's ability to integrate realistic, time-varying stimuli more complex than flashes and beeps.

SAVE TIME AT EVERY STAGE FROM MEASUREMENT THROUGH TO REPORT

FOR GOOD MEASURE



Taking accurate acoustic measurements is critical – but not everything. You also need to efficiently document and report on what you're doing to finish a project.

Introducing the new Measurement Partner, a suite of innovative solutions that supports customers through every step of the measurement process. Take remote measurements with its Wi-Fi-enabled sound level meters so sound fields aren't disturbed. Efficiently make integrated annotations with its smartphone field app. And automatically upload data to the cloud to immediately begin using its post-processing software. Completing the solution are a resource-rich learning centre and an online community with blog posts and discussion forums.

See more at **www.bksv.com/measurementpartner**

Brüel & Kjær

BEYOND MEASURE

Bruel & Kjaer North America Inc. (HQ)
2815-A Colonnades Court
Norcross, Georgia 30071-1588
www.bksv.com

OCCUPATIONAL NOISE AND HEARING PROTECTION - PROTECTION AUDITIVE

Effect Of Ear Canal Occlusion On Loudness Perception
Fabien Bonnet, Jérémie Voix, Hugues Nélisse 154

Using Csa Standard Z107.56 To Investigate Noise Exposure Of Call Center Employees
Shira Daltrop, Ryan Bessey 156

Ear Canal Deformations By Various Earplugs: An In Situ Investigation Using Mri
Simon Benacchio, Arthur Varoquaux, Eric Wagnac, Olivier Doutres, Virginie Callot, David Bendahan, Franck Sgard 158

Custom Molded Silicon Earplugs: Effect Of Material Properties On Acoustic Attenuation And Mechanical Skin Contact
Guilhem Viallet, Jérémie Voix 160

Pilot Study On Individual Dose-Response Relationship Evaluated Through Otoacoustic Emission Measurements In Controlled Noise Exposure: Influence Of Circadian Rhythm
Vincent Nadon, Annelies Bockstael, Dick Botteldooren, Jérémie Voix 162

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 164

EFFECT OF EAR CANAL OCCLUSION ON LOUDNESS PERCEPTION

Fabien Bonnet [1], Jérémie Voix [*1] and Hugues Nélisse [2]
[1]École de Technologie Supérieure, Université du Québec, 1100 rue Notre-Dame Ouest, Montréal, Québec H3C 1K3, Canada
[2]Institut de Recherche Robert-Sauvé en Santé et Sécurité au Travail (IRSST), 505, Boulevard de Maisonneuve Ouest, Montréal, Québec H3C 1K3, Canada

1 Introduction

In a recent paper [1], some of the benefits of in-ear noise dosimetry were brought forward together with its challenges with regard to the prevention of Noise Induced Hearing Loss (NIHL). As part of the challenges, the influence of ear occlusion on noise sensitivity had to be further studied. This is to ensure that appropriate acoustic corrections are used to make in-ear noise levels truly representative of the noise dose received by someone having those noise levels measured under a potential HPD (Hearing Protector Device). In other words: if a given sound pressure level is applied to the eardrum of an occluded ear, is it likely to provoke the same hearing damage as if it were to be applied to the eardrum of an open ear? Keidser et al. [2], who compared the equal loudness noise contours in the open and occluded ear, found that "when balancing or rating loudness, normal-hearing listeners tend to select an average of 10 dB higher level for low-frequency sounds at 500 Hz when listening with the ear occluded".

This paper presents data from an experiment that was made using a rather innovative methodology, while sharing the hypothesis that loudness should well represent the damage that noise can inflict to the hearing system. The results, which confirmed and generalized earlier findings to a wider frequency range, also allowed to remove several factors to explain the observed difference.

2 Method

The experiment led six human participants to carry out a one-hour loudness balance test with headphones. As narrowband noise was played in both ears simultaneously, the subject was to balance the left and right channels to have the same loudness on both sides. In the last two tasks, the balance was done having one ear occluded by an earplug.

2.1 Instrumentation

The test took place in an audiometric sound booth and used FitCheck headphones [3]. During all the experiment, in-ear sound signals were recorded on both sides by means of miniature electret microphones (Knowles GA series) placed inside the ear canal, and connected to a PXI-4462 (National Instruments) acquisition module. Sound pressure levels (SPL) were computed using Matlab scripts. Each subject had access to a computer mouse connected to a laptop for balancing the audio channels using PureData software. ETY plugs [4] were used to balance noise in the open vs. occluded condition.

[*] jeremie.voix@etsmtl.ca

2.2 Subjects

Ten volunteers were pre-selected for the experiment. Those all had hearing threshold levels of 20 dB HL, or less, across the frequency range 125-8000 Hz, and an otoscopic examination on both ears revealed no abnormalities. Each subject's ability to balance loudness with enough precision was assessed during the first task of the experiment. During this task, the participant was asked to balance several frequencies on four separated trials. At each frequency, four values were therefore collected (the collected value was the voltage output difference between the two channels) and the inter-trial standard deviation was calculated to evaluate the intra-subject repeatability. Four subjects, who had an average standard deviation across all frequencies of more than 3dB, were not retained for the rest of the experiment.

2.3 Procedure

The loudness balance was done in a bilateral way. Thus, when the subject increased the input level by +X dB in one ear, the level decreased by –X dB in the opposite ear. This is to prevent the participant to focus his attention on one ear, which could be a source of bias. The mouse wheel was used to balance noise from one ear to the other, with a resolution of 1 dB SPL and a range of ±15 dB SPL in each ear. The subject was asked to click the left mouse button whenever he was finished balancing and ready to switch to the next frequency. Every time a new track/frequency was started, the initial audio balance was randomly chosen among the 30 positions available to ensure the participant remained active during the exercise. The initial input gains were pre-calibrated on a dummy head at each frequency so that, where the two input channels meet at the same level, the SPL generated at the eardrum should correspond to a value of approximately 4 sones.

Each retained subject was asked to balance narrowband noise with a bandwidth of one-third of an octave centered on nine frequencies: 125, 250, 500, 1000, 2000, 3150, 4000, 6300, and 8000 Hz. This was done in three separated tasks, with several trials per task (each trial contained 10 tracks as the frequency 1000 Hz was assessed twice and averaged in each trial). In the first task, both ears were open and the intra-subject repeatability was assessed as well as the initial difference of sensitivity between the subject's left and right ears. This open-ear correction was defined as the mean transfer function between the two in-ear microphones at equal loudness, across four trials. In the second task, the participant had to balance noise in the open vs. occluded condition. An earplug was first inserted inside the subject's left ear and its insertion loss was estimated as the transfer

function between the two in-ear microphones when white noise was being played on four speakers (one in every corner of the sound booth). The insertion loss was used to correct the level in the left ear channel so that the balance can be done approximately at the same levels as in the first task. The occluded-ear correction was then computed as the mean equal loudness level difference between the occluded ear and the open ear, across two trials, and corrected with the open-ear correction. The third task consisted only in repeating the second task with the right ear occluded.

3 Results

Figure 1 shows the distribution of the twelve resulting occluded-ear corrections measured on the six retained subjects. Each subject's individual mean correction (arithmetic average of the occluded-ear corrections obtained on both ears) was then computed and used to calculate the intersubject standard deviation σ (Table I).

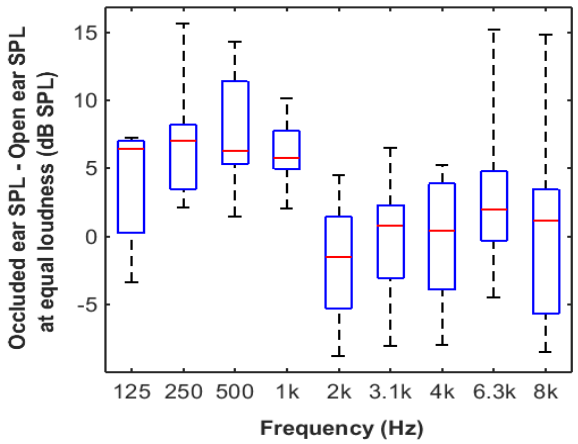


Figure 1: Distribution of the differences in equal loudness levels measured on six subjects (two results per subject, i.e. twelve values) at nine frequencies in the open vs. occluded ear. The boxes show the 2nd and 3rd quartiles separated by the median (in red), while the dashed whiskers show the 1st and 4th quartiles.

Frequency (Hz)	125	250	500	1k	2k	3.1k	4k	6.3k	8k
Difference (dB)	*4.1*	*6.7*	*7.8*	*6.1*	*-1.7*	*0.0*	*-0.2*	*3.0*	*0.4*
σ (dB)	*3.1*	*2.7*	*3.1*	*0.9*	*3.5*	*3.1*	*3.6*	*3.1*	*3.4*

Table I: Average difference (N = 6) in equal loudness levels measured at nine frequencies in the open vs. occluded ear. σ is the intersubject standard deviation.

4 Discussion

For safety purposes, the microphones were never closer than 1 cm to the eardrum, and the results for frequencies ranging from 3.1 to 8 kHz should be carefully interpreted as they might be influenced by some resonance in the open ear. Nevertheless, it clearly appears that up to 1 kHz sounds were selected at much higher levels in the occluded ear than in the open ear, until this difference drops at 2 kHz.

To our knowledge, this is the first time loudness in the open-ear and occluded-ear conditions were compared using the same transducer on both sides, which is extremely important when discussing the factors that may cause such discrepancies. Indeed, one of the main potential factors given by Keidser et al. to explain the difference was the source location effect described in Rudmose's experiments [5]: "the near source is regarded as having a "smaller acoustic size" and consequently must produce more sound pressure in the ear canal to equal the loudness of the distant and "larger" loudspeaker". Having used headphones on both sides in our experiment, this factor should be discounted with regard to the results seen in Fig. 1, just like any factor that is to do with the use of a loudspeaker (e.g. mechanical vibration through the subject's chair).

As stated by Keidser et al. [2], "one possibility is that occlusion of the canal changes the relationship between SPL near the eardrum and the power entering the middle ear". We have been studying many ways of modeling this effect, but none has given entire satisfaction so far.

Another possibility mentioned in Keidser's work is the acoustic reflex explanation. This reflex typically introduces a transmission loss of 5 to 10 dB through the middle ear. It is also known to mostly affect low frequency sounds (<1.2 kHz), which matches surprisingly well our results and the drop at 2 kHz. Hence, even though there is no evidence in the literature that the acoustic reflex may be more easily triggered by occluding the ear, we believe this to be a serious path of investigation. What if ear occlusion, when detected by the human brain, was interpreted by the latter as a sign of danger for our hearing, and therefore responded in a self-protective manner by triggering the acoustic reflex?

5 Conclusions

While the reasons that may cause someone to perceive sounds softer when listening with an occluded ear are still to be discovered, these results raise important questions as regards to the determination of noise exposure received by people wearing earplugs, and perhaps other types of HPDs. In the long term, such data may even have an impact on the standards that define the free-field corrections to be used for the determination of sound immission from communication equipment (ISO 11904, CSA Z107.56). Finally, they are a key point to consider for in-ear dosimetry applications.

Acknowledgments

The IRSST (Quebec occupational health and safety research institute) is gratefully thanked for their financial support.

References

[1] F. Bonnet, *et al.* "The opportunities and challenges of in-ear noise dosimetry," in *Acoustics Week in Canada 2015*, 2015.

[2] G. Keidser *et al.* "Relative loudness perception of low and high frequency sounds in the open and occluded ear," *J. Acoust. Soc. Am.*, vol. 107, no. 6, pp. 3351–3357, Jun. 2000.

[3] www.earplugstore.com/fitcheck-solo-earplug-fit-tester.html

[4] www.etymotic.com/consumer/hearing-protection/er20.html

[5] W. Rudmose, "The Case of the Missing 6 dB," *J Acoust Soc Am*, vol. 71, no. 3, pp. 650–659, 1982.

USING CSA STANDARD Z107.56 TO INVESTIGATE NOISE EXPOSURE OF CALL CENTRE EMPLOYEES

Shira Daltrop*[1] and Ryan Bessey†[2]
[1]Golder Associates Ltd., Suite 200 - 2920 Virtual Way, Vancouver, BC, Canada, V5M 0C4
[2] Golder Associates Ltd., Suite 910 - 141 Adelaide Street W., Toronto, Ontario, Canada M5H 3L5

1 Introduction

A study was performed to measure the noise exposure of call centre employees working in a new LEED Platinum office space that is equipped with low-height partitions and minimal acoustical treatments. Due to the nature of the acoustical environment, complaints of excessive noise have arisen along with concerns of health and safety noise impacts. The noise exposure of the employees was measured and compared with criteria in B.C. Reg. 382/2004 [1], following the guidelines of CSA Standard Z107.56 [2], as required by B.C. Reg. 382/2004. The B.C. Reg. states that an employer must ensure that a worker is not exposed to noise levels above either of the following exposure limits: 85 dBA Lex daily noise exposure level and 140 dBC peak sound level.

The study involved performing unattended background noise monitoring within the call centre area as well as measuring the noise contribution of the call centre employees' headsets using a Head and Torso Simulator (HATS) manikin.

2 Method

2.1 Background Noise Monitoring

Background noise monitoring was performed to determine the ambient noise within the call centre. A Brüel & Kjær 2250 Type 1 integrating Sound Level Meter (SLM) was set up within the call centre at one of the desks, chosen due to its central location of the largest open area of the call centre. The height of the microphone was set at 1.1 m to represent the height of a typical seated worker. Sound levels were logged every minute during the monitoring period. Monitoring occurred over a 24 hour period and was recorded to allow for post-measurement data review.

2.2 Headset Measurements

Measurements of sound levels from a typical call centre headset were performed following the guidelines of Section 7.3.2 in CSA Z107.56-13. A manikin (Brüel & Kjær Type 4128C HATS) with microphones at its eardrums was set up in the chair of the same desk where the monitoring was performed and connected to the SLM.

As the phone system had multiple volume controls, each setting was verified by a call centre employee. Once the volume setting was adjusted to the maximum level, just below the sound level where clipping and/or distortion in the headset started to occur, the headset was carefully placed on the manikin. Since the manikin microphones were connected to the SLM, it was possible to monitor the manikin signal from a secondary headset connected to the SLM. The placement of the headset on the pinna of the manikin was carefully checked and adjusted until good coupling was achieved, as observed visually and through monitoring with the secondary headset.

With the headset installed on the manikin, a set of three scripted test calls were acted out by Golder staff so that the microphones in the manikin's ears would receive a realistic signal (using actual calls was not possible with the current call centre equipment). The scripts consisted of typical call centre conservations with a calm tone, a medium tone, and an elevated tone. This was done to determine the variation in call sound level that might occur due to the type of conversations that may be expected.

In the call centre, Golder staff spoke into the headset microphone on the manikin and listened through a secondary headset connected to the SLM. Using the call centre phone system, calls were placed and received to Golder's office where Golder staff first used a typical office phone in a quiet meeting room followed by a cell phone next to a busy roadway. The purpose of these scenarios was to capture the variation in call sound level that may be due to the customer's background sound level, type of phone used, and speaking environment.

Once the data was collected, it was processed following the CSA Z107.56 recommended procedure for converting the measured manikin eardrum sound pressure levels into diffuse field equivalent sound pressure levels. The conversion is required for proper comparison to the applicable 85 dBA criterion. The 'long method' was utilized based on one third octave band data. It should be noted that the CSA Z107.56 standard does not provide any guidance on eardrum-to-diffuse field correction factors for peak sound levels for comparison against the 140 dBC criterion. Even so, the SLM only provides an overall C-weighted peak sound level (i.e. no peak spectrum data was available), which would make applying a correction difficult. Therefore, it was decided to report uncorrected eardrum measured peak C-weighted sounds levels, which represents a conservative assessment.

3 Results

3.1 Background Noise Monitoring

Based on a review of audio recordings, some of the 1 minute intervals during the monitoring period were excluded from analysis, due to call centre employees talking very close to the measurement microphone or making other anomalous noises. The time period from 7 pm to 6 am was also excluded

* sdaltrop@golder.com
† rbessey@golder.com

since the call centre was closed and the lower noise levels were not representative of typical working conditions. Based on the valid 1 minute intervals, the sound level statistics summarized in Table 1 were calculated.

Table 1: Sound level statistics during the monitoring period.

Metric	SPL, dB, A or C Weighted
Min	41 dBA
L_{99}	44 dBA
L_{90}	48 dBA
L_{50}	53 dBA
L_{eq}	54 dBA
L_{10}	57 dBA
L_{1}	61 dBA
Max	63 dBA
Peak	108 dBC

3.2 Headset Measurements

The results of the headset measurements on the maximum volume level are summarized in Table 2 below.

Table 2: Headset call measurement results.

Caller Location	Script Tone	Diffuse-field SPL (L_{eq}, dBA)	Eardrum Peak SPL (Peak,
Quiet Office	Mild	74	110
Quiet Office	Medium	74	112
Quiet Office	Elevated	77	113
Busy Road	Mild	81	115
Busy Road	Medium	81	114
Busy Road	Elevated	83	116

For comparison against the applicable criterion of 85 dBA for an 8-hour average, a time-weighing correction must be applied based on the amount of time expected in each of the measured call scenarios. However, it can already be determined that since none of the scenarios were above 85 dBA, it is not possible for the resulting time-weighted average to exceed the 85 dBA criterion.

Since a breakdown of call types and caller noise environments was not provided, it was assumed that 90% of the calls had a mild / medium tone and 10% had an elevated tone, and that 75% of calls were made from a quiet environment and 25% from a loud environment. Based on these assumptions, the average call sound level was determined to be 78 dBA. It should be noted that even if 100% of the calls were made with an elevated tone and from a noisy environment using a cell phone, the average call sound level would still only be 83 dBA.

The average background noise level in the call centre was measured to be 54 dBA. During an 8-hour shift a typical call centre worker will spend up to 6 hours on calls, 1 hour on break, and 1 hour doing other activities such as computer work at their desk. Assuming that during the 2 hours not on calls the workers are exposed to a background noise level of 54 dBA and for the remaining 6 hours are exposed to average call sound levels of 78 dBA (assumed to be using the maximum volume setting), the daily noise exposure for a typical call centre worker (Lex, 8hr) is 76 dBA.

Reviewing the measured peak dBC weighted sound levels, the maximum peaks during the background sound level monitoring and headset measurements were 108 dBC and 116 dBC, respectively. It should be noted that the peak level from the headset measurements were not corrected using an eardrum to diffuse field transfer function and thus likely over-estimate actual noise exposure.

4 Discussion

Based on the measurement results and calculation of daily noise exposures, the 8-hour average and peak sound levels were found to be in compliance with B.C. Reg. 382/2004 criteria even when making conservative assumptions and with headsets on the maximum volume setting.

According to CSA Z107.56, for effective communication a call's average sound level is typically required to be 15 dB above the background noise level. Based on the average background sound level of 54 dBA for the call centre, the minimum required sound level should be 69 dBA in order to provide at least a 15 dB signal-to-noise ratio. As reported in Table 2, the minimum measured average call sound level was 74 dBA, which is 5 dB higher than expected for effective communication. This suggests this assessment represents a conservative analysis, as even for quieter calls a 5 dB lower volume setting could have been used and would still allow for effective communication. This also suggests that CSA Z107.56's 'calculation method' to predict noise exposure of call centre employees based on measured background sound levels and an assumed 15 dBA signal-to-noise ratio could under-predict employees' noise exposure.

The CSA Z107.56 also provides a 'short method', which uses a single number correction when converting the eardrum sound levels to diffuse field sound levels, instead of using the third octave band corrections. It recommends subtracting 5 dB from the A-weighted eardrum sound level to derive the A-weighted diffuse field sound level. Using the third octave band corrections, the average difference between the diffuse field and eardrum sound levels measured here was 10.5 dBA. Therefore, using the 5 dB single number correction could over-estimate call centre employees' noise exposure.

5 Conclusion

An investigation related to noise exposure of employees in a new call centre was undertaken. Measurements of background noise within the call centre as well as the sound level generated by the call centre's headsets were performed. The investigation was performed in accordance with B.C. Reg. 382/2004 following the guidelines of CSA Standard Z107.56. Based on the results, the daily noise exposure of the workers was found to be less than the 85 dBA criterion and peak sound levels were found to be less than 140 dBC.

References

[1] Workers' Compensation Board. Occupational Health and Safety Regulation, B.C. Reg 382/2004. 2004.

[2] Canadian Standards Association. Measurement of Noise Exposure, CAN/CSA Z107.56-13. 2013.

EAR CANAL DEFORMATIONS BY VARIOUS EARPLUGS: AN IN SITU INVESTIGATION USING MRI

Simon Benacchio[*1,2,3], **Arthur Varoquaux**[†4,5], **Éric Wagnac**[‡1,3], **Olivier Doutres**[§1], **Virginie Callot**[¶4,5], **David Bendahan**[‖4,5], **and Franck Sgard**[**2]

[1]École de technologie supérieure, 1100 Rue Notre-Dame O, Montréal, QC H3C 1K3

[2]Institut de recherche Robert-Sauvé en santé et sécurité du travail, 505 Boulevard de Maisonneuve O, Montréal, QC H3A 3C2

[3]Centre de recherche de l'Hôpital du Sacré Cœur de Montréal, 5400, Boulevard Gouin O, Montréal, QC H4J 1C5

[4]Aix Marseille Université, CNRS, CRMBM UMR 7339, Marseille, France

[5]APHM, Hôpital de la Timone, Pôle d'imagerie médicale, CEMEREM, Marseille, France

1 Introduction

To be efficient, a Hearing Protection Device (HPD) should be worn during the time the workers are exposed to high level noises. However, considering the corresponding hearing difficulties and/or physical discomfort, workers do not use HPDs properly and consistently. In the past, some indicators of physical (static pressure on the ear canal (EC) walls) [1] and auditory (insertion loss, occlusion effect) [2, 3] discomfort induced by earplugs (EPs) have been assessed using numerical models. However, the *in situ* deformation of EPs and tissues surrounding the EC has never been observed experimentally. These deformations could provide useful information to investigate how EPs and tissues should be modelled and characterized mechanically in order to predict EP efficiency indicators such as sound attenuation, occlusion effect and mechanical pressure applied on the EC walls. In this study, we assessed the deformation of the open and occluded EC of a single human subject using high-resolution magnetic resonance imaging (MRI) and for seven different EPs.

2 Method

2.1 Earplug insertion in the ear canal

Ear canal. The deformation of open and occluded EC was investigated for both ears. In order to raise the EPs conspicuity and the external auditory canal delineation, ECs were filled with natural almond oil.

Earplugs. Three groups of EPs were investigated: the moldable EPs made from polyvinyl chloride or polyurethane foam materials, the pre-molded EPs made from silicone or foam and plastic and the custom molded EPs made from silicone, acrylic or thermosoft materials. The seven EPs and a qualitative description of their rigidity and initial shape are presented in Table 1. A view of each EP is given in Figure 1.

2.2 High-resolution magnetic resonance imaging

As few pathologies involve the outer ear, this part of the body has never been worthy of interest in MRI studies. Thus, a 3D space MRI sequence described in Table 2 has been optimized to non-invasively discriminate the tissues surrounding the EC. Using these parameters, the fat tissues and the almond oil appear white on images, the skin and the cartilaginous parts appear grey and the bones, the foam, the plastic, the acrylic, the thermosoft material and the air appear black (see for example Fig. 1(b)). Given the relatively small diameter of the RF coil (20 cm) as compared to the cranial diameter, the left and right pinna of the subject were slightly deformed during MR acquisitions. Moreover, ear muffs were used during open EC measurements to protect the subject from the MR system noise. In this case, the pinnae of the subject were largely deformed. However, according to first observations, it was considered that, from the concha to the eardrum, the EC deformation due to the RF coil and ear muffs was small enough and thus acceptable.

Table 1: Description of the earplugs. The mathematical symbols (-/+) qualitatively sort the rigidity and initial shape of earplugs.

EP name (and type)	Materials (and rigidity)	EP/EC similarity
$\mathbf{M_f}$ (moldable)	foam (- -)	- -
$\mathbf{PM_s}$ (pre-molded)	silicone (-)	- -
$\mathbf{PM_{fp}}$ (pre-molded)	foam + plastic (-)	- -
$\mathbf{M_{s25}}$ (molded)	silic. shore A 25 (+)	++
$\mathbf{M_{s40}}$ (molded)	silic. shore A 40 (++)	++
$\mathbf{M_t}$ (molded)	thermosoft (+++)	++
$\mathbf{M_a}$ (molded)	acrylic (++++)	++

Table 2: Main parameters of the optimized MRI sequence.

MR system	Siemens 3T (Verio)
Radiofrequency (RF) coil	Head 32 channels
Sequence	Space 3D
Weighting	T1
Plane	Axial
Slices	128
Resolution [mm]	0.6 iso.
Acquisition time [min]	9

2.3 MRI post-processing

To accurately estimate the EC deformation, all images have been repositioned using a rigid registration method from the *Advanced Normalization Tools (ANTs)* library toolkit [4]. Then, each EC was segmented using one slice of the axial plane. This slice was arbitrarily chosen to observe the EC deformation for each EP and is presented in Figure 1. The boundaries of the segmentation were given by the EC walls and a sagittal plane located in the concha. On the one hand,

[*]Simon.Benacchio@irsst.qc.ca
[†]DamienArthur.VAROQUAUX@ap-hm.fr
[‡]Eric.Wagnac@etsmtl.ca
[§]Olivier.Doutres@etsmtl.ca
[¶]Virginie.CALLOT@univ-amu.fr
[‖]david.bendahan@univ-amu.fr
[**]Franck.Sgard@irsst.qc.ca

this plane was chosen to observe the largest part of the EC and, on the other hand, to avoid the deformation due to the RF coil and ear muffs. The example of the open EC segmentation is given by the red area in Figure 1(b). Finally, the EC deformation was related to the number of voxels of each segmentation and qualitatively evaluated by a radiologist.

3 Results and discussion

Figures 1(a) and 1(b) show the non occluded (**NO**) EC. The other images present the rigidly registrated occluded EC using the moldable (Fig. 1(c) and 1(d)), pre-molded (Fig. 1(e) and 1(f)) and custom molded (Fig. 1(g), 1(h), 1(i) and 1(j)) EPs. The images are displayed using *FSLView* [5]. Table 3 gives the segmented number of voxels for both open and occluded ECs. The deformation of the EC clearly depends on

Table 3: Number of voxels for the segmentations of the open and occluded ECs.

EP (right)	**NO**	$\mathbf{M}_f$	$\mathbf{PM}_{fp}$	$\mathbf{M}_{s25}$	$\mathbf{M}_a$
Voxels	141	332	330	281	290
EP (left)	**NO**	$\mathbf{M}_f$	$\mathbf{PM}_s$	$\mathbf{M}_{s40}$	$\mathbf{M}_t$
Voxels	196	302	280	254	232

the type of EP. According to the number of voxels given in Table 3, the moldable EPs and the $\mathbf{PM}_{fp}$ pre-molded EP apply the largest deformation on the EC. Then, the deformation decreases using the $\mathbf{PM}_s$ pre-molded, and the molded EPs respectively. Using the pre-molded EPs, the EC fits the shape of the round and conic EPs as shown in Figure 1(e) and 1(f) respectively. The custom molded EPs induce the smallest but non negligible deformation of the EC as they are made to fit its shape. Note that these observations are only valid for the slice of interest.

4 Conclusions

This pilot study aimed at observing the deformations of the ear canal by seven different earplugs. Magnetic resonance images providing a good contrast between the tissues surrounding the ear canal were acquired to non-invasively characterize these deformations. Using segmentations of the open and occluded ear canal within one slice of the axial plane, we identified the largest deformations with the moldable and the round pre-molded earplugs, while they were smaller with the other pre-molded and the molded earplugs. However, it is not possible to conclude on the global deformation of the EC since only one slice is used for segmentation. The next step of the study is to compute a quantitative analysis on the global deformation of both EPs and EC using non linear registration methods.

Acknowledgments

This work was founded by the Institut de recherche Robert-Sauvé en santé et sécurité du travail. The authors would like to thank Arnaud Le Troter for his help with post-processing.

References

[1] A. T. Baker, S. H. Lee, and F. Mayfield. Evaluating hearing protection comfort through computer modeling. *2010 SIMULIA Customer Conference*, 2010.

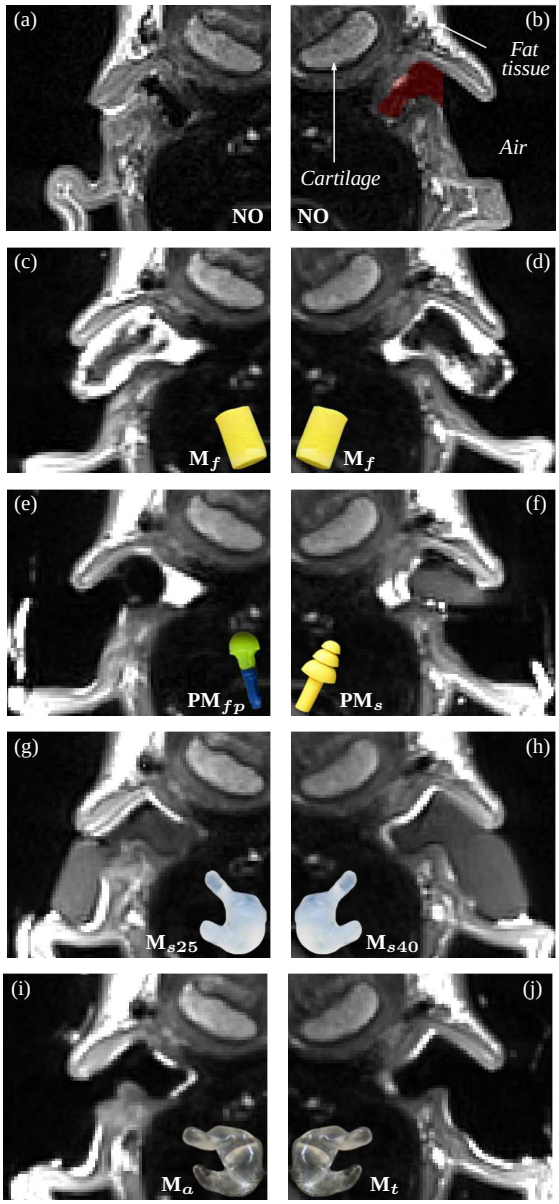


Figure 1: Reference images: (a,b) right and left non occluded (**NO**) ear canal (EC). Rigid registration of the EC occluded by different earplugs: (c,d) moldable ($\mathbf{M}_f$), (e,f) pre-molded ($\mathbf{PM}_{fp}$ and $\mathbf{PM}_s$), (g,h) molded silicone ($\mathbf{M}_{s25}$ and $\mathbf{M}_{s40}$), (i,j) molded acrylic and thermosoft ($\mathbf{M}_a$ and $\mathbf{M}_t$). $\mathbf{PM}_{fp}$, $\mathbf{M}_a$ and $\mathbf{M}_t$ are non visible on MR images but a view of each earplug is given in the bottom corner of images. The red area in Figure (b) shows the segmentation of the open EC.

[2] G. Viallet, F. Sgard, F. Laville, and J. Boutin. A finite element model to predict the sound attenuation of earplugs in an acoustical test fixture. *J. Acoust. Soc. Am.*, 136(3):1269–1280, September 2014.

[3] M. K. Brummund, F. Sgard, Y. Petit, and F. Laville. Three-dimensionnal finite element modeling of the human external ear: Simulation of the bone conduction occlusion effect. *J. Acoust. Soc. Am.*, 135(3):1433–1444, March 2014.

[4] B. B. Avants, N. J. Tustison, G. Song, and J. C. Gee. Ants: Advanced open-source normalization tools for neuroanatomy. *Penn Image Computing and Science Laboratory*, 2009.

[5] M. Jenkinson, C. F. Beckmann, T. E. Behrens, M. W. Woolrich, and S.M. Smith. Fsl. *Neuroimage*, 62(2):782–790, 2012.

CUSTOM MOLDED SILICON EARPLUGS: EFFECT OF MATERIAL PROPERTIES ON ACOUSTIC ATTENUATION AND MECHANICAL SKIN CONTACT.

Guilhem Viallet[1], Jérémie Voix[1]
[1]Dept.de génie mécanique, École de technologie supérieure (ETS),
1100, rue Notre-Dame Ouest, Montréal (Qc), Canada, H3C 1K3, guilhem.viallet@etsmtl.ca

1. MOTIVATION

Silicone is one of the most commonly material used for custom earplugs. Nevertheless, the influence of its mechanical properties on both sound attenuation and mechanical skin contact (which can be assimilated to a comfort factor) are based on empirical statements. The knowledge of silicon's properties is frequently limited to its Shore hardness value which is necessary but insufficient to fully describe the vibro-acoustic behavior of this material. This knowledge can be relevant to improve the design of passive hearing protectors. Viallet et al. [1] performed sensitivity analyses for simulated the acoustic attenuation of a custom earplug however the range of variation chosen for the mechanical properties was arbitrary. Past research works defined and included the contact between the earplug and the skin as mechanical impedance [2] however the variation of this mechanical impedance with the material properties has not been studied yet.

The objective of this work is to quantify the impact of the variation of silicon material properties for both the provided acoustic attenuation and mechanical skin contact. Regarding acoustic attenuation, the choice of an indicator is easy and previous author's works [1] can be compared to the results obtained here for the earplug insertion loss variations. For the mechanical skin contact, this research is more exploratory and one novel indicator has been implemented.

Six medical grade silicones ranging from Shore A 15 to 40 were mechanically tested with a quasi- static mechanical analyzer to get material properties that can be used as a realistic range of variation. Then, a validated 2D axisymmetric finite element model of the occluded auditory canal (acoustical test fixture - ATF - configuration) was used to perform sensitivity analyses upon the range of the measured silicon's mechanical properties.

2. METHOD

2.1 Mechanical properties

The earplug's material is considered as a linear viscoelastic material. The parameters of interest are the following: Young's modulus (E), Poisson's ratio (ν), isotropic loss factor (η) and density (ρ). A quasi-static mechanical analyser (Mecanum) was used to measure the elastical properties of the six selected different types of silicones. The method, according to ISO18437-5, is based on the dynamic compression of two samples of different shape factors combined with polynomial relations, and enable the simultaneous characterization of the elastical properties of interest [3]. The extreme values obtained for each material property are reported in table 1.

Table 1: Extreme values measured for silicon material properties

Property	Min	Max
E (Mpa)	0.395	1.39
ν	0.4	0.49
ρ ($kg.m^{-3}$)	881.8	1085.5
η	0.05	0.25

These extreme values are used as a -1 and + 1 input level code for the design experience detailed hereafter.

2.2 Modeling the EP-EC system and selected indicators

A 2D axisymmetric finite element model of a silicon earplug coupled to an artificial auditory canal, as well as an ATF configuration, was implemented. The model was solved using COMSOL Multiphysics and validated experimentally with attenuation measurments (ANSI-ASA 12-42) carried out with a 45CB ATF. More details about the model can be found in [1].

The insertion loss (IL) was selected as an output indicator to reflect the acoustic attenuation, because of its documented definition in ANSI-ASA S12-42 standard. For the mechanical skin contact, the difference of mechanical impedance's modulus at the skin interface with and without the earplug (Z_{mec}) was calculated and chosen as an indicator. An optimizing value is the minimal distance between those two mechanical impedances. It is important to notice that this relation is local and depend upon the exact position along the interface. In a first approximation, three different positions were considered: at the ear canal entrance (P1), at the middle (P2) and the end of the earplug (P3). Based on these two indicators, two multi-variance analyses were then performed using STATGRAPHICS Centurion software (V16.1.11). The results of these analyses provide the 1^{st} and 2^{nd} order effects of each parameter, i.e. the variation in dB around the average value of the IL or Z_{mec} when the parameter goes from the -1 level to the +1 level codes.

3. RESULTS

3.1 Acoustic attenuation

Figure 1 illustrates the simulated IL together with the first order effects of the material's properties. The top chart shows the possible variation of the IL as a function of frequencies. The bottom part of the figure shows which parameters are responsible for the IL variations. Some

classical mechanical effects were retrieved like the fact that the system is driven mostly by it stiffness at low frequencies. An increase of the Young's modulus typically increases the IL up to the first mode of the earplug (around 1 kHz). At higher frequencies all the parameters seem have a comparable influence on IL's variations and some coupling between parameters begins to occur (not shown here for concision).

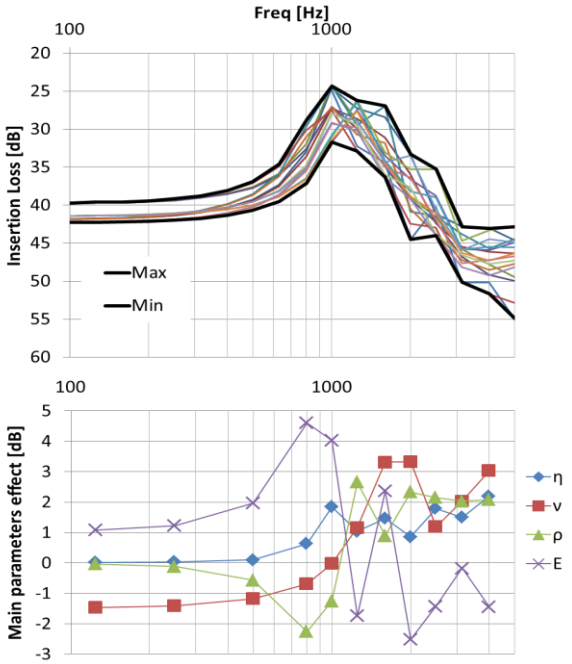


Figure 1: Insertion loss in third octave bands computed for each combination of parameters (16 runs) (top) and the associated first order effects of the material's properties (bottom).

This chart also demonstrates the difficulty to find a unique and/or optimized solution when material properties have negative or positive influences depending on the frequency of interest. These results show that generally the influence for the IL, in dB, is not critical or tremendous (5dB maximum). This founding can be relevant if another indicator has to be optimized, as a function of the material's properties, but without significantly changing the acoustic attenuation.

3.2 Mechanical skin contact

Figure 2 illustrates the simulated variations of the indicator Z_{mec} at the position P2 together with the associated first order effects of the material properties. As well as the acoustic attenuation simulation, the system behaviors become more complex after 1 kHz. At lower frequencies the earplug has a "piston like" behavior and the curves present regular fluctuations. By contrast with the IL results, the main effects are important relatively to the indicator order of magnitude. Up to 1 kHz a decrease of the Young's modulus from level -1 to +1 can reduce by 50% the mechanical impedance difference with and without the earplug. The sensitivity of the indicator Z_{mec} as a function of the position (P1, P2 et P3) exhibits minor differences. An explanation for this minor influence of the position might be the cylindrical geometry (ATF configuration) considered in this work and needs further investigations.

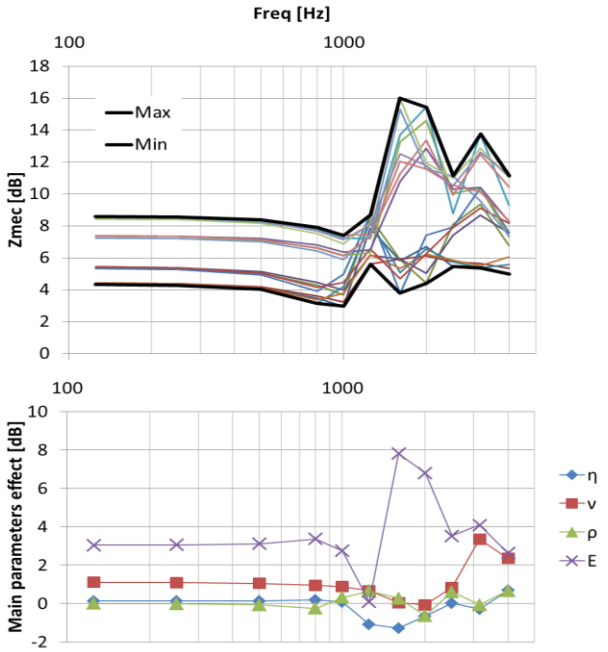


Figure 2: Z_{mec} at position P2 in third octave bands obtained for each combination of parameters (16 runs) (top) and the associated first order effects of the material's properties (bottom).

CONCLUSIONS

In this work, two sensitivity analises relatives to the impact of the material properties of silicone on acoustic attenuation and mechanical skin contact have been presented. Futur works will focus on finding an optimized set of parameters to meet a compromised criteri between effiency and confort for the earplug. The following limitations might be considered for future works. First, the variation of Z_{mec} and its associated sensitivity as a function of the position along the ear canal in a more realistic auditory canal geometry. Second, an adjustement of the range of material properties as a function of the compression rate and the frequency.

REFERENCES

[1] G. Viallet, F. Sgard, F. Laville, et J. Boutin, « A finite element model to predict the sound attenuation of earplugs in an acoustical test fixture », *The Journal of the Acoustical Society of America*, vol. 136, n° 3, p. 1269‑1280, 2014.

[2] P. Hiselius, « Attenuation of Earplugs – Objective Predictions Compared to Subjective REAT Measurements », *Acta Acustica united with Acustica*, vol. 91, p. 764‑770, 2005.

[3] C. Langlois, R. Panneton, et N. Atalla, « Polynomial relations for quasi-static mechanical characterization of isotropic poroelastic materials », *J. Acoust. Soc. Am*, vol. 110, n° 6, p. 3032‑3040, déc. 2001.

PILOT STUDY ON INDIVIDUAL DOSE-RESPONSE RELATIONSHIP EVALUATED THROUGH OTOACOUSTIC EMISSION MEASUREMENTS IN CONTROLLED NOISE EXPOSURE: INFLUENCE OF CIRCADIAN RHYTHM

Vincent Nadon [1,2], Annelies Bockstael [2], Dick Botteldooren [2], and Jérémie Voix* [1]
[1]École de technologie supérieure, 1100, Notre-Dame Ouest, Montreal, Quebec, Canada, H3C 1K3
[2]Ghent University, WAVES research group, Technologiepark Zwijnaarde 15, B-9052, Ghent, Belgium

1 Introduction

Over 22 million of North American workers are exposed everyday to noise exposure doses that may induce hearing loss [1]. Even if they use hearing protection devices (HPD) in order to limit their exposure, but the protection is limited since the HPDs may not be correctly fitted nor worn at all times. As a result, Noise Induced Hearing Loss (NIHL) remains one of the biggest cause of invalidity and indemnity in North America [1, 2].

Health and safety practitioners in the workplace periodically perform audiometric measurements to monitor the workers' hearing levels. These measurements are conducted on too long intervals and usually after the hearing damage has appeared, and do not prevent the occupational hearing loss. Therefore, a system able to simultaneously record and process the noise characteristics to calculate the cumulative dose and measure the resulting hearing fatigue continuously, possibly through distortion-product otoacoustic emission (DPOAE), might enable better prevention of NIHL.

Objective

In an attempt to monitor hearing status changes, an experiment was conducted on human subjects in controlled noise exposure conditions measuring pure-tone audiometry (PTA) hearing threshold levels (HTL), stapedial and medial olivocochlear reflexes in addition to continuous otoacoustic emissions monitoring.

The objective of this paper is to consider the possible effects of the circadian rhythm on the hearing health monitoring measurements. The effects of the time of day (TOD) on pre/post exposure comparisons is studied for subjective tests such as the PTA and also for objective measurements such as the stapedial and medial olivocochlear reflex thresholds as well as otoacoustic emissions. Ideally the circadian rhythm should have a minimal effect on the differences between pre (morning) and post (afternoon) noise exposure measurements in order to distinguish true variations caused by the noise exposure. Eventually a thorough analysis of the aggregated results will be conducted in an extensive study to evaluate the true effects of noise exposure on hearing health.

2 Potential effects of circadian rhythm on subjective and objective hearing tests

To detect changes due to noise exposure during their work shift, workers would be tested at the beginning of the work day, in the morning, and at the end of their work day in the afternoon. Workers' focus can be influenced by the TOD possibly affecting the subjective hearing measurements. The metabolic activity of cells involved in the hearing system, such as the cochlea's hair cells, may also depend on the TOD and therefore a difference between morning and afternoon measurements may be related to this activity instead of other factors such as noise exposure.

2.1 Potential effects on pure-tone audiometry

In Ezzatian's study [3] a gap detection test was used to assess the hearing ability of the listeners, a method similar to classic audiometry, it considers the time of detection instead of the HTL. Such measurement gives a good indication whether the participant is wide awake or sleepy. Such participant status could influence the delay to answer to audiometric stimuli and might result in the participant not answering to his true HTL if the audiometry is performed too fast. Although the status of the participants can influence the outcome of subjective hearing tests, it seems that the effects of TOD are not large enough to influence performance in the typical testing conditions as Ezzatian reported in [3].

2.2 Potential effects on stapedial reflex

At the time of writing, no study was found specifically on the effects of TOD on stapedial reflex thresholds. However, the study from Venet [4] addressed the effects of noise exposure on the efferent reflex (ER) and therefore Venet recorded measurements in the morning and the afternoon. Although it is not clear according to Guinan [5] whether Venet [4] is measuring a mixture of medial olivocochlear (MOC) reflex and middle-ear muscle (MEM) activity or simply the MEM activity alone, there is an observable change of about 1-2 dB average in ER threshold levels for the control group, not exposed to noise, which could be attributed to the effects of TOD.

2.3 Time of Day effects on DPOAEs

In spite of the results from the studies on circadian rhythm effects [3], the study from Cacace [6] revealed that an effect of the TOD on DPOAEs could be detected. To detect such effects, Cacace processed time variations in DPOAE levels with a discrete fourier transform (DFT) before analyzing the data with an ANOVA to eliminate phase differences in the changes between subjects and simply look at the frequency of the changes. Although this TOD effect is statistically significant, it is not large enough to influence the test interpretations in standard laboratory test conditions [6].

*jeremie.voix@etsmtl.ca

3 Experiment description

For the experiment, 9 volunteers had to pass a hearing screening test starting with an otoscopic examination, followed by a standard DPOAE test and a classic PTA test. Subjects were exposed on three different days for the following noise conditions: Industrial, Constant, Quiet. On the days of the hearing status monitoring measurements, they first went through the pre-exposure tests, which consisted in a manual PTA, stapedial reflex measurements with tympanometry and medial olivocochlear reflex by contralateral suppression (CAS) DPOAEs test. The subjects were also tested for DPOAEs after the calibration of the DPOAE stimuli signals, right before the noise exposure. Once the noise exposure started, every 15 minutes a new DPOAE measurement of about 5 minutes duration was performed with a designed system [7], or a reference system. The noise exposure was recorded with a sound level meter positioned in the room close to the test subjects' ears. A noise level around 85 dB(A) was set so that the resulting exposure inside the HPD protected ear canal was about 70 dB(A), slightly below the average MEM reflex threshold. After the 4-hours noise exposure the subjects went through the post-exposure tests which were the same as the pre-exposure tests with the addition of a DPOAE test in silent condition at the beginning of this test sequence.

4 Preliminary experimental results

4.1 Pure-tone audiometry

Measured differences between pre and post-exposure PTA HTLs are on average within an audiometer step (5 dB(HL)). Surprisingly, better hearing or in other words a lower HTL in post-exposure levels (e.g. paired Wilcoxon test mean of differences $= 4.44$ dB(HL), $p < 0.05$ for Industrial noise conditions at 3 kHz), was measured in the afternoon whether the subject was exposed to noise or not.

4.2 Stapedial reflex

No significant changes in acoustic reflex thresholds (ART) were observed for the Quiet conditions (e.g. mean of differences $= -3.13$ dB(SPL), $p > 0.1$ at 4 kHz), although the average is negative in contrast with averages for Impulsive and Constant constant conditions (e.g. mean of differences $= 2.22$ dB(SPL), $p > 0.1$ at 2 kHz for Constant noise conditions), indicating elevated ART for post-exposure where ART are slightly decreased after a regular Quiet day. This might indicate some minor effect of the circadian rhythm as reported by [6] and non significant effects of noise exposure with the current statistics approach.

4.3 DPOAEs

In general, the measured difference between afternoon and morning DPOAE measurements for the Quiet conditions are within 1 dB (e.g. mean of differences $= 0.84$ dB(SPL), p ¡ 0.05 at 4382 Hz), a magnitude similar to what is found in [6]. Therefore, the effects of TOD would be smaller and of opposite sign compared to the envisioned noise induced changes of possibly < -2 dB(SPL).

5 Discussion

Previous studies [3, 4, 6] along with the current preliminary results indicate that the influence of the circadian rhythm, if present, is relatively small and should not influence the test interpretations for standard laboratory conditions. Other factors might affect the results for subjective and objective hearing tests, such as the learning factor for subjective tests where a decrease in average HTL of 1-2 dB(HL) could be observed for PTA in the post-exposure measurements [8].

6 Conclusions

In summary, circadian rhythm effects on hearing tests are not large enough to influence the interpretation of the results in standard test conditions. Other factors might have a bigger effect and therefore should be considered first. It is not necessary to correct for this TOD effect for the upcoming extensive study of the DPOAE level variations over the test day.

Acknowledgements

Annelies Bockstael gratefully acknowledges the support of the Research Foundation-Flanders (FWO). Vincent Nadon is especially grateful to the IRSST, the Quebec occupational health and safety research institute, for its support for the project as well as the technical support from the EERS-ÉTS industrial research chair in in-ear technologies.

References

[1] National Institute for Occupational Safety and Health. NOISE AND HEARING LOSS PREVENTION. Online, 2016. https://www.cdc.gov/niosh/topics/noise/.

[2] CSST. Surdité professionnelle : le boom d'un mal silencieux. Online, 2015. http://preventionautravail.com/reportages/266-surdite-professionnelle-le-boom-d-un-mal-silencieux.html.

[3] P. Ezzatian, M. K. Pichora-Fuller, and B. A. Schneider. Do circadian rhythms affect adult age-related differences in auditory performance? *Canadian Journal on Aging*, 29(02):215–221, 2010.

[4] T. Venet, P. Campo, C. Rumeau, A. Thomas, and C. Parietti-Winkler. One-day measurement to assess the auditory risks encountered by noise-exposed workers. *International journal of audiology*, 53(10):737–744, 2014.

[5] J. J. Guinan Jr, B. C. Backus, W. Lilaonitkul, and V. Aharonson. Medial olivocochlear efferent reflex in humans: otoacoustic emission (oae) measurement issues and the advantages of stimulus frequency oaes. *Journal of the Association for Research in Otolaryngology*, 4(4):521–540, 2003.

[6] A. T. Cacace, W. A. McClelland, J. Weiner, and D. J. McFarland. Individual differences and the reliability of 2f1-f2 distortion-product otoacoustic emissions effects of time-of-day, stimulus variables, and gender. *Journal of Speech, Language, and Hearing Research*, 39(6):1138–1148, 1996.

[7] V. Nadon. DÉVELOPPEMENT D'UNE OREILLETTE POUR LA MESURE DES ÉMISSIONS OTOACOUSTIQUES. Master's thesis, École de technologie supérieure, 2014.

[8] W. Burns and R. Hinchcliffe. Comparison of the auditory threshold as measured by individual pure tone and by békésy audiometry. *The Journal of the Acoustical Society of America*, 29(12):1274–1277, 1957.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Investigation Of Aircrew Noise Exposure Onboard The Nrc Dassault Falcon Aircraft Through Flight Testing

Andrew Price, Sebastian Ghinet, Viresh Wickramasinghe

The NRC Dassault Falcon 20 is a small business jet operated by the NRC Flight Research Laboratory as a customizable research and experimental facility. As such, the NRC Falcon flies in a variety of cabin configurations and flight conditions including parabolas for microgravity. In-flight cabin noise measurements were performed on the NRC Falcon throughout a number of representative flight conditions. The data was analyzed in 1/3rd octave bands in accordance with ISO 5129 and evaluated in accordance with the Canadian Aviation Occupational Health and Safety Regulations. It was determined that all aircrew equipped with properly fitted hearing protection will not be overexposed to noise while onboard the NRC Falcon. The most severe noise exposure was experienced at the rear of the cabin near the engines during level flight while accelerating to 300 KIAS. While not equipped with properly fitted hearing protection, an occupant will reach the maximum allowed noise dose in a cumulated 9 minutes and 36 seconds under these conditions.

OCEAN AND MARINE-MAMMAL ACOUSTICS - ACOUSTIQUE DES OCÉANS ET DES MAMMIFÈRES

An Automated Real-Time Vessel Sound Measurement System For Calculating Monopole Source Levels Using A Modified Version Of Ansi/asa S12.64-2009
David Edward Hannay, Xavier Mouy, Zizheng Li 166

Blind Channel Estimation Of Underwater Acoustic Waveguide Impulse Responses Using Marine Mammal Vocalizations
Brendan P. Rideout, Eva-Marie Nosal, Anders Høst-Madsen 168

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 170

AN AUTOMATED REAL-TIME VESSEL SOUND MEASUREMENT SYSTEM FOR CALCULATING MONOPOLE SOURCE LEVELS USING A MODIFIED VERSION OF ANSI/ASA S12.64-2009

David E. Hannay*, Zizheng Li† and Xavier Mouy‡
JASCO Applied Sciences (Canada) Ltd.
2305-4464 Markham Street, Victoria, British Columbia, Canada, V8Z 7X8

1 Introduction

Underwater noise from vessels permeates many of the world's oceans. While vessel sound emissions are at levels typically below those that would be acutely injurious to marine fauna, this noise can interfere with normal use of sounds, such as for prey and predator detection, socialization and mate attraction. Exposures to vessel noise over extended times, especially in key habitat areas, is likely to lead to chronic adverse effects including reduced feeding efficiency and difficulty finding mates.

Assessments of the effects of shipping noise on marine fauna often use acoustic propagation models to predict the levels of sound exposure. These models require accurate vessel sound emission source levels. American National Standards Institute (ANSI) standard S12.64-2009 (reaffirmed in 2014) describes procedures for measuring underwater sound from ships. The standard deals with radiated noise level (RNL) source levels that assume 20 Log(r) transmission loss (TL) between the vessel positions and the measurement hydrophones. That approach does not account for interference from surface and seabed reflections. Most acoustic models directly account for these effects, and therefore require monopole source levels (MSL). MSL assumes all acoustic energy originates at a single point in the water, at a specified depth.

Few fully systematic measurements of source levels of large commercial vessels are available. A few recent studies have published source levels obtained from large numbers of vessel passes, but those measurements are typically partly opportunistic, with hydrophones in shallow water or located several kilometers distance from the vessel paths. Also, most of the existing published measurements report RNL but not MSL.

The Strait of Georgia Underwater Listening Station (ULS) and JASCO Applied Science's PortListen™ processing software were designed to obtain systematic measurements of large numbers of vessels in relatively deep water (173 m). The system reports RNL measurements in approximate conformance with ANSI/ASA S12.64 (2009) Grade-A processing but with Grade-C geometry and noting only single vessel passes are acquired per transit. MSL calculations are made similarly but with a modified back-propagation method.

* David.Hannay@jasco.comca
† Xavier.Mouy@jasco.com
‡ Zizheng.Li@jasco.com

2 Methods

2.1 Acoustic Monitoring Equipment

Two seabed-mounted tetrahedral arrays, each with 4 calibrated GeoSpectrum M-15 hydrophones, at 1.5 m separation (Figure 1) and a calibrated GeoSpectrum projector, are deployed on Ocean Network Canada (ONC)'s Strait of Georgia East Node. The arrays are separated by approximately 43 m. All 8 hydrophones are sampled at 64 kHz (24-bit) by JASCO AMAR-G3 recorders. Synchronized data are streamed to shore in near real-time (5-minute files) via the VENUS undersea network.

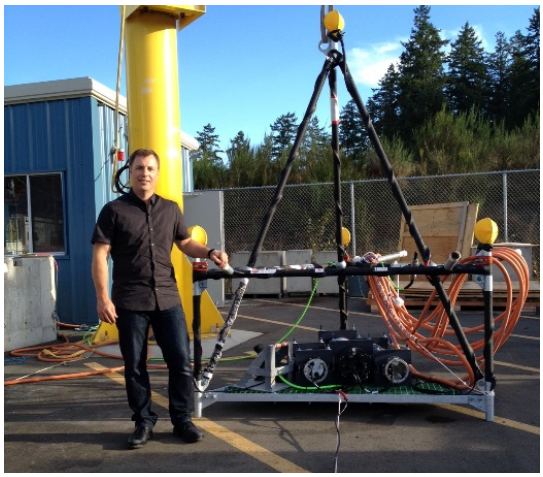


Figure 1. One of two tetrahedral hydrophone arrays comprising the Underwater Listening Station (ULS).

The systems were deployed on September 21, 2015.All components remain fully functional and in-calibration as of August 1, 2016.

2.2 Location, Vessel Paths and Tracking

The ULS is deployed on the seabed in 173 m depth, along the northbound shipping route, approximately 30 km southwest of Vancouver BC. Vessels are tracked with a dedicated Automatic Identification System (AIS) deployed on-shore near the ULS. Vessel pilots are requested to sail vessels through a measurement transit area, consisting of an entrance funnel, measurement zone and exit zone (Figure 2). This approach ensures that the vessels are not turning or accelerating during the measurement. The measurement zone (shaded region in Figure 2) allows closest point of approaches (CPA)'s from 250-550 m.

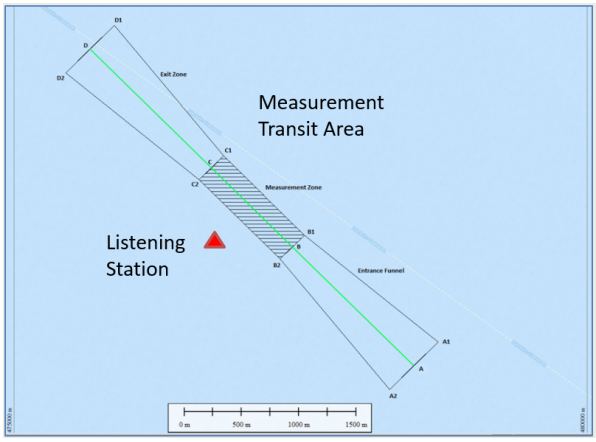


Figure 2. ULS and Vessel Measurement Transit area.

2.3 Source Level Calculation Software System

Acoustic data and AIS data are processed automatically in real-time by the JASCO Applied Science's PortListen™ software. The system uses a combination of AIS and acoustic tracking to extract a data window, defined by the time period over which the vessel's acoustic centre is within ±30° of the CPA. It then processes a single acoustic channel in 1 second time periods stepped in ½ second intervals (to allow 50% overlap of Hanning time windowing prior to fft's).

The software applies the ANSI/ASA S12.64 Grade-A method for back-propagation distance: it determines instantaneous vessel range (R) each 1-second time step through the data window. The RNL back propagation of 20 Log(R) is applied to the band levels of each step. MSL back-propagation is performed each second with predictions from Parabolic Equation model RAM, modified to treat shear wave reflection losses up to 5 kHz, and an image reflectivity model at higher frequencies. MSL back-propagation requires a source depth, which we define as a Gaussian distribution centred at the shaft depth minus 0.7 of the propeller radius, when that information is available, or half the vessel draft otherwise.

The software calculates background noise in each frequency band when the vessel is more than 2 km away. The measured source band levels are accepted only if they exceed the background levels by 10 dB or more. The levels are corrected if they exceed background by 3-10 dB, and they are rejected if they are less than 3 dB above background.

3 Results and Discussion

The ULS made 846 vessel measurements of vessel passes with CPA's between 250-550 m between September 21, 2015 and July 31, 2016. Of these 304 were accepted after automatic and manual review, 249 were rejected, and 293 have not yet been reviewed. The system processes a larger set of CPA's, from 100-1000 m. There were 1623 measurements over this larger range, of which 524 were accepted, 511 rejected, and 588 not yet checked.

3.1 Vessel Noise Directivity

The system produces 1/3-octave band levels in 1 second steps through the duration of the ±30° data window. Directivity features, such as higher high-frequency levels post-CPA than pre-CPA, are apparent in many of the measurements. The system likely can produce more direction-dependent emissions information than S12.64-2009 defines.

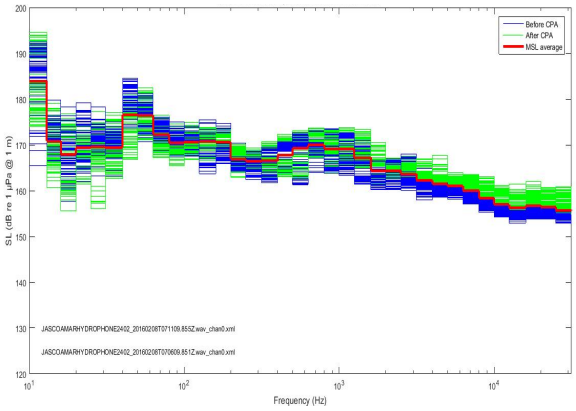


Figure 3. 1/3-octave MSL each second through the duration of the ±30° data window.

3.2 MSL Versus RNL

MSL's in 1/3-octave bands are generally greater than RNL at low frequencies (< ~100 Hz), asymptotically tending to RNL–3 dB at high frequency. Lower RNL at low frequencies is due to destructive interference from the sea-surface reflection that is not accounted for by 20 log(R) back-propagation. Higher RNL at high frequencies is due to the incoherent contribution of surface-reflected energy. A typical vessel measurement example from PortListen™, showing both MSL and RNL, is given in Figure 4.

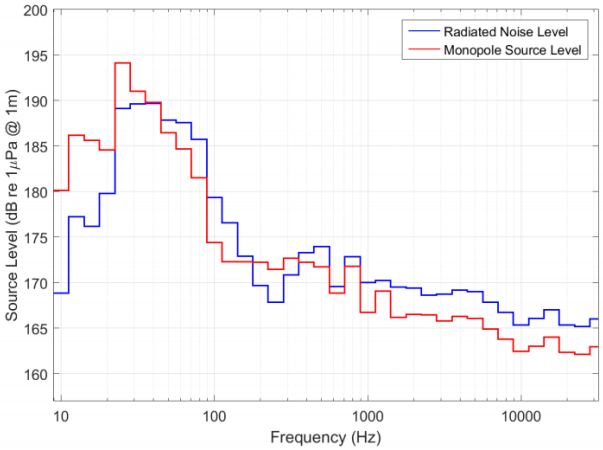


Figure 4. MSL (red) and RNL (blue) 1/3-octave band source levels measured by the ULS.

A large database of systematic source level measurements has been compiled. Analysis of these data is just starting.

BLIND CHANNEL ESTIMATION OF UNDERWATER ACOUSTIC WAVEGUIDE IMPULSE RESPONSES USING MARINE MAMMAL VOCALIZATIONS

Brendan P. Rideout[*1], **Eva-Marie Nosal**[†1], **and Anders Høst-Madsen**[‡2]
[1]Department of Ocean & Resources Engineering, University of Hawaii at Manoa, 2500 Campus Road, Honolulu, Hawaii, USA
[2]Department of Electrical Engineering, University of Hawaii at Manoa, 2500 Campus Road, Honolulu, Hawaii, USA

1 Introduction

This paper presents results from an ongoing research project aimed at improving the reliability of tracking underwater marine mammals. In particular, this project aims to expand the usefulness of specific high-performance tracking approaches (e.g., those using time of arrival-based localization) to a wider range of species, including humpback whales, than is currently possible. This expansion will facilitate the identification of critical habitat for marine sanctuaries and allow sound-producing oceanic activities, including scientific, industrial, and military endeavors, to be more responsibly conducted.

A sound that is heard underwater is a function of both the sound made by the source and the environment the sound passes through in traveling from source to receiver. These environmental influences include ocean surface wave conditions, ocean bottom sediment properties, and relative source & receiver locations. The net effect of all these influences is called the impulse response for the acoustic channel between the source and receiver. Knowledge of the underwater acoustic impulse response permits source localization, and potentially ocean environmental parameter estimation, in many cases. In this work, a technique called Blind Channel Estimation (BCE) is employed to estimate the impulse responses, where blind refers to the estimation algorithm not requiring *a priori* knowledge of the source waveform. BCE was originally conceived as a way to improve cell phone reception as the user moves through a complicated environment (e.g., between tall buildings).

2 Theory

This section will present a brief development of the theory behind BCE for a one source/two receiver case. Vectors (e.g., $\mathbf{h}_i$) and matrices (e.g., $\mathbf{X}$) are shown in lower and uppercase boldface characters, respectively. This approach is generalizable to any number of receivers.

For the one source/two receiver case, as shown in Xu et al. (1995), the received (i.e., sampled) waveforms ($\mathbf{x}_i$ and $\mathbf{x}_j$) are expressed as

$$\mathbf{x}_i(k) = \mathbf{h}_i(k) \otimes \mathbf{s}(k)$$
$$\mathbf{x}_j(k) = \mathbf{h}_j(k) \otimes \mathbf{s}(k)$$

where $\mathbf{h}_i(k)$ is the impulse response for the i^{th} receiver, and $\mathbf{s}(k)$ is the source waveform. These two equations can be expressed as

$$\mathbf{h}_i(k) \otimes \mathbf{x}_j(k) = \mathbf{h}_i(k) \otimes \big(\mathbf{h}_j(k) \otimes \mathbf{s}(k)\big)$$
$$\mathbf{h}_i(k) \otimes \mathbf{x}_j(k) = \mathbf{h}_j(k) \otimes \big(\mathbf{h}_i(k) \otimes \mathbf{s}(k)\big)$$
$$\mathbf{h}_i(k) \otimes \mathbf{x}_j(k) = \mathbf{h}_j(k) \otimes \mathbf{x}_i(k)$$
$$\mathbf{h}_i(k) \otimes \mathbf{x}_j(k) - \mathbf{h}_j(k) \otimes \mathbf{x}_i(k) = 0$$

This convolution relationship can be expressed as a system of linear equations:

$$\begin{pmatrix} \mathbf{X}_j(L) & -\mathbf{X}_i(L) \end{pmatrix} \begin{pmatrix} \mathbf{h}_i \\ \mathbf{h}_j \end{pmatrix} = 0$$
$$\mathbf{X}(L)\mathbf{h} = 0 \tag{1}$$

where

$$\mathbf{X}_j(L) = \begin{pmatrix} x_j(L) & x_j(L+1) & \dots & x_j(2L) \\ x_j(L+1) & x_j(L+2) & \dots & x_j(2L+1) \\ \vdots & \vdots & \ddots & \vdots \\ x_j(N-L) & x_j(N-L+1) & \dots & x_j(N) \end{pmatrix}$$

$$\mathbf{h}_i \equiv [h_i(L), \dots, h_i(0)]^T$$

and $L+1$ is the length of the longest IR in the set.

Due to the ambiguities inherent to the problem formulation in Equation (1), BCE is unable to resolve the start time and scaling factor for the channels being estimated. In acoustical terms, there is magnitude and phase ambiguity in the solution. The literature presents at least two possibilities to resolve this ambiguity. In Xu et al. (1995), the constraint $|\mathbf{h}|_1 = 1$ is presented. This addresses the scaling factor but not the phase ambiguity. In Zeng et al. (2013), the ℓ^{th} entry in $\mathbf{h}$ is defined to be equal to 1. All other values of $\mathbf{h}$ are estimated relative to this value. To achieve this, the ℓ^{th} column of $\mathbf{X}$ is defined as $\mathbf{b}$ and ℓ is set equal to some constant (frequently, $(L+1)/2$). Thus, the governing equation becomes (for the two channel case):

$$\mathbf{A}\mathbf{c} = \mathbf{b} \tag{2}$$

where

$$\mathbf{c} = [[h_1(L), \dots, h_1(\ell+1), h_1(\ell-1), \dots, h_1(0)], \mathbf{h}_2^T]$$

$\mathbf{b}$ is the ℓ^{th} column of $\mathbf{X}$, and $\mathbf{A}$ is obtained by removing the l^{th} column of $\mathbf{X}$.

Using Equation (2) as a starting point, the blind channel estimation problem can be formulated as a quadratically

[*]bprideou@hawaii.edu
[†]nosal@hawaii.edu
[‡]ahm@hawaii.edu

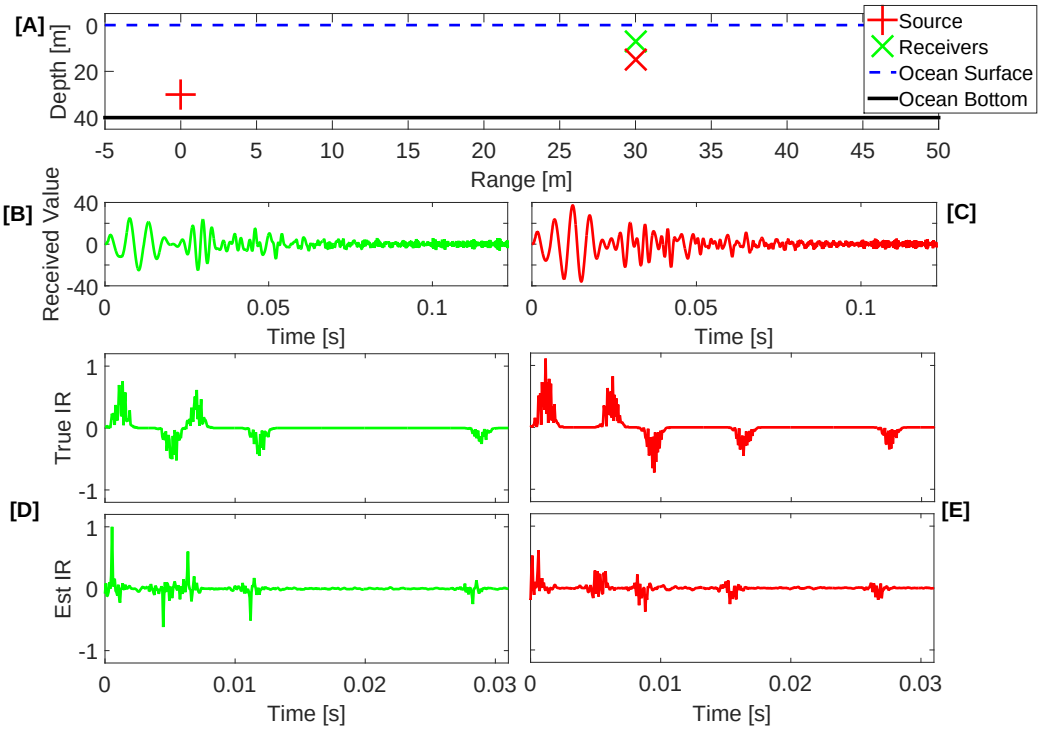


Figure 1: BCE performance under simulated ocean conditions. [A] Simulation conditions, with a uniform water depth of 40 m, a single source 30 m below the surface, and two receivers 30 m away from the source and at depths of 7 and 15 m, respectively. Received waveforms at the top and bottom are shown in [B] and [C], respectively. The true and estimated impulse responses for the top receiver are shown in [D], while those for the bottom receiver are shown in [E].

constrained ℓ_1-minimization problem (as per Equation 1.2 in Becker (2011)):

$$\min \| c \|_{\ell_1} \text{ subject to } \| \mathbf{Ac} - \mathbf{b} \|_{\ell_2} \leq \epsilon \qquad (3)$$

where $\mathbf{c} \epsilon R^p$ and $p = 2 * (L + 1) - 1$ (for the two channel case). This type of problem can be solved using established optimization approaches (e.g., Becker (2011)).

3 Simulation Results

To test the performance of the BCE approach using realistic marine mammal vocalizations, a set of simulations were carried out. For this paper, the source is a frequency sweep (i.e., a chirp) from 100-4300 Hz, modeled after a humpback whale call recorded offshore of Maui in May 2015. Figure 1 shows the results of this simulation. The ocean environment used for this simulation is shown in subplot **[A]**. The chirp is observed by the shallow and deeper simulated receivers as shown in **[B]** and **[C]**, respectively. The estimated, and true, impulse responses for each of the two channels are shown in **[D]** and **[E]**. The good match between true and estimated channels suggests that blind channel estimation can reveal enough about the impulse response to allow the sound source to then be located. It is likely that some environmental conditions and source characteristics are better suited for estimating the underlying impulse responses than others. For instance, if the underlying channels have significant time-varying components (e.g., due to ocean surface waves or relative motion between the source and receivers) the estimated impulse responses tend to be less informative.

4 References

[1] G Xu, H Liu, L Tong, and T Kailath. A least-squares approach to blind channel identification. Signal Processing, IEEE, 43(12):2982–93, 1995.

[2] Wen-Jun Zeng, Xue Jiang, and Hing Cheung So. Sparse-representation algorithms for blind estimation of acoustic-multipath channels. The Journal of the Acoustical Society of America, 133(4):2191–7, 2013.

[3] S Becker. Practical Compressed Sensing: modern data acquisition and signal processing. PhD thesis, California Institute of Technology, 2011.

Acknowledgments

This work was supported in part by the National Science Foundation.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Efficient Estimation Of Range-Dependent Seabed Properties From Large Data Volumes Of A Towed Source And Receiver-Array System

Jan Dettmer, Jorge E Quijano, Stan E Dosso, Charles W Holland, Eric Mandolesi

Knowledge of geophysical seabed properties is important for shallow-water sonar applications, including detection and classification of unexploded ordnance. However, state-of-the-art surveying methods such as seismic profiling, coring, or acoustic inversion are of limited use when surveying large areas with high spatial sampling density. We consider a new acoustic survey method based on a towed source and receiver array which produces large volumes of seabed reflectivity data that contain unprecedented and detailed seabed information. These data can be analyzed by inversion, but require efficient computation of reflection coefficients, efficient inversion algorithms and efficient use of computer resources. This work applies a particle filter to quantify information content of multiple data sets by considering results from previous data along the survey track to inform the importance sampling at the current point. Challenges arise from rapid environmental changes along the track where the complexity of sediment layers and their properties change. This is addressed by including trans-dimensional steps in the filter which allow the layering complexity to change along a track. Efficiency is improved by tempering the likelihood function of particle subsets and including exchange moves (parallel tempering). The filter is implemented on a hybrid computer that combines central processing units (CPUs) and graphics processing units (GPUs). The algorithm exploits three levels of parallelism: (1) parallel computation of spherical reflection coefficients with a GPU implementation of Levin integration; (2) updating particles by concurrent CPU processes which exchange information using automatic load balancing; (3) overlapping CPU-GPU communication (a major bottleneck) with GPU computation by staggering CPU access to the multiple GPUs (via multi process service). The algorithm is applied to simulated spherical reflection coefficients for 170 data sets along a 7-km track. We demonstrate substantial efficiency gains over previous methods, providing uncertainty quantification for 100+ data sets per 24 hours.

Automatic Detection And Visualization Of Marine Mammal Sounds In British-Columbia, Canada.

Xavier Mouy, Pierre-Alain Mouy, David Hannay

Planned developments of marine terminals along the British Columbia (B.C.) coast will lead to increased exposures of marine mammals to noise from vessel traffic. Monitoring marine mammals off B.C. is becoming increasingly important for assessing their interactions with anthropogenic activities. Over the last decade, a large number of autonomous and cabled hydrophones have been deployed in B.C. waters to detect marine mammal calls and to measure ambient noise levels. These systems collect large data volumes that require considerable time and effort to fully analyze and interpret. That often precludes a comprehensive analysis, and data are commonly archived before their full value is exploited. In this study we describe techniques to automatically detect, classify, and visualize sounds produced by several species of cetaceans that frequent B.C. waters. Blue and fin whale calls are detected using a spectrogram correlation approach. Killer whale, humpback whale and Pacific white sided dolphin vocalizations are detected by calculating the local variance of intensity in the spectrogram and then classified using a random forest classifier. A web visualization interface (PAMview) is used to easily navigate through, display, and share detection and classification results. This interface is organized as three interconnected visualization panels: 1) a geographic interface displays maps of the total number of detections for each species at all monitoring locations within an adjustable time period, 2) an interactive detection time series displays temporal varia-

tions of detections for several species at a given monitoring location, and 3) a multimedia panel allows the user to visualize spectrograms, listen to sounds and to validate automatic detections. A demonstration of the acoustic monitoring system developed will be performed using archival and real-time data from the VENUS and NEPTUNE ocean observatories.

Passive Acoustic Localization And Density Estimation Of Fin Whales Using A Single Hydrophone

Brendan P. Rideout

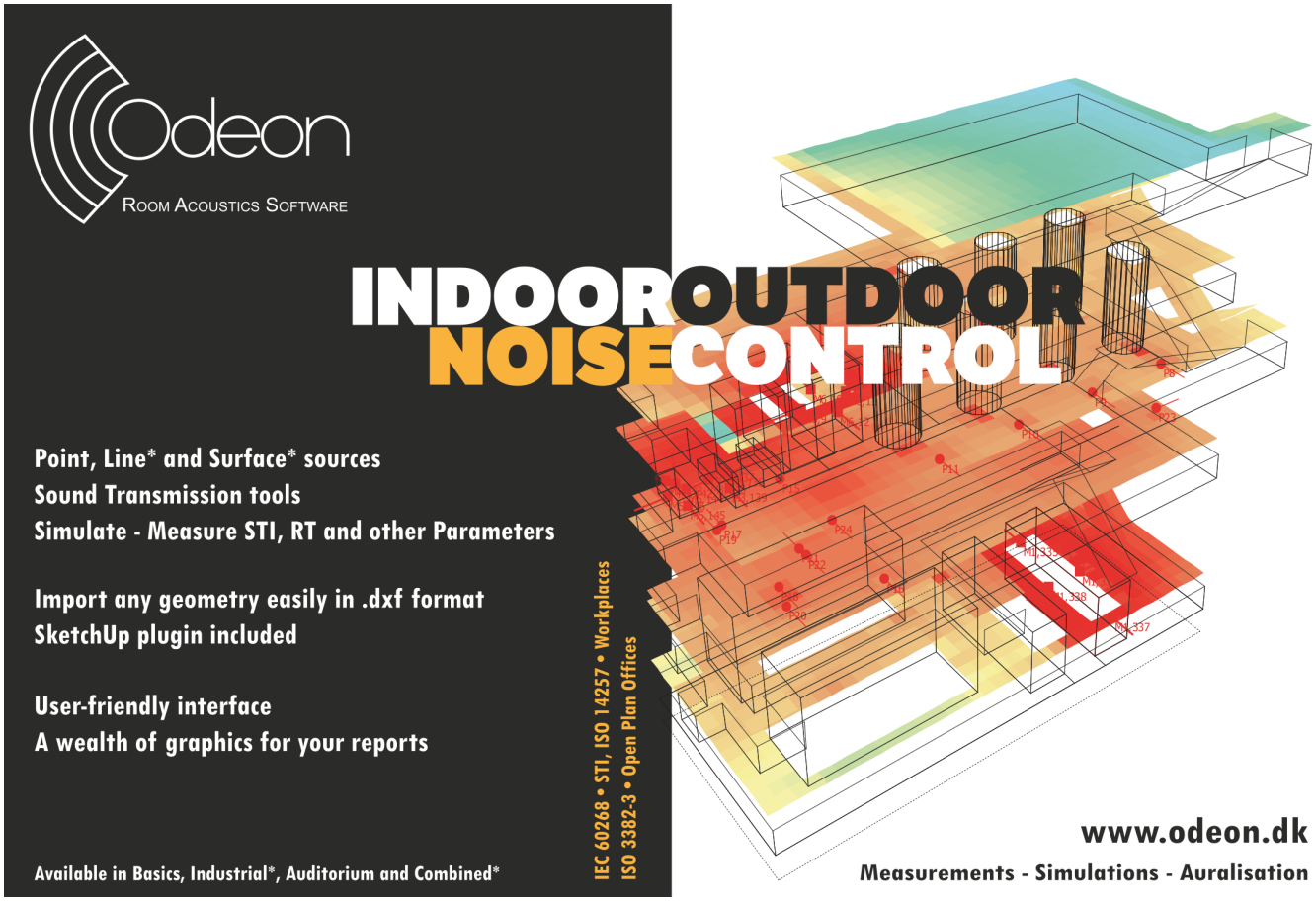
This paper presents an approach for carrying out passive underwater acoustic localization of a vocalizing marine mammal with data from a single receiver, and aims to build upon earlier work on localization based on the time of arrival of acoustic energy traveling either directly from source to receiver, or bouncing one or more times off the ocean surface and/or bottom. This localization approach will be applied to a multi-year data set consisting of thousands of fin whale vocalizations, necessitating an approach with a high degree of automation. This approach uses a comparison between the recorded fin whale vocalizations and model-predicted acoustic waveguide impulse responses at a set of candidate ranges and depths relative to the receiver. The location whose estimated impulse response best matches the recorded fin whale call is deemed to be the estimated source position. A small number of calls distributed throughout the data record were localized individually using a linearized Bayesian localization approach with a quantitative uncertainty analysis incorporating both data and environmental uncertainties. The data for this study were recorded by the single hydrophone in the ALOHA cabled observatory, located 100 km N of Oahu in 4728 m of water. Repeated hydrographic, chemical, and biological sampling has been done at this site since 1988, facilitating acoustic propagation modeling in the vicinity of the ALOHA site. The primary motivation for this work is to estimate the local population density of fin whales observed near the ALOHA cabled observatory.

Interaction Between Learned And Naturally Selected Features In The Design Of Killer Whale Calls And Their Functions

Harald Yurk

All cetaceans, and particularly all dolphins, rely heavily on acoustic signalling to survive and reproduce. The killer whale or orca is the largest dolphin and occurs in all ocean basins. Killer whales produce stereotyped calls with complex temporal and frequency structure that allow spectrographic distinction into different call components. Call components differ in directionality and attenuation and may be used to synchronize travelling and foraging behaviours by allowing recognition of whales at great distances and identification of the direction of approaching and moving senders. Killer whale communities or populations that avoid social contact may still occupy the same areas (sympatry). Members of these populations are distinguishable by ecological and social differences such as are prey preferences (fish versus mammals) and social behaviours. Mutually unintelligible call repertoires seemingly are drivers for social segregation. Calls appear to be learned traits transmitted from parent generation to offspring generation in stable matrilineal groups which form the basis of all well studied populations. This preference for social learning is the basis for stable long lasting vocal traditions or dialects and may also influence other stable behaviours such as prey preferences. The stability of dialects can be traced through the observation of acoustic divergence among groups: high dialect similarity among socializing and closely related groups diminishes with social and genetic distance leading to diverse and discrete call dialects among interbreeding clans and among non socializing, non-interbreeding communities. This pattern of selective social learning appears to be the root cause for distinct dialects which allows mate selection to avoid inbreeding and fosters cooperation among clan and community members while it also maintains social segregation and genetic isolation between non-cooperating mammal eating and fish eating populations. It is possible that call

structure design together with call dialects allow killer whales to be highly efficient foragers living in tight knit genetically viable social communities at higher densities in a given area than would be possible without ecological segregation.



OTHER TOPICS AND DISCUSSION - AUTRES SUJETS ET DISCUSSIONS

The Lasso And Elastic-Net Algorithms For Predictive Sound Quality Models Using Large Pool Of Predictive Metrics And Factors
Abdelghani Benghanem, Philippe-Aubert Gauthier, Lévy Leblanc, Alain Berry 174

The Case For A Canadian Trade Association For Acoustic Consulting Companies
Eric De Santis, Steve Meszaros 176

Effect Of Variability In Micro-Geometry Of Polyurethane Foams On The Double Wall Transmission Loss
Mohammad Sadegh Gholami, Olivier Doutres, Noureddine Atalla 178

THE LASSO AND ELASTIC-NET ALGORITHMS FOR PREDICTIVE SOUND QUALITY MODELS USING LARGE POOL OF PREDICTIVE METRICS AND FACTORS

Abdelghani Benghanem [*1], **Philippe-Aubert Gauthier** [†1,2], **Lévy Leblanc** [‡1], **and Alain Berry** [§1,2]
[1]Groupe d'Acoustique de l'Université de Sherbrooke, Sherbrooke, Québec, Canada
[2]Centre for Interdisciplinary Research in Music, Media and Technology, McGill University, Montréal, Canada

1 Introduction

Part of the perceived quality is communicated by sound quality (SQ). SQ studies present a twofold challenge [1]. The first is that they rely on listening tests. However, listening tests are considered time-consuming. The second challenge relates to the development of predictive SQ models. Often, these prediction models are derived using linear regression on a limited set of predictive metrics. However, nowadays, many metrics are available and there is no computational burden that should limit the number of potential metrics. An issue is that regression using more metrics than observations cannot lead to meaningful predictive models since all the metrics are selected. In this paper, different algorithms are compared to construct a SQ predictive model that does not suffer from these limitations. These algorithms achieve an automatic selection of few metrics from a large pool. The lasso [2] and elastic-net [3] are tested for the prediction of listening tests results of consumer product.

2 Listening tests: Method

Listening tests were conducted with 40 participants according to a forced-choice pairwise comparison method [1]. Tests were conducted using calibrated and equalized binaural reproduction over headphones. The test was based on a full-fractional design of experiments (DOE) with four factors (two levels). The sound samples were based on a 5-second binaural sound that has been transformed into 16 different sounds according to the DOE using parametric time-varying equalizers. Levels were corresponding to increased or decreased content corresponding to factors. The participants controlled the playback and their selection with a user interface. The pairwise comparison tests included 120 pairs that were compared with random permutation each time.

3 Lasso and elastic-net algorithms

The aim of SQ studies is the derivation of predictive SQ models based on results of listening tests (herein the merit values v_i of sound sample i) and potential predictive factors or metrics derived from the actual sound samples or from DOE factors. A common form of predictive SQ model is expressed as follow for each of the I merit values

$$v_i = \sum_{m=1}^{M} F_{im} b_m + b + e_i \qquad (1)$$

with merit values v_i obtained from listening test statistics, matrix of M factors and/or metrics F_{im}, M linear regression coefficients b_m, b is the intercept and e_i is the prediction error. Note that the matrix of factors and predictors F_{im} can not only include factors and metrics, but interactions of factors and non-linear functions of metrics (squared, log, etc.). In this case, although the model of expression Eq. (1) is linear, the actual SQ model is not. However, Eq. (1) can still be solved with linear algorithm: using linear regression. For the case of *wide data* with much less observations than predictive metrics (i.e. $I << M$), classical linear regression algorithms will typically achieve over-fitting and all the coefficients b_m will be non-zero. Hence limiting the interpretable meaning of the resulting model for any engineering uses or design guidelines. The aim of the lasso and elastic-net algorithm is to circumvent this issue by rewriting the problem as a composite cost function:

$$J_{\lambda,\alpha} = \frac{1}{2I}\left(\sum_{i=1}^{I}\left(v_i - b - \sum_{m=1}^{M} b_m F_{im}\right)\right)^2 + \lambda \sum_{m=1}^{M}\left(\frac{(1-\alpha)}{2} b_m^2 + \alpha |b_m|\right) \qquad (2)$$

The first right-hand side term corresponds to the quadratic sum of the predictor errors and the second right-hand side term is a regularization term with regularization amount $\lambda > 0$. The regularization combines 2-norm regularization b_m^2 and 1-norm regularization $|b_m|$ based on the elastic-net parameter $0 \leq \alpha \leq 1$. For the lasso algorithm, $\alpha = 1$, only the 1-norm regularization is included. This induces solution sparsity, i.e. few coefficients b_m will be non-zero and selected in the model [2]. The sparsity is controlled by the regularization amount λ. The elastic-net algorithm [3] involves $0 < \alpha < 1$ and it will typically smooth out the selection towards the extreme case of 2-norm-only regularization with $\alpha = 0$ (this corresponds to Tikhonov regularization). The minimum of the composite cost-function cannot be found by analytical means since it is non-linear. Therefore, this is solved iteratively using coordinate-descent algorithm. α is a user-selected parameter. The penalization amount λ must be set rigorously: The experimenter defines a limit on the maximum number of metrics that should be included. Next, the problem is solved for a wide range of λ starting with the largest λ so that all coefficients b_m are zeros. Then the λ iteration stops if: 1) the maximum number of non-zero coefficients b_m is reached or 2) the minimum prediction error is obtained for k-fold cross-validation of the prediction. In this paper, the maximum number of predictive metrics is 10 and the cross-validation is 8-fold.

*. Abdelghani.Benghanem@USherbrooke.ca
†. philippe_aubert_gauthier@hotmail.com
‡. Levy.Leblanc@USherbrooke.ca
§. Alain.Berry@USherbrooke.ca

4 Factors and metrics

Various predictors F_{im} were used. The four DOE factors (namely $F1$, $F2$, $F3$, and $F4$) were included. Next, interactions of factors were also included ($F1F2$, $F1F3$, etc. for a total set of 14 predictive interaction factors). Spectral informations was gathered as predictive metrics: 1) third-octave-band levels, 2) specific loudness on Bark scale. Finally, other metrics and related time statistics (mean value, standard deviation, and slope) were included using the MIR Toolbox (Music Information Retrieval) [4]. All metrics and potential predictors were centered and normalized before the construction of the predictive SQ model. The pool of predictors included 141 metrics and factors.

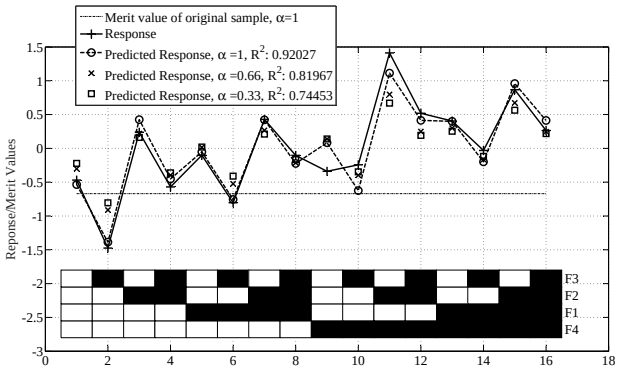
5 Post-processing and results

Once listening tests were completed, few post-processing steps were done: 1) participants clustering using a Gaussian mixture model in order to extract the most coherent cluster of participants, 2) one-way ANOVA was performed, the obtained p-value is 9.67e-31 which is very low, hence validating the test (i.e. the presented sound samples had an effect), and 3) the merit values v_i were obtained based on the probability matrix and Gumbel distribution [1].

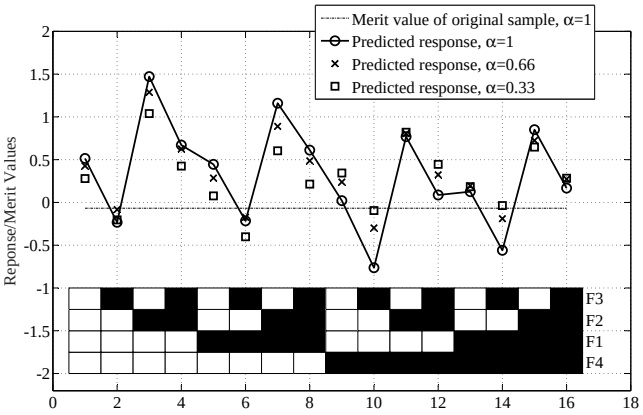
Merit values v_i and predicted merit values are reported in Fig. 1(a) for the lasso and elastic-net algorithms. Excellent predictions of the merit values are observed. For the lasso, only 4 metrics were selected. For the elastic-net, 5 or 8 predictors were selected in the final SQ model for $\alpha = 0.66$ and $\alpha = 0.33$, respectively. The three algorithms predict very similar merit values, the major difference between the algorithms is not related to prediction accuracy but with the number of selected predictors. It is worth noting that for the lasso (which provides the most sparse predictor selection) none of the DOE factors were selected, i.e. signal processing and psychoacoustic metrics are judged as more appropriate for the predictive SQ model. An illustrative example for which the usefulness of the prediction model is shown in Fig. 1(b). The three obtained predictive SQ models for Fig. 1(a), as obtained for the described listening tests, were applied to another set of 16 sounds based on a different original sound (same kind of product, different model) that has been through modifications as reported for the DOE.

6 Conclusions

The lasso and elastic-net algorithms were compared for the prediction of listening tests results of consumer product for which 141 metrics and DOE factors were used as predictors. It was shown that the most promising algorithm is the lasso which is able to efficiently and strictly limit the number of metrics and provide a model that can be used in order to provide a set of understandable design guidelines. Typically, the elastic-net includes more predictors in the predictive SQ model.



(a) v_i and predicted v_i for the 16 sound samples used in listening tests.



(b) Predicted v_i of the new 16 sound samples.

Figure 1: (a) Merit values and predicted merit values for the predictive SQ models using the lasso ($\alpha = 1$) and elastic-net ($\alpha = 0.33$ and $\alpha = 0.66$) for the 16 sound samples used in listening tests. (b) Predicted merit values of the new 16 sound samples for the predictive SQ models using the lasso ($\alpha = 1$) and elastic-net ($\alpha = 0.33$ and $\alpha = 0.66$) algorithms. Factors of the DOE are shown as black and white patches. Black and white patches mean high and low levels, respectively.

Acknowledgments

This work was financially supported by the National Sciences and Engineering Research Council of Canada, BRP and CTA for the dSkibel research project.

References

[1] N.C. Otto and G.H. Wakefield. A Subjective Evaluation and Analysis of Automotive Starter Sounds. *Noise Control Engineering Journal*, 41(3):377, 1993.

[2] R. Tibshirani. Regression Selection and Shrinkage via the Lasso. *Journal of the Royal Statistical Society*, 58:267–288, 1994.

[3] H. Zou and T. Hastie. Regularization and variable selection via the elastic-net. *Journal of the Royal Statistical Society*, 67:301–320, 2005.

[4] P. Lartillot O., Toiviainen. A Matlab Toolbox for Music Feature Extraction from Audio. *Proceedings for the 10th International Conference on Digital Audio Effects (DAFx-07), Bordeaux, France*, 2007.

THE CASE FOR A CANADIAN TRADE ASSOCIATION FOR ACOUSTIC CONSULTING COMPANIES

Eric de Santis, M.Sc. P.Eng. [*1], **Steve Meszaros, M.Sc. P.Eng.** [†2]
[1]BAP Acoustics Ltd., 201A – 101 Klahanie Drive, Port Moody, BC, Canada V3H 0C3
[2]RWDI Air Inc., 280 – 1385 West Broadway, Vancouver, BC, V6H 3V9

1 Introduction

The demand for acoustic consultancy services in Canada has been steadily rising over the past several decades. This increased demand for our professional services has likely been spurred by new or updated noise regulation, contractual acoustic requirements for Public Private Partnership projects, and perhaps a greater awareness of the value of noise and its effects on health, in general.

The authors of this paper and their respective consultancy firms advocate the case for the creation of a trade association to represent and promote our growing industry and to protect the public. Similar trade associations for acoustic consulting companies exist in the United States, the United Kingdom and Australia. These associations include the US National Council of Acoustic Consultants (NCAC), the UK Association of Noise Consultants (ANC) and the Association of Australian Acoustical Consultants (AAAC). Reference is made to these associations in discussing the benefits of the proposed Canadian trade association for acoustic consulting companies.

2 Benefits of Association

2.1 Protection to the Public

One of the primary objectives of the proposed association would be to protect the public from unqualified practitioners. This objective can be realized by screening applicant member firms according to the qualifications of the principals, partners and directors (hereafter referred to as principals to avoid unnecessary repetition) taking technical responsibility for firm (office, or division as the case may be for larger corporations) and general staff. The ANC, for example, requires that the principals of the firm be competent, practicing acousticians. The ANC also requires that at least 50% of technical staff hold graduate status and hold Corporate Membership in the Institute of Acoustics (IOA) or equivalent. Both the NCAC and AAAC take a similar approach in accepting member firms by reviewing the qualifications and experience of firm principals and by requiring that principals hold full membership with their respective national acoustic academic associations.

[*] eric@bapacoustics.com
[†] Steve.Meszaros@RWDI.com

However, the AAAC goes one step further in requiring that consultancies also demonstrate ongoing economic and professional viability by having been in practice for at least two years. In the author's opinion, and within the Canadian context for the proposed trade association, it would make sense for acoustic principals to hold registration with the relevant provincial Association of Professional Engineers and Geoscientists or by demonstration of relevant experience and ethical conduct (to be evaluated by a committee) in addition to holding membership with the Canadian Association of Acoustics (CAA).

A suitably drafted Code of Ethics for the proposed association would also serve the public well by requiring member firms to practice with the highest ethical and professional standards. While the details of the Code of Ethics are yet to be established, the authors envision clauses in relation to professional standards, client confidentially and disclosure, competence, conflict of interest and fair competition (particularly in relation to the review of another member's work).

In Canada, it is quite common for municipal, provincial, national jurisdictions or Crown authorities to source a single consultant to propose guidelines and/or legislation for the control of noise in buildings and the environment. New regulation can have a profound influence on both our practice and the community. In a similar fashion to both the ANC and the AAAC, and in line with the proposed trade association's mandate to protect the public, it would be ideal for the association to also liaise with these authorities to comment on proposals for new guidelines, standards and legislation. This would provide an opportunity for all member firms to provide feedback on proposed legislation.

2.2 Advancing the Industry

While acoustic consulting firms compete with each other for market share, it is important to recognize that all firms have a common interest in seeing an overall increase in the market itself. That is, all members have an interest in seeing a greater demand for acoustic consulting services. Aside from public protection, the proposed trade association can assist in advancing this demand.

The introduction of new or updated regulation is an effective means of stimulating an increase in industry demand. Currently, the National Building Code provides a minimum requirement for the control of airborne sound transmission within residential multi-family buildings. Unfortunately, there are no code requirements for the

control of structureborne sound transmission, interior noise resulting from both internal and external sources of noise or to control noise from building services to neighbouring properties. Local by-laws that attempt to fill some of these gaps are inconsistent and often weak or unenforceable.

In response to a similar building code situation in Australia, members of the AAAC implemented the Star Rating System [1] to promote a better standard of acoustic quality in multi-family buildings. The authors envision the development of a similar Canadian building acoustics rating that could be promoted to local planning authorities. In the authors' opinion, however, it would be prudent to devise an acoustic rating system that reflects the manner in which a multi-family building is marketed (e.g., luxury, comfort, standard).

Rating schemes and general guidelines could also be developed and promoted for other applications (e.g. schools, hospitals offices etc.).

2.3 Guidelines for Professional Practice

There are many instances where acoustic consultants must reference acoustic criteria from international publications or from jurisdictions outside of their project's jurisdiction. For example, when assessing noise bylaw compliance on behalf of a municipality that has not adopted quantitative noise bylaw limits, consultants may need to reference noise limits adopted by other municipalities, potentially outside of the province. In certain instances, municipal noise limits may not be appropriate, and in this case the consultant would need to reference guidance offered by international publications. It would be preferable to reference a Canadian organization that has either published a relevant guideline or endorsed one that has been approved by the membership. Both the AAAC and the ANC have published guidelines for a variety of applications.

2.4 Market Reporting

Little to no information is available on the acoustic consultancy market in Canada. As one of the main objectives of the trade association would be to promote the industry, it would be useful for the association to keep track of and publish market information to member firms. Revenue information, ideally broken down in specific fields of practice (e.g., building acoustics, environmental) could be offered by member firms via a third party independent accounting firm (i.e., a firm not employed by either the association or by the reporting member firm) in a manner such that the association, or its members, would not be able to determine the revenue of a specific reporting member firm.

3 Implementation

3.1 Interest Discovery

On September 23, 2016, during the CAA Acoustics Week in Canada Conference in Vancouver, a presentation on the subject of the proposed trade association will be provided by Steve Meszaros of RWDI Air. Following review of this paper, and the presentation, the association proponents would encourage interested parties to sign-up as potentially interested members.

3.2 Founding Committee

Subject to the feedback received at the conference, and the number of potentially interested member firms, a founding committee would need to be established. Ideally, the committee would compromise 3 to 5 individuals each representing a different member firm. In consultation with the Canadian Acoustical Association (CAA) Board of Directors, the founding committee would be tasked with the following:

- Naming the association
- Incorporating the association
- Formulating the Constitution and Code of Ethics
- Developing application criteria for corporate members
- Marketing the association (website, logo etc.)
- Devising the initial membership fee structure and developing other funding strategies to support the association (associate membership, sponsorships, etc.).

4 Discussion

It should be borne in mind that membership in the trade association would be, at least initially, corporate in nature, and not individual. Over time, the proposed trade association may be able to provide licensing registration to qualified individuals. Alternatively, the trade association may liaise with provincial engineering associations to officially recognize the discipline of acoustic engineering and to develop specific qualification requirements for individual practitioners.

The authors would like to reiterate that the proposed association should, by no means, be seen as a form of competition to the CAA. On the contrary, both organizations may stand to benefit from each other. At this point in time, the idea for the trade association is in its infancy. However, the proponents would anticipate that the CAA would hold an honorary membership within the trade association that would carry the same voting privilege as a corporate member.

The creation of the trade association for acoustic consulting companies in Canada would be an important step within our industry. The association would reflect recognition among members that, while we compete with each other within the market, we share common interests in public protection, the advancement of the industry and in the pride of our services to the community.

References

[1] AAAC. Guideline for Apartment and Townhouse Acoustic Rating. September 2010.

EFFECT OF VARIABILITY IN MICRO-GEOMETRY OF POLYURETHANE FOAMS ON THE DOUBLE WALL TRANSMISSION LOSS

Mohammad S. Gholami[*1], **Olivier Doutres**[2], **and Noureddine Atalla**[1]
[1]Groupe d'Acoustique de l'Université de Sherbrooke (GAUS), Sherbrooke (Québec) Canada.
[2]École de technologie supérieure (ÉTS), Montréal (Québec) Canada

1 Introduction

Propagation of waves in elastic porous media, e.g. polymeric foams, is described by Biot-Allard's theory [1]. Two classes of characteristic parameters are needed to describe the porous media in this theory. First, non-acoustic parameters: porosity ϕ, thermal characteristic length Λ', viscous characteristic length Λ, flow resistivity σ, and tortuosity α_∞ which are used in Johnson-Champoux-Allard (JCA) semi-phenomenological model [1]. Second, mechanical parameters, which in the case of isotropic material, are: bulk density ρ, Young's modulus E, loss factor η, and Poisson coefficient ν. The mechanical and non-acoustical properties are inherently dependent on the micro-structure of these materials. But, the internal structure of most porous material, e.g. polyurethane (PU) foam, is too complicated to be studied quantitatively. Therefore, the lattice of PU foams with low relative density (ρ_r) is commonly idealized by a tetrakaidecahedral, periodic unit cell (called PUC) and the macroscopic behavior recovered from a dedicated micro-macro approach. Furthermore, the cell windows can be randomly or partially closed, and cells are elongated in the rise direction. Measurements of such lattice results in variability in micro-structure properties of PUC. Hence, a clear understanding of the impact of variability associated with micro-structure properties measurement, and macroscopic Biot's parameters on vibro-acoustical performance of PU foams is of great importance in the design and optimization of such foams.

The impact of microstructure variability on JCA non-acoustical parameters and sound absorption was investigated by Doutres *et. al* [2]. This paper build on this work by investigating (i) the effect of the microstructure variability on the elastic properties of the foams and (ii) on the transmission loss of the foam when coupled with an elastic system (a double wall system is studied here). In this regard, a global sensitivity analysis (the FAST method) is applied to the micro-macro based models presented by Doutres *et. al* [3] and Gholami *et. al* [4]. The former model links the micro-structure properties of PUC (thickness and length of struts, and the closed windows content) of polyurethane foams to their non-acoustical parameters of JCA model. While, the latter model correlates the micro-structure properties and membrane thickness of PU foam to the Young's modulus in Biot-Allard's model.

*mohammad.sadegh.gholami@usherbrooke.ca

2 Methodology

2.1 The micro-macro model

The microstructure and non-acoustical properties of PU foams are linked using microstructural based models [3]. Gong's [5] analytical correlation between Young's modulus and micro-structure of fully reticulated PU foam was modified by Gholami *et. al* [4], numerically, to add the effect of closed pore content and membrane thickness on the Young's modulus:

$$\left(\frac{E}{E_s}\right)_{R_w} = C_E\,(1-Rw)^{1.4} + \left(\frac{E}{E_s}\right)_{R_w=1}. \quad (1)$$

where $C_E = 0.0017$ when memebrane thickness is $2.1\mu m$, and $(E/E_s)_{R_w=1}$ is the relative Young's modulus of fully reticulated PU foam [5].

2.2 Sensitivity Analysis Method

The contribution of input parameters on the output, in variance based techniques, is investigated by quantifying the impact of input variation on the output variance. A global sensitivity analysis method is used to determine the output sensitivity when the inputs vary over wide ranges. The FAST method is an efficient technique to explore the n-dimensional space of inputs [6]. FAST is used to estimate the first sensitivity index $SI(i)$ (which is the main effect of parameter i), and to calculate the total sensitivity index TSI which is defined as the sum of all SIs. The ratio of standard deviation to the mean of output, namely Normalized Standard Deviation (NSD) is an efficient indicator that shows the level of variability of the feature of interest [6].

2.3 Description of vibro-acoustic systems

As shown in Fig. 1, $15mm$ PU foam is bonded to the excited, $1mm$ steel plate. The foam is relaxed from a $2mm$ heavy layer (septum) by adding a $5mm$ air space. Decoupling the heavy mass from PU foam by air-gap allows us to account, at the same time, for absorption and isolation problem. Laterally infinite considered system is excited acoustically by an oblique plane wave. The transmission loss is calculated using transfer matrix method [1].

3 Results

3.1 Variability in PUC and Biot'a properties

Two studies are performed in this section. First, the impact of variability in PUC properties of two representative foams, $V1$ and $V2$ (see Table 1), are studied. $V1$ is considered to present a highly reticulated foams and $V2$ contains a low content of

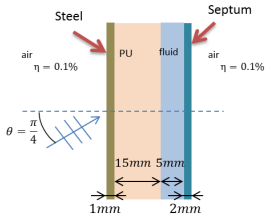


Figure 1: Vibro-acoustic configuration of double-wall system.

opened windows. The variability in the PUC is set to 10%. In the second study, the impact of variability in Biot-Allard's

Foam	$t \times 10^{-6}(m)$	$l \times 10^{-6}(m)$	$R_w(\%)$
$V1$	[51.3, 62.7]	[157.9, 192.5]	[70, 90]
$V2$	[51.3, 62.7]	[157.9, 192.5]	[10, 30]

Table 1: Inferior and superior limits for microstructure properties.

parameter (see Table 2), plate properties, and mass layer on the TL is studied. The impact of variation in PUC properties

Parameters	Inferior limit	Superior limit
$\phi(-)$	0.95	0.98
$\sigma(N.s.m^{-4})$	5,000	40,000
$\alpha_\infty(-)$	1.05	2.00
$\Lambda(mm)$	0.041	0.205
$\Lambda'(mm)$	0.190	0.319
$E(kPa)$	320	480
$\nu(-)$	0.3	0.4
$\rho(kgm^{-3})$	22	27
$\eta(-)$	0.1	0.15
$\rho_{Septum}(kgm^{-2})$	4	6
$\rho_{Al}(kgm^{-3})$	2605	2879
$E_{Al}(GPa)$	65.55	72.45

Table 2: Inferior and superior limits for Biot's properties.

of $V1$ and $V2$ are shown in Fig. 2. It is shown that strut length is main parameter for $V1$. While, the reticulation rate is the dominant parameter for $V2$ mainly at double wall frequency, frame born resonance and at high frequency in the cavity absorption region. The contribution of macroscopic parameters of vibro-acoustic system on TL are shown in Fig. 3. As expected the mass of plate and septum are the dominant parameter at low frequency. However, tortuosity and flow resistivity are dominant at high frequency in the cavity absorption controlled region where the performance of the foam is important. Mechanical properties of the foam are important at the frame resonance of of the foam.

4 Conclusions

The impacts of PU foam microstructure variability and macroscopic properties on the TL of a laterally infinite double wall system are studied. It is shown that the impact of strut length is dominant where foam is highly reticulated. While, the reticulation rate takes the lead when close pore content is

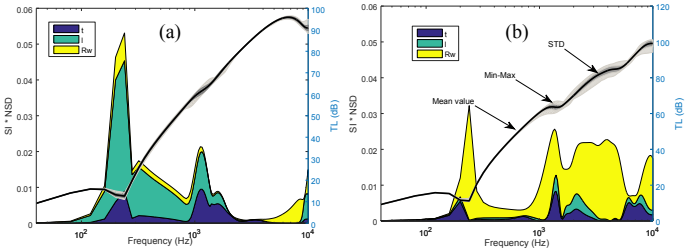


Figure 2: PUC variability impact on TL foam. a) V1, b) V2.

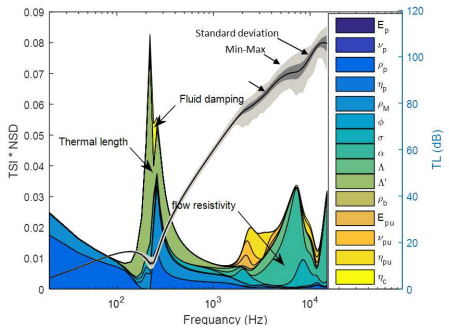


Figure 3: Macroscopic properties variability impact on TL.

high. Therefore, measuring precisely these parameters helps to reduce uncertainty in TL prediction of such a system. It should be mentioned that, for macroscopic properties, septum mass and plate are important at low frequencies, mechanical properties of the foam are dominant at foam resonance and non acoustical properties are important at high frequency.

Acknowledgments

The authors would like to thank the Natural Sciences and Engineering Research Council of Canada (NSERC) for providing financial support, and to gratefully acknowledge Dr. Morvan Ouisse for his kind generous support on the sensitivity analysis section.

References

[1] Jean Allard and Noureddine Atalla. *Propagation of Sound in Porous Media: Modelling Sound Absorbing Materials*. WILEY, 2e edition, 2009.

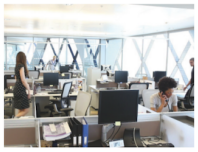
[2] Olivier Doutres, M Ouisse, N Atalla, and M Ichchou. Impact of the irregular microgeometry of polyurethane foam on the macroscopic acoustic behavior predicted by a unit-cell model. *The Journal of the Acoustical Society of America*, 2014.

[3] Olivier Doutres, Noureddine Atalla, and Kevin Dong. Effect of the microstructure closed pore content on the acoustic behavior of polyurethane foams. *Journal of Applied Physics*, 2011.

[4] Mohammad S Gholami, Olivier Doutres, and Noureddine Atalla. Effect of Variability in Microgeometry of Polyurethane Foams on their Macroscopic Acoustic Performance. In *Proceedings of NOISE-CON 2016*, Providence, 2016.

[5] L. Gong, S. Kyriakides, and W.-Y. Jang. Compressive response of open-cell foams. Part I: Morphology and elastic properties. *International Journal of Solids and Structures*, 2005.

[6] Morvan Ouisse, Mohamed Ichchou, Slaheddine Chedly, and Manuel Collet. *Journal of Sound and Vibration*, 2012.






CadnaR®
Prediction of
Noise Levels inside Rooms

CadnaR is the powerful software for the calculation and assessment of sound levels in rooms and at workplaces

Intuitive Handling

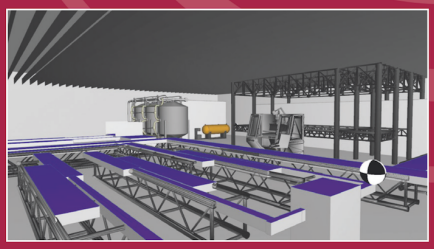
The clearly arranged software enables the user to easily build models and make precise predictions. At the same time you benefit from the sophisticated input possibilities as your analysis becomes more complex.

Efficient Workflow

Change your view from 2D to 3D within a second. Multiply the modeling speed by using various shortcuts and automation techniques. Many time-saving acceleration procedures enable a fast calculation process.

Modern Analysis

CadnaR uses scientific and highly efficient calculation methods. Techniques like scenario analysis, grid arithmetic or the display of results within a 3D-grid enhance your analysis and support you during the whole planning and assessment process.

Fields of Application

Office Environments

- Process your acoustic calculations and assessments according to DIN 18041, VDI 2569 and ISO 3382-3
- Receiver chains serve as digital “measurement path” and provide you with relevant insights into the acoustic quality of rooms during the planning phase
- Import of DWG-/DXF-/SKP-files (e.g. pCon.planner, AutoCAD, SketchUp)
- Visualization of noise propagation, noise levels and parameters for quality criteria like the Speech Transmission Index STI

Production Plants

- Calculation of the sound load at workplaces based on the emission parameters specified by the machine manufacturer according to the EC guideline 2006/42/EC while also taking the room geometry and the room design into account
- Tools for enveloping surfaces and free field simulations to verify the sound power of the sources inside of the enveloping surface
- Calculation of the sound power level based on technical parameters such as rotational speed or power

DataKustik

Distributed in the U.S. and Canada by: Scantek, Inc. Sound and Vibration Instrumentation and Engineering
6430 Dobbin Rd, Suite C | Columbia, MD 21045 | 410-290-7726 | www.scantekinc.com

SOUNDS OF EMOTION - SONS DES ÉMOTIONS

Perception Of Emotional Speech By Listeners With Hearing Aids
Huiwen Goy, M. Kathleen Pichora-Fuller, Gurjit Singh, Frank A. Russo 182

Skin Conductance Responses To Emotional Speech In Hearing-Impaired And Hearing-Aided Listeners
Gabriel A Nespoli, Gurjit Singh, Frank A Russo 184

There's Something About That Groove: Rhythm Improves Detection Of Audio But Not Vibrotactile Asynchronies
Andrew P. Lauzon, Frank A. Russo, Laurence R. Harris 186

Environmental Sound And Its Relation To Human Emotion
Barry Truax 188

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 190

PERCEPTION OF EMOTIONAL SPEECH BY LISTENERS WITH HEARING AIDS

Huiwen Goy [*1], M. Kathleen Pichora-Fuller [†1,2], Gurjit Singh, [‡2,3] and Frank A. Russo [♩4]
[1]Department of Psychology, University of Toronto, 3359 Mississauga Rd, Mississauga ON L5L 1C6
[2]Toronto Rehabilitation Institute, 550 University Ave, Toronto ON M5G 2A2
[3]Phonak Canada, 80 Courtneypark Dr #1, Mississauga, Ontario L5W 0B3
[4]Department of Psychology, Ryerson University, 350 Victoria St, Toronto ON M5B 2K3

1 Introduction

Recognizing the emotional state of a talker is an important part of social interaction. Previous research has shown that normal-hearing older adults do not recognize vocal emotions as well as younger adults [1], but little is known about how older adults with hearing loss perceive vocal emotion. In older adults who are not hearing aid users, hearing sensitivity predicts their valence ratings on a vocal emotion recognition task [2]. Older adults who are hearing aid users may therefore be expected to perform more poorly than their better-hearing peers. Hearing aids may restore the audibility of some vocal cues for emotion recognition, or they may distort the original intonation contour due to frequency and amplitude compression. The present study investigates how well older adults with hearing loss can recognize emotional speech and identify vocal emotions, and whether hearing aids are beneficial for both tasks.

2 Method

2.1 Participants

Participants learned English before the age of 5 years in an English-speaking country and were in average to excellent health with no speech or neurological disorders. Eleven older adults (mean age = 77.2 years) with hearing loss (Figure 1) were recruited from the community. They were hearing aid users with a median of 5 years of experience, who reported being satisfied with their hearing aids. They had at most two tested frequencies with an inter-aural difference >15 dB HL, except for one older adult who had three tested frequencies with a difference of >15 dB HL.

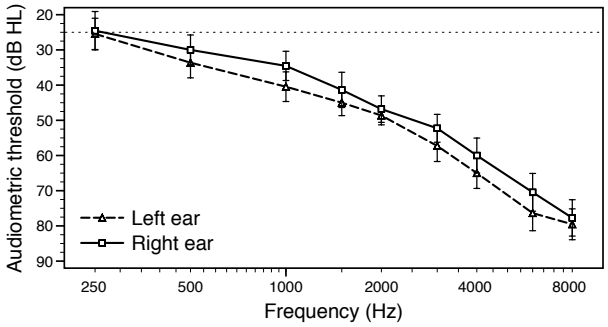


Figure 1: Average audiogram of older adults with hearing loss, with standard error bars.

Twenty-seven younger adults (mean age = 18.3 years) were students recruited from an introductory psychology course. They had normal audiometric thresholds of ≤25 dB HL from 250 Hz to 8 kHz in both ears, with inter-aural differences ≤15 dB HL at all tested frequencies, except for one younger adult who had a difference of 20 dB HL at 8 kHz.

2.2 Stimuli and procedure

The stimuli were sentences from the Toronto Emotional Speech Set spoken by a young adult female actor to portray seven emotion conditions : Angry, Disgust, Fear, Happy, Neutral, Pleasant Surprise and Sad [3]. These sentences consisted of keywords spoken after a standard carrier phrase (e.g., *Say the word bean*). Listeners were seated in an IAC sound-attenuating booth facing a loudspeaker placed 1.8 m in front at head height. Fourteen younger adults and 11 older adults with hearing loss heard these sentences in quiet, while 13 younger adults heard these sentences in babble noise from the Speech Perception in Noise Test (Revised; SPIN-R) [4] at -5 dB SNR. After each sentence, listeners reported the keyword and used a computer touchscreen to identify the emotion portrayed by the talker. The task order of word recognition and emotion identification was counterbalanced across listeners in each group. Each listener completed 14 practice trials and 168 experimental trials in which 24 words were presented in each of 7 emotion conditions. There were seven available sentence lists that combined each target word with each emotion across lists. Each younger listener was presented with only one list. Each older listener with hearing loss was presented with two different lists in separate sessions: the first without wearing hearing aids and the second while wearing hearing aids.

2.3 Data analysis

For each listener group, a repeated-measures analysis of variance (ANOVA) was conducted for each outcome measure of word recognition and emotion identification. For younger listeners, the single factor was emotion. For older listeners, the two factors were emotion and hearing aid condition (unaided or aided). Post-hoc *t*-tests with Holm-Bonferroni correction were conducted to further investigate significant main effects of emotion.

3 Results

3.1 Effects of emotion and hearing aid use on word recognition

Results are shown in Figure 2, including data from a previous study on normal-hearing older adults who were tested in SPIN-R babble noise at +2 dB SNR [5]. The word recognition accuracy of older adults with hearing loss improved when they used hearing aids compared to when

[*] huiwen.goy@utoronto.ca
[†] k.pichora.fuller@utoronto.ca
[‡] Gurjit.Singh@phonak.com
[♩] russo@psych.ryerson.ca

they were unaided, as confirmed by a significant main effect of hearing aid condition (Table 1). Compared to normal-hearing older adults tested in noise, older adults with hearing loss performed worse when unaided, but better than older adults in noise when aided (p's < 0.001).

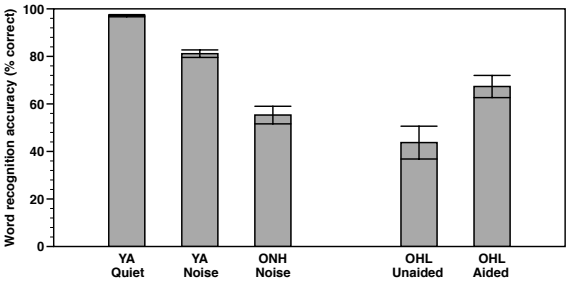


Figure 2: Word recognition accuracy of younger adults in quiet and in noise, normal-hearing older adults in noise, and older adults with hearing loss in quiet with and without hearing aids.

There was a significant main effect of emotion on word recognition accuracy for all listener groups except younger adults in quiet, whose performance was at ceiling (Table 1). For younger adults in noise, word recognition accuracy was better in Disgust/Angry/Neutral than in Happy/Sad conditions. For older adults with normal hearing, performance was better in Neutral/Fear than in Happy/Sad conditions, and better in Fear than all other conditions. For older adults with hearing loss, performance was better in Neutral/Angry than in Happy/Sad conditions, and better in Fear than the Happy condition (all p's < 0.05).

Table 1: F-values from ANOVAs for word recognition accuracy. Symbols legend: ^{ns} $p > 0.05$; $^{\times}p < 0.05$ but no significant pairwise differences; $^{*}p < 0.05$; $^{**}p < 0.001$.

	Y Quiet	Y Noise	ON Noise	OH Quiet
Emotion	$2.44^{\times}$	11.44*	14.4*	7.49*
HA use	-	-	-	18.8*
Emotion x HA	-	-	-	1.62^{ns}

3.2 Effects of emotion and hearing aid use on emotion identification

Results are shown in Figure 3, including data from a previous study on normal-hearing older adults who were tested in quiet [1]. Older adults with hearing loss clearly performed worse than normal-hearing older adults. There was a small but significant improvement in emotion identification when older adults with hearing loss used hearing aids compared to when they were unaided (Table 2).

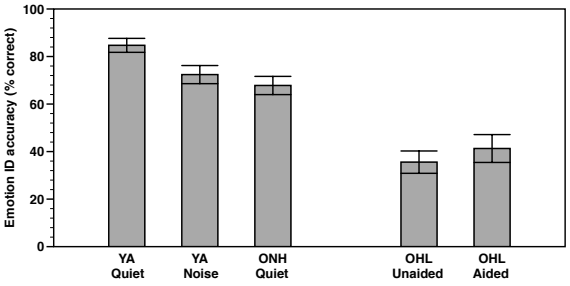


Figure 3: Emotion identification accuracy of younger adults in quiet and in noise, normal-hearing older adults in quiet, and older adults with hearing loss in quiet with and without hearing aids.

There was a significant main effect of emotion on emotion identification accuracy for normal-hearing younger and older adults, but not for older adults with hearing loss (Table 2). For younger and older adults in noise, emotion identification accuracy was better in Happy/Sad conditions than in the Disgust condition (p's < 0.05).

Table 2: F-values from ANOVAs for emotion identification accuracy. Symbols legend is identical to that of Table 1.

	Y Quiet	Y Noise	ON Quiet	OH Quiet
Emotion	$4.71^{\times}$	6.79**	7.17**	1.86^{ns}
HA use	-	-	-	6.33*
Emotion x HA	-	-	-	0.44^{ns}

4 Discussion

Without hearing aids, older adults with hearing loss performed worse than normal-hearing older adults on both word recognition and emotion identification. The use of hearing aids improved performance on both tasks, but much more so for word recognition (24 percentage points) than for emotion identification (6 percentage points). It may be concluded that hearing aid processing benefits word recognition by restoring the audibility of speech cues, but without much benefit to the perception of cues for vocal emotion. Another possibility is that listeners with hearing loss no longer attend to the same emotion cues as normal-hearing listeners or process them in the same way. Listeners with hearing loss were somewhat like listeners with normal hearing in that they found words to be more intelligible if spoken in the Neutral condition than in the Happy or Sad conditions. Unlike normal-hearing listeners, however, the type of emotion did not affect their emotion identification performance. Future directions may include measuring the output of hearing aids to determine how hearing aid processing affects cues for emotion in the speech signal, and determining which hearing aid settings work best for emotion recognition without compromising intelligibility.

Acknowledgments

This study was funded by a MITACS Accelerate Cluster Grant (PI: Russo) and an NSERC grant (PI: Pichora-Fuller).

References

[1] Dupuis K, Pichora-Fuller MK. Aging affects identification of vocal emotions in semantically neutral sentences. *J Speech Lang Hear Res*. 2015;58(3):1061-1076. doi:10.1044/2015_JSLHR-H-14-0256

[2] Schmidt J, Janse E, Scharenborg O. Perception of emotion in conversational speech by younger and older listeners. *Front Psychol*. 2016;7:781. doi:10.3389/fpsyg.2016.00781

[3] Dupuis K, Pichora-Fuller MK. Effects of emotional content and emotional voice on speech intelligibility in younger and older adults. *Can Acoust.* 2008:35(3):114-115.

[4] Bilger RC, Nuetzel JM, Rabinowitz WM, Rzeczkowski C. Standardization of a test of speech perception in noise. *J Speech Hear Res.* 1984;27(1):32-48. doi:10.1044/jshr.2701.32

[5] Dupuis K, Pichora-Fuller MK. Intelligibility of emotional speech in younger and older adults. *Ear Hear.* 2014;35(6):695-707. doi:10.1097/AUD.0000000000000082

SKIN CONDUCTANCE RESPONSES TO EMOTIONAL SPEECH IN HEARING-IMPAIRED AND HEARING-AIDED LISTENERS

Gabriel A. Nespoli [*1], Gurjit Singh [†2] and Frank A. Russo [‡2]
[1]Department of Psychology, Ryerson University, 350 Victoria St., Toronto, ON M5B 2K3
[2]Phonak AG, Toronto Rehabilitation Institute, 550 University Ave, Toronto, ON M5G 2A2

1 Introduction

Age-related hearing loss is present in many individuals and can have negative effects on an individual's quality of life [1]. Hearing aids are a useful device to combat this problem, and operate by amplifying sounds in the environment so that they are more audible at the level of the eardrum. The focus of optimization in hearing aids tends to be speech intelligibility. Perception of speech emotion tends to be a secondary consideration, and at present we have little research available on what parameters support this important aspect of hearing. Signal processing in hearing aids may lead to erroneous attributions of emotion; for example, because both joy and anger tend to be loud, they both result in dynamic compression of the signal, reducing discriminability of these emotions, and potentially leading to social misunderstanding. Current work from our group with hearing impaired older adults has found that while word identification in spoken sentences improves with the use of hearing aids, emotion identification does not.

The skin conductance response (SCR) is a measure of the amount of sweat on the skin and is considered an index of sympathetic nervous system activity, or arousal. In normal hearing adults, characteristic skin conductance responses are triggered in response to emotional stimuli such as pictures [2] and music [3]. In these studies, greater SCR magnitudes are seen in response to more arousing stimuli. How this response might be affected by hearing impairment—or by the use of hearing aids—has not been investigated.

The current study sought to investigate the effects of hearing impairment on the SCR to emotional speech. Additionally, we were interested in the extent to which hearing aids would help to recover the SCR seen in normal hearing participants. It was predicted that hearing impairment would result in a dampened SCR, and that hearing aids would not be able to recover the normal response. Based on previous work, we also predicted that hearing impairment and the use of hearing aids would decrease both accuracy and reaction time of categorizing emotional speech.

2 Method

Normal hearing (NH) and hearing impaired/aided (HI/HA) participants listened to semantically neutral sentences spoken to convey different emotions (happy, sad, angry, calm). These emotions vary with respect to arousal and valence and are representative of the four quadrants of the circumplex model of affect [4]. Stimuli were taken from the Ryerson Audio-Visual Database of Speech and Song (RAVDESS), and were validated in a separate study prior to the main study [5]. The RAVDESS database contains audio and audio-visual stimuli spoken in 8 different emotions and 24 different actors. HI/HA participants performed two blocks of the task (stimulus sets with different actors), one without their hearing aids (the same stimulus set as NH) and one with their hearing aids. The order of stimuli and blocks was randomized and were presented over loudspeakers in a sound-attenuated booth. Since loudness is a cue to emotion in speech, loudness was not normalized across stimuli, resulting in a range of presentation sound levels.

For each stimulus, participants were asked to respond on a computer keyboard (with their left hand) which emotion they thought was conveyed (four-alternative forced choice). For the duration of the experiment, skin conductance levels were measured using sensors attached to the index and ring fingers of the right hand (Biopac, Inc.).

All data analyses were conducted in MATLAB using the PHZLAB Toolbox (github.com/gabenespoli/phzlab). For all analyses, trials with a response time greater than 7 seconds were discarded. Skin conductance data were filtered with a 4th-order high-pass Butterworth filter and baseline-corrected by subtracting the mean of 1-second prior to stimulus onset from the entire epoch. Trials with values exceeding 0.05 µS (microsiemans) were discarded as artifact and the resulting trials were smoothed using a 400 ms sliding average. Since the speech stimuli had an average length of 3.9 s (range 3.3-4.5) and SCR typically has a 1-second delay, a target window was defined from 1 to 4.9 seconds post stimulus onset. The mean of the target window was taken as the index of physiological response to a given stimulus.

3 Results

For behavioural responses (accuracy and reaction time), a 1-way ANOVA was carried out with Group as a between-subjects factor (3 levels: NH, HI, and HA). There was a significant effect of Group on accuracy ($F(2,33)=3.729$, $p=0.035$), with NH performing best overall. There was also a significant effect of Group on reaction time (RT) ($F(2,33)=5.002$, $p=0.013$), with NH being faster overall.

[*] gabe@psych.ryerson.ca
[†] Gurjit.Singh@phonak.com
[‡] russo@ryerson.ca

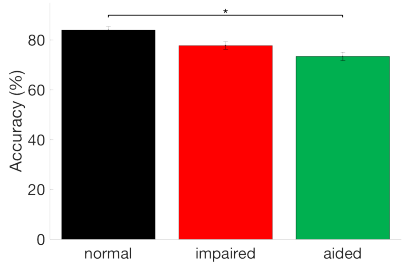


Figure 1: Mean accuracy scores across the three groups.

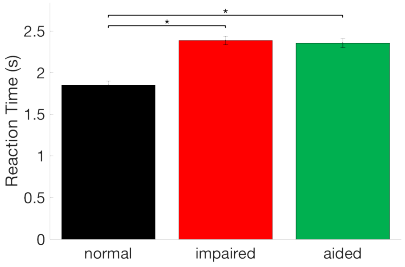


Figure 2: Mean reaction times across the three groups.

For physiological responses (mean SCR), a mixed ANOVA was carried out with Group as a between-subjects factor (3 levels: NH, HI, and HA), and with Arousal (2 levels: high and low) and Valence (2 levels: positive and negative) as within-subjects factors. This ANOVA revealed marginally significant main effects of Arousal ($F(1,33)=2.783$, $p=0.105$) and Group ($F(2,33)=2.238$, $p=0.123$), as well as a marginally significant Arousal-by-Group interaction ($F(2,33)=2.059$, $p=0.144$).

To summarize, high-arousal emotions evoke larger SCRs than low-arousal emotions. It also appears that HI have much smaller SCRs than the NH and HA, especially for the high-arousal emotions.

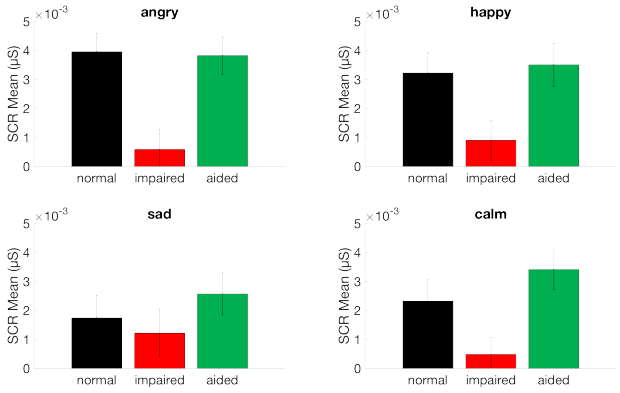


Figure 3: Mean skin conductance responses (SCR) from 1-5 s post stimulus onset. Normal hearing participants (NH) show a larger response for high-arousal emotions (angry, happy). Hearing impaired participants (HI) have a dampened response, while hearing aided participants (HA) seem to show increases for all emotions.

4 Discussion

As predicted, NH participants displayed superior accuracy and reaction time when discriminating emotional speech. Thus, hearing impairment impairs the ability to recognize emotion in speech, and hearing aids are unable to ameliorate this deficit. Also, there appears to be a trend where hearing impairments is associated with a deficit in the SCR to emotional speech. Hearing aids seem to be able to recover the SCR overall, but SCR levels are not discriminable across emotions.

It seems likely that loudness was driving many of the results seen in this study. For the normal hearing participants, they might have used loudness as a cue to the level of arousal of the emotion being expressed; for example, a louder stimulus would be more likely to be happy or angry. For the hearing impaired participants, all stimuli would have been perceived as very quiet. As such, they are slower to respond when discriminating emotion, and they do not show a characteristic arousal response in terms of SCR. For the hearing aided participants, their hearing aids would have applied gain and compression in order to maximize intelligibility. This would have minimized sound level differences between the quieter stimuli (i.e., sad and calm) and the the louder stimuli (i.e., happy and angry), which would have a) eliminated the use of loudness as a cue to emotion, causing them to respond slower and less accurately, and b) made all stimuli sound loud and potentially arousing, thus increasing SCRs for all emotions.

5 Conclusion

The current study found behavioural and physiological evidence for a deficit in the perception of emotion in speech for hearing impaired individuals. In addition, hearing aids were unable to ameliorate these deficits.

References

[1] Dalton, D. S., Cruickshanks, K. J., Klein, B. E. K., Klein, R., Wiley, T. L., & Nondahl, D. M. (2003). The impact of hearing loss on quality of life in older adults. The Gerontologist, *43*(5), 661–668.

[2] Lang, P. J., Greenwald, M. K., Bradley, M. M., & Hamm, a O. (1993). Looking at pictures: affective, facial, visceral, and behavioral reactions. *Psychophys.*, *30*(3), 261–273.

[3] Khalfa, S., Isabelle, P., Jean-Pierre, B., & Manon, R. (2002). Event-related skin conductance responses to musical emotions in humans. *Neurosci. Letters*, *328*(2), 145–149.

[4] Russell, J. A. (1980). A circumplex model of affect. *J. of Pers. and Soc. Psych*, *39*(6), 1161-1178.

[5] Livingstone, S. R., Peck, K., & Russo, F. A. (2013, June). Acoustic differences in the speaking and singing voice. In *Proc. of Meetings on Acoustics, Ac. Soc. of Am.*, *19*(1), 035080.

THERE'S SOMETHING ABOUT THAT GROOVE: RHYTHM IMPROVES DETECTION OF AUDIO BUT NOT VIBROTACTILE ASYNCHRONIES

Andrew P. Lauzon[*1,2], Frank A. Russo[†2], and Laurence R. Harris[‡1]
[1] Department of Psychology, York University, 4700 Keele St., Toronto, ON, Canada, M3J 1P3.
[2] Department of Psychology, Ryerson University, 350 Victoria St., Toronto, ON, Canada, M5B 2K3.

1 Introduction

It is well established that context can powerfully influence sensory perception and that rhythm can induce entrainment in the form of regularly occurring evoked potentials in neural activity. These evoked potentials are defined by the interval between salient beats, and are thus oscillatory in nature. These neural oscillations are presumed to modulate (a) sensory input responses (Schroeder & Lakatos, 2009) as well as (b) attention, an idea described by Jones, Moynihan, MacKenzie and Puente (2002) to explain their finding that judgments regarding small pitch deviations were most accurate when target stimuli occurred closest to an expected temporal interval (i.e. on the beat).

The present study examined the influence of an isochronous rhythm on judgements of simultaneity in auditory and vibrotactile modalities by measuring thresholds of detection (TOD) for tones presented with slight asynchronies, both with and without the context of an isochronous pattern. A significant improvement in detection when accompanied by isochrony would provide support for the previously mentioned theoretical framework, and a difference in thresholds obtained between the auditory and vibrotactile modalities would be in line with the notion of an auditory advantage for temporal processing.

2 Method

2.1 Participants

A convenience sample of 10 volunteers was recruited by word-of-mouth and consisted of four females and six males (age range 22-48; mean age = 31.7); all had normal hearing, and musical experience ranged from professional musicians to those with no musical training. Hearing and musical experience were determined by a questionnaire which was completed prior to the experiment. Participants were not paid. Experiments were approved by the York University and Ryerson Research Ethics Board and adhered to the Treaty of Helsinki.

2.2 Apparatus

Experimental trials were run using Cycling '74 Max MSP software on a 2010 Macbook Pro with an RME Fireface 400 firewire audio interface. Audio for the auditory conditions was delivered to participants through Sennheiser HD518 over-ear headphones. Vibrotactile stimuli also originated as audio, and were sent to vibrating voice coils embedded in the seat and back of a slightly reclining padded chair (Emoti-chair; Karam, Russo, & Fels, 2009) in which participants would sit for the duration of the experiment. A pre-study was conducted in order to equalize perceived magnitude between the auditory and vibrotactile stimuli.

Tactaid VBW32 transducers were attached to the participants' mastoid bones to deliver pink noise for the purpose of masking auditory detection of bone conducted vibrations potentially received via the voice coils. Pink noise was also delivered through the headphones to mask any additional airborne sound that was emitted by the voice coils in the chair.

2.3 Stimuli

Target stimuli consisted of a pair of pure tones (200 Hz and 300 Hz) presented either in perfect synchrony or with one of 10 possible stimulus onset asynchronies (SOA). The same tones were used for both modalities, however SOAs for the auditory conditions ranged from 5-23 milliseconds (ms) whereas vibrotactile SOAs ranged from 10-190 ms. Target stimuli were presented in one of the three different rhythmic contexts for both modalities, for a total of six experimental conditions:

Regular Rhythm (RR) : Eight beats occurred with an inter stimulus interval (ISI) of 500 ms, or 120 beats per minute (BPM). Beats 1-6 and 8 were the context stimuli, which consisted of the 200 Hz and 300 Hz pure tones played in perfect synchrony and beat 7 was the target stimulus which was the same two tones, either identical to the context stimuli or with a slight SOA.

Irregular Rhythm (IR) : Target stimuli were present within the context of an irregularly occurring, unpredictable beat sequence. This was identical to the regular rhythm condition, with the exception that the ISI duration changed on every beat for the first six beats. Target stimuli occurred at the same relative time point within each trial, as compared to all other conditions.

No Rhythm (NR) : Target stimuli were presented unaccompanied (with no context). This condition was identical to the regular rhythm, except the context stimuli (beats 1-6 and 8) were silent.

[*] alauzon@my.yorku.ca
[†] russo@ryerson.ca
[‡] harris@yorku.ca



Figure 1: Graphical representation of the three rhythmic context conditions with asynchronous target stimuli.

A two-alternative forced choice procedure was used. For each trial, participants would hear or feel the target stimulus twice: once with the asynchrony at beat 7 and once with no asynchrony, to decide which of the two presentations contained the SOA, the order of which was randomized. Answers were indicated by entering either "1" or "2" on a computer keyboard. A block design was employed with each block consisting of one of the three rhythmic contexts, the order of which was counterbalanced.

2.4 Data Analysis

Data were plotted as a function of SOA value, and for each participant a percentage correct score was calculated for each SOA. A logistic curve was fit to each participant's data, and each participant's TOD was determined as the SOA value at which they achieved 75% correct. Standard deviation for each participant was extracted from the slope of the curve, and overall variability was compared between modalities as a measure of response consistency.

Separate repeated-measures ANOVAs were performed for each modality to compare TODs, and a two-way repeated measures ANOVA was performed to compare overall variability between the two modalities. Pairwise comparisons used the Bonferroni correction.

3 Results

3.1 Detection Thresholds

A significant effect of rhythmic context was found in the auditory conditions, $F(2, 18) = 3.56$, $p = .05$, partial $\eta^2 = 0.283$ (large). Pairwise comparisons revealed a significant difference between the no rhythm ($M = 8.97$, $SD = 3.06$) and regular rhythm ($M = 6.12$, $SD = 1.60$) conditions which corresponds to a ~32% reduction in thresholds ($MD = 2.85$, $SE = .929$, $p = .04$), but no difference when comparing either of the conditions to irregular rhythm ($M = 7.66$, $SD = 2.76$). A repeated-measures ANOVA performed on the vibrotactile conditions failed to reach significance $F(2, 18) = 2.86$, $p = .084$, partial $\eta^2 = 0.241$.

3.2 Variability

A two-way repeated measures ANOVA compared variability between all conditions (both modalities and all rhythmic contexts) and showed a highly significant difference $F(1, 9) = 23.34$, $p = .001$, partial $\eta^2 = .722$. A pairwise comparison of the overall mean variability scores of each modality ($MD = 6.23$, $SE = 1.29$, $p = .001$) was also significant. This suggests the vibrotactile conditions were more difficult as a whole.

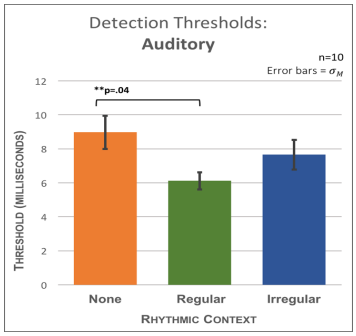


Figure 2: Detection thresholds for the auditory conditions. Note the significant reduction (p = .04) in the regular rhythm condition compared to no rhythm. No significant differences were found between any of the vibrotactile conditions.

4 Discussion and Conclusions

These results support the hypothesis that context in the form of an isochronous rhythmic pattern does indeed affect simultaneity judgements in the auditory domain. It also contributes further support to the hypothesis of privileged temporal processing for auditory stimuli and prompts questions about vibrotactile perception, cross-modal plasticity, and higher order processing of temporal information.

The high variability and lack of significant results in the vibrotactile conditions suggests a failure to entrain, and may simply be due to a lack of acuity in lower-level somatosensory processes, which are unsuccessful in transmitting neural impulses with enough temporal precision to synchronize attentional dynamics, expectations, or neural oscillations. On the other hand, an experience-dependent explanation could support a more top-down, cortically mediated process in which timing information received via somatosensory receptors is robust, but processed inefficiently and subject to improvement with practice.

References

Karam, M., Russo, F. A., & Fels, D. I. (2009). Designing the Model Human Cochlea: An Ambient Crossmodal Audio-Tactile Display. *IEEE Transactions on Haptics*, *2*(3), 160–169.

Jones, M. R., Moynihan, H., MacKenzie, N., & Puente, J. (2002). Temporal Aspects of Stimulus-Driven Attending in Dynamic Arrays. *Psychological Science*, *13*(4), 313–319.

Schroeder, C. E., & Lakatos, P. (2009). Low-frequency neuronal oscillations as instruments of sensory selection. *Trends in Neurosciences*, *32*(1), 9–18.

ENVIRONMENTAL SOUND AND ITS RELATION TO HUMAN EMOTION

Barry Truax, Prof. Emeritus
School of Communication, Simon Fraser University, Burnaby, BC

1 Introduction

Speech, music and environmental sound refer to three specialized areas of acoustic communication and its study, and these areas can be regarded as forming a continuum of human aural experience [7]. Although they have traditionally been studied separately, there are many factors today pointing to their overlap and interaction. Most obviously, contemporary audio technology has blurred their distinctions by making reproduced speech and music into common environmental sounds, often structured by media and individuals as accompaniment environments [1]. The widespread contemporary presence of music-as-environment in public and personal usage can be understood as an extension of the programmed use of music as a background and more recently, foreground presence in both public and commercial spaces. Of the many purposes such music serves, one function is to transfer the emotional aspects of the music onto the environment where it is heard, just as music in audio advertising becomes 'attached' to a specific product. Although the relatively intense affective responses that we regard as the expressions of emotions through speech and music have been studied extensively, but often separately, any equivalent role of environmental sounds has been largely ignored. We can therefore ask how environmental sounds, with or without a speech and music component, and with or without human causality, can affect and effect human emotions.

Recent advances in brain functioning have begun to suggest that there are underlying mechanisms, related to specific parts of the brain, which can be linked to known psychological responses to both music and speech. One important clue to their relationship is that the emotional (and other) aspects of speech are conveyed by paralanguage (i.e. the nonverbal aspects of vocalization), the parameters for which are closely related to the musical parameters of melody, such as patterns of pitch inflection, rhythm (including tempo and patterns of stress), loudness and dynamic contours, timbre, and silence. Although it is beyond the scope of this paper, one could argue that the structure of musical melody developed historically as an extension of spoken dialogue, both in singing and instrumental performance. It is significant that both paralanguage and musical structure act as continuous, analog phenomena, basically 'how' the speech or music is communicated, rather than the more discrete content of words and notes [10]. Can similar structures be found with environmental sounds?

2 Environmental Sounds and Emotion

This paper wishes to extend this current line of research to the neglected area of environmental sound as it is perceived by individuals in context, namely as the soundscape. Soundscape competence, it is argued, co-evolved with the specialized areas of speech and music, and today with the widespread phenomenon of music-as-environment, as well as other media practices, it is useful to re-connect old arguments about music and emotion with contemporary soundscape experience. To do this, we will draw on the extensive literature summary by Patrick Juslin and Daniel Västfjäll concerning the underlying psychological mechanisms that are involved in the musical induction of emotions [5]. They identify the following six types of mechanisms.

2.1 Brain Stem Reflex

This reflex refers to the autonomic responses to certain acoustical characteristics that can be found in both music and the soundscape, namely loud, sudden, noisy sounds, those with very low or high frequency, or those that are dissonant (i.e. with spectral components less than a critical bandwidth apart). Such sounds are associated with arousal, potentially signifying danger. In the case of environmental sound, psychological reactions to noise are often based on these qualities, and in the converse situation, loud percussive sounds are often an organized part of public celebrations, sports events and entertainment intended to generate excitement and pleasure.

2.2 Evaluative Conditioning

When music is repeatedly paired with another event that is valenced positively or negatively, it can produce a conditioned emotional response when experienced later on its own. There is no reason why an environmental sound might not produce a similar conditioning. For instance, an 'earwitness account' that was recorded by the World Soundscape Project in its Vancouver study described the experience of hearing local foghorns late at night or early in the morning. She said that 'despite the fact that they were mournful, we seem to remember them as somehow comforting' [7, p. 20]. This is clearly a situation where the thick low-pitched sound of the foghorn, with its drooping pitch bend at the end, reflecting the paralanguage of mourning, provoked an emotional reaction of comfort. Moreover, this habitual response was still vivid 50 years later during the interview. In other interviews, sounds from nature, particularly when heard frequently or on special

occasions, became associated with particular life events. Given the power of such personal associations, it is not surprising that advertisers attempt to use such sound effects to associate particular qualities to their products.

2.3 Episodic Memory

Similar to evaluative conditioning, episodic memory refers to music (and presumably environmental sounds) evoking the memory of a particular event in someone's life. Music is well known to contribute to self-identity, particularly in one's youth, and to nostalgia later in life [2]. Although most people do not recall environmental sounds as readily as music, the practice of soundwalking has been extended to 'memory walks' where the subject is taken back to a significant locale in their life. Even if considerable change has occurred, being back in the original context seems to provoke sound memories quite effectively [4].

2.4 Emotional contagion

Emotional contagion refers to the induction of an emotion in a listener that is similar to that perceived in music (or expressed by another person), perhaps because of the similarity to vocal expression, i.e. its paralanguage, in a kind of mirror function. Bull [1] and others point to the deliberate use of media to control or stimulate one's emotions and 'control inner chaos'. People often report anecdotally about the 'soothing' effect of environmental sounds such as waves on a beach, or birds singing, as well as the excitement of, for instance, a windstorm, bells or fireworks. It is also common to attribute human emotions to natural sounds (e.g. an angry storm, a sighing wind, joyful birds, etc.).

2.5 Visual imagery

Justin and Västfjäll [5] also discuss emotion induced in a listener who conjures up visual imagery while listening to music, even if this phenomenon varies widely with different individuals. Given that environmental sounds are linked to actual visual contexts, there is a close connection between them when the listener imagines the source of the sound, with perhaps an emotional reaction. Such pairings are the basic language of film sound where the mood of an image or action is supported by the accompanying sound effects, as well as music.

[10]

2.6 Musical expectancy

One classic theory about emotion and music [3, 6] is that certain syntactical features of music induce expectations that may be disrupted but then resolved in a satisfactory manner. Such reactions clearly depend on learning the language of a particular musical style, i.e. developing musical competence. Similarly, the pattern of a sequence of environmental sounds (e.g. footsteps on a stair leading to a door) may generate a similar sense of expectation that might be fulfilled (e.g. a knock on the door) or disrupted (e.g. a package being delivered) with similar emotional effect.

3 Conclusion

Justin and Västfjäll regard brain function 'mechanisms as information-processing devices at various levels of the brain that use various means to track significant aspects of the environment, and that may produce conflicting outputs' (568-9). They provide an extensive list of mechanisms with their associated survival value, information focus, ontogenetic development, key brain regions involved and other aspects. Although designed specifically for sorting out the mechanisms involved with emotion and music, their approach seems equally valid for researching the affective and emotional response to soundscapes, particularly at a time when qualitative approaches to soundscape design are being developed, and composers are treating environment as music in soundscape composition [8-9].

References

[1] M. Bull, *Sound Moves: iPod Culture and Urban Experience*, London: Routledge, 2008.

[2] R. Herbert, *Everyday Music Listening: Absorption, Dissociation and Trancing,* Aldershot: Ashgate, 2011.

[3] D. Huron, *Sweet Anticipation*, MIT Press, 2006.

[4] H. Järviluoma, "Lesconil, My Home – Memories of Listening," in H. Järviluoma, B. Truax, H. Uimonen, M. Kytö and N. Vikman (eds.) *Acoustic Environments in Change*, Tampere: University of Joensuu, 2009.

[5] P. Juslin and D. Västfjäll, "Emotional Responses to Music," *Behavioral and Brain Sciences* 31: 559-621, 2008.

[6] L. B. Meyer, *Emotion and Meaning in Music*, University of Chicago Press, 1956.

[7] B. Truax, *Acoustic Communication*, 2nd ed. Norwood, NJ: Ablex Publishing, 2001.

[8] B. Truax, "Techniques and Genres of Soundscape Composition as Developed at Simon Fraser University," *Organised Sound* 7(1): 5-14, 2002.

[9] B. Truax, "Soundscape Composition as Global Music: Electroacoustic Music as Soundscape," *Organised Sound* 13(2): 103-109, 2008.

[10] B. Truax, "Sound, Listening and Place: The Aesthetic Dilemma, *Organised Sound* 17(3): 193-201, 2012.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Perception Of Emotion In Music By Hearing-Impaired And Hearing-Aided Listeners

Frank A. Russo, Domenica Fanelli

Recent research suggests that hearing impairment adversely affects processing of emotion in speech and that hearing aids can restore some but not all of the deficit (e.g., Schmidt et al., 2015). The current study sought to investigate whether similar patterns of deficit exist for perception of emotion in music. Hearing impaired, hearing-aided and normal listeners were asked to make forced-choice emotion responses for 50 15-sec musical excerpts. The excerpts were drawn from a prior study (Eerola, Vuoskoski, 2010), in which excerpts were validated to be representative of 5 target emotions (anger, fear, happy, sad, tender). Results showed that normal hearing listeners outperformed hearing impaired as well as hearing aided listeners and that overall performance in hearing impaired and hearing aided listeners was not differentiable. Follow-up regression analyses will assess performance with respect to audiometric data and differences from fitting targets (NAL-NL2). Confusions matrices will also be considered in order to provide insight into the source of the music emotion deficit.

Emotional Communication In Music: Implications For Understanding Links Between Between Speech And Music

Michael Schutz

Parallels in the structure of speech and music have long fascinated great thinkers from Plato to Darwin. The communication of emotion is no exception, as it touches on one of the primary motivations for musical listening – feeling "moved by the music." Although cues such as pitch height and timing are widely recognized as playing an important role in both domains, their role in music has been less researched than in language. Here I will discuss several of my team's explorations aimed at providing a musical complement to existing work in speech and linguistics. The plethora of speech corpora provide useful data for examining the "natural" use of acoustic communicate emotion. Unfortunately, the clear contrast between the number of 'effective speakers' and the number of 'effective composers' complicates efforts to fully explore parallels in structural cues between these domains. Numerous studies based on manipulations of simplified musical stimuli such as newly composed single line melodies suggest important parallels between the communication of emotion in speech and music. Melodies transposed higher in pitch sound "happier" and melodies played at slower tempi sound "sadder" – paralleling the use of these cues in speech. However, it is not clear whether these simplified approaches capture the nuanced ways in which great composers employ such cues in their writing. Here I will discuss my team's research on this important issue, which mixes the techniques of acoustical analysis, psychophysical testing, data visualization, and empirical musicology to provide a diverse exploration of musical emotion complementing and extending speech studies. In doing so my team aims to deepen our understanding of mechanisms shared between these ubiquitous human activities, providing insights useful to linguistics, musicians, psychologists, and cognitive scientists alike.

SPEECH ACQUISITION - ACQUISITION DE LA PAROLE

Infants' Use Of Temporal And Phonetic Information In The Encoding Of Audiovisual Speech
D. Kyle Danielson, Cassie Tam, Padmapriya Kandhadai, Janet F. Werker 192

Does Swallowing Bootstrap Speech Learning?
Connor Mayer, Francois Roewer-Despres, Ian Stavness, Bryan Gick 194

Perceptual Learning Of Speech Sounds: A Bias For 'sipper' Over 'zipper'?
Molly Babel, Zoe Lawler, Carolyn Norton 196

Does Training To Perceive L2 English Vowels In One Phonetic Context Transfer To Other Phonetic Contexts?
Ron I Thomson 198

Breathing Position And The Perception Of Consonants
Henny Yeung, Mark Scott, Lionel Granjon 200

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 202

INFANTS' USE OF TEMPORAL AND PHONETIC INFORMATION IN THE ENCODING OF AUDIOVISUAL SPEECH

D. Kyle Danielson*, Cassie Tam †, Padmapriya Kandhadai§, and Janet F. Werker ‡
Department of Psychology, The University of British Columbia, 2136 West Mall, Vancouver BC V6T 1Z4, Canada

1 Introduction

Infants utilize both auditory and visual information to perceive the speech signal. From as early as two months after birth, infants detect correspondence in the *content* of seen and heard consonants in their native language(s) [1-2] and (until they are about nine months of age) in non-native languages as well [3]. Infants' auditory perception can also be modified by the imposition of matching or mismatching visual information from a dynamic talking face [4-5]. Infants are also sensitive to the *temporal* correspondence of heard and seen speech, detecting asynchrony greater than approximately 500 ms therein [6].

What remains unclear is how infants perceive audiovisual speech when the auditory and visual signals are incongruent. When the auditory and visual signals provide conflicting information to the infant, on which of the two signals does she rely more heavily? Second, given that infants are sensitive to the temporal correspondence of speech as well, does the temporal order of the auditory and visual components affect which of the two signals the infant uses? Relying on unfamiliar non-native speech to control for the effect of prior experience on infants' behaviour, we test these two questions in the current study.

2 Method

2.1 Sample

Sixty full-term six-month-old monolingual English-learning infants (mean age = 179 days; 30 females) were tested.

2.2 Materials

Audiovisual stimuli for this experiment were recorded from a native female speaker of Hindi (Figure 1). The speaker produced consonant-vowel (CV) syllables (duration ≈ 1 s) consisting of the Hindi voiced dental stop [d̪] and the Hindi long vowel [ɑː] (dental syllables) and the Hindi voiced retroflex stop [ɖ] and the long vowel [ɑː] (retroflex syllables). Incongruent syllables were constructed by splicing the auditory track from one syllable type with the visual track from a duration-matched syllable of the opposite type. This procedure was repeated, resulting in three unique, temporally synchronous tokens of each mismatching type. Temporally asynchronous tokens were constructed by offsetting the auditory and visual tracks of the mismatching syllables by 333 ms, resulting in two types of asynchronous syllable: visual-first or auditory-first.

*kdanielson@psych.ubc.ca
† cassietam@alumni.ubc.ca
§ priyak@psych.ubc.ca
‡ jwerker@psych.ubc.ca



Figure 1 : Model producing dental and retroflex syllables

2.3 Procedure

Infants were seated on a caregiver's lap in a silent room facing a television screen equipped with a small camera. First, during the pre-test, infants listened to auditory-only exemplars of the Hindi dental and retroflex tokens while watching a black-and-white checkerboard. Then, infants were familiarized to 15-s sequences of Hindi audiovisual syllables in one of three conditions: synchronous, visual-first, and auditory-first. To ensure that infants only perceived audiovisual (not auditory-only) speech, stimuli were only presented when infants were looking at the screen. Each infant accumulated 120 s of looking time to the screen prior to advancing to the test phase of the experiment. Infants were tested using four auditory-only, checkerboard trials similar to those presented during the pre-test. Looking time was measured offline to determine whether infants exhibited a matching preference to the stimulus that they saw during familiarization (visual match) or to the stimulus that they heard during familiarization (auditory match).

3 Results

3.1 Main analysis

Looking time to auditory-match versus visual-match test sequences are visualized by condition as difference scores (Figure 2).

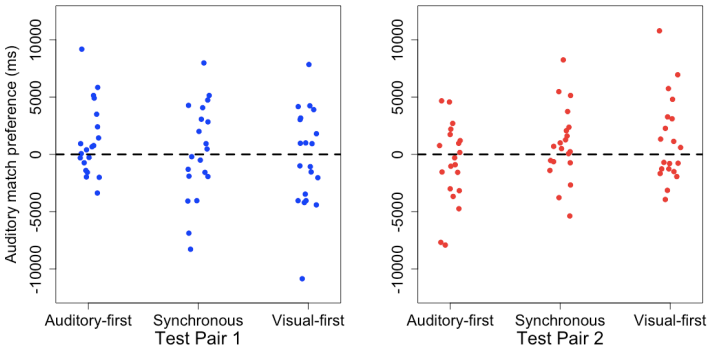


Figure 2: Difference scores (auditory match minus visual match in ms) for both pairs of test trials across three conditions (auditory-first, synchronous, visual-first). Positive scores indicate a preference for the auditory match sequences.

A 2 x 3 mixed-design ANOVA with match type as a within-subjects factor (auditory match, visual match) and familiarization condition (auditory-first, visual-first, synchronous) revealed no effects of match type ($F(1,57) = 1.40$, $p = .241$, $\eta^2_P = .02$) or of condition ($F(2,57) = 0.36$, $p = .703$, $\eta^2_P = .01$) on looking time. Importantly, the ANOVA also revealed no interaction between condition and match type ($F(2,57) = 0.11$, $p = .893$, $\eta^2_P < .01$), indicating that—regardless of familiarization condition—infants looked equally to the auditory match and visual match sequences at test.

3.2 *Post-hoc* analysis

Infants in this experiment were tested using unfamiliar Hindi dental and retroflex consonants. It is possible that infants treated these two types of consonants differently from one another, thus overshadowing the hypothesized effect. To explore this possibility, data were collapsed across all three temporally manipulated conditions, and then split based on whether the auditory information presented was dental or retroflex. These reformatted data are visualized as difference scores in Figure 3.

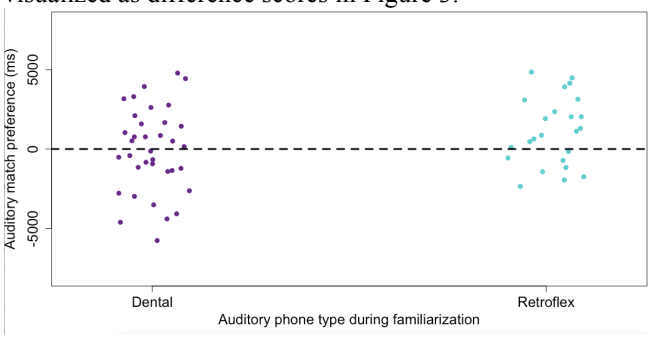


Figure 3: Difference scores (auditory match minus visual match in ms) for both pairs of test trials, split by auditory phone type during the familiarization phase.

A 2x2 mixed-design ANOVA was fitted to these looking time data, with match type (auditory match, visual match) as a within-subjects factor and auditory familiarization phone (dental, retroflex) as a between-subjects factor. A marginal interaction between match type and auditory familiarization phone emerged ($F(1,58) = 3.35$, $p = .072$, $\eta^2_P = .05$). Infants familiarized to stimuli in which the auditory component was a retroflex consonant exhibited a preference for retroflex test sequences.

4 Discussion

We familiarized infants to one of three types of audiovisual speech (visual-first, auditory-first, and synchronous), all consisting of tokens in which the visual and auditory information were incongruent. We then tested infants to determine whether they exhibited a matching preference for the syllables that they heard or that they saw. The main analysis did not reveal any interaction between familiarization condition and preference at test, indicating that infants' perception of the speech sounds was not affected by this manipulation. One possibility is that, when observing this type of audiovisually incongruent speech, infants' resulting percept is intermediate to the two syllables (as in the McGurk effect), matching neither of the two test items and resulting in no difference in preference. Another possibility is that infants perceived the incongruent visual and auditory syllables discretely, rendering both test items equally familiar. Alternatively, such a null result could be due to short exposure (120 s) or to the short temporal offset between the auditory and visual signals (333 ms), which was intentionally set below infants' hypothesized threshold for audiovisual integration [6].

However, a follow-up analysis revealed that infants familiarized to syllables in which the auditory component was a retroflex syllable exhibited a moderate preference for that syllable at test, regardless of familiarization condition. This result indicates that infants' sensitivity to the retroflex syllable may overshadow their perception of the visual dental consonant, regardless of whether the latter is presented before, simultaneously, or after the auditory retroflex consonant. Further research is required to determine why the retroflex auditory consonant, specifically, may attract more attention from English-learning infants than does the dental consonant.

5 Conclusion

The results of this study indicate that further research is necessary to determine whether infants preferentially utilize visual or auditory information to process speech sounds when the two signals provide conflicting information and when one precedes the other temporally. However, some preliminary evidence indicates that infants' speech perception may be primarily driven by auditory information from one type of unfamiliar speech sound over another.

Acknowledgments

We gratefully acknowledge the support of the National Institutes of Health (grant R21HD079260 to JFW), the Acoustical Society of America's Raymond H. Stetson Scholarship to DKD, and the Natural Sciences and Engineering Research Council of Canada's Undergraduate Student Research Award to CT.

References

[1] Kuhl, P. K., & Meltzoff, A. (1982). The bimodal perception of speech in infancy. *Science, 218*(4577), 1138-1141.

[2] Patterson, M. L., & Werker, J. F. (2003). Two-month-old infants match phonetic information in lips and voice. *Dev Sci, 6*(2), 191-196.

[3] Pons, F., Lewkowicz, D. J., Soto-Faraco, S., & Sebastian-Galles, N. (2009). Narrowing of intersensory speech perception in infancy. *PNAS, 106*(26), 10598-10602.

[4] Rosenblum, L. D., Schmuckler, M. A., & Johnson, J. A. (1997). The McGurk effect in infants.
Percept Psychophys, 59(3), 347-357.

[5] Teinonen, T., Aslin, R. N., Alku, P., & Csibra, G. (2008). Visual speech contributes to phonetic learning in 6-month-old infants. *Cognition, 108*(3), 850-855.

[6] Lewkowicz, D.J. (2010). Infant perception of audio-visual speech synchrony. *Dev Psychol*, *46*(1), 66-67.

DOES SWALLOWING BOOTSTRAP SPEECH LEARNING?

Connor Mayer[1,*], François Roewer-Despres[2,†], Ian Stavness[2,‡], and Bryan Gick[1,3,°]
[1]Department of Linguistics, University of British Columbia
[2]Department of Computer Science, University of Saskatchewan
[3]Haskins Laboratories, New Haven, CT

1 Introduction

Although speech movements must be learned, many of our most complex oral behaviours, such as swallowing, suckling, vocalizing, and breathing, can be produced at birth. This indicates a degree to which the biomechanical and neural structures needed for complex action appear to be built in. It has been hypothesized that phylogenetically-encoded structures such as these are used to bootstrap speech learning [1][2]: complex, innate movements may be broken down into constituent submodules that are then recruited for use in speech. This is consistent with developmental studies that have found that features of speech movements in infants are qualitatively similar to suckling movements, but these patterns are subsequently refined [3].

The learning of speech movements can be modeled as a search of a high-dimensional muscle activation space for sets of activations that satisfy task-specific criteria relevant to the speech learner. Even when considering a single speech movement in isolation, the dimensionality and size of the search space are large enough to provide significantly problems for an unstructured search: the number of sets of activations that result in a solution for a given task has been shown to be very small relative to the number of possible sets of activations [4], and the number of redundant sets of activations that are solutions for a given task makes predicting muscle activation difficult [5].

In this paper we explore these ideas using the 3D biomechanical modelling platform Artisynth (www.artisynth.org; e.g., [6][7]), specifically in the context of tongue bracing. The sides of the tongue are in contact with the upper molars during the vast majority of speech, and this bracing of the tongue against the molars requires dedicated muscle activation [4]. We compare the sets of neuromuscular activations that result in tongue bracing with the sets that result in the full oral closure stage of a swallow, which has the tongue pressed against the roof of the mouth as well as the upper molars. Although there is no guarantee that in such a high-dimensional, non-linear activation space the set of activations resulting in the full oral swallowing closure will overlap with those that result in tongue bracing, our hypothesis generates the prediction that these activation sets will overlap, providing a plausible starting point for learners' searches of the activation space.

* connor.joseph.mayer@gmail.com
† francois.roewerdespres@usask.ca
‡ ian.stavness@usask.ca
° gick@mail.ubc.ca

2 Methods

We used muscle-driven tongue simulations to examine the muscle effort required to establish various types of tongue-palate contact. All possible sets of muscle activations in the tongue model were generated at three activation levels (0%, 20%, and 50%) for a group of ten muscles: superior longitudinal (SL), inferior longitudinal (IL), transverse (TRANS), verticalis (VERT), hyoglossus (HG), mylohyoid (MH), styloglossus (STY), and the posterior, medial, and anterior genioglossus (GGP, GGM, and GGA respectively). This generated 3^{10}, or approximately 60,000, activations. Virtual contact sensors were placed on the hard palate and upper teeth of the model to detect contact between the tongue and these surfaces. This allowed us to partition the activation space into four different types of contact. Bilateral contact is defined as lateral contact on both sides of the palate. Anterior contact is contact in the anterior region of the palate. Anterior-bilateral contact is bilateral contact as well as anterior contact: this is a representative example of tongue bracing during speech. Swallowing contact is bilateral, back, and mid contact, representing the end of the oral phase of swallowing, immediately after the tongue has moved the bolus into the pharynx. A more detailed methodological description of the simulations and sensor regions can be found in section 4.1 of [4].

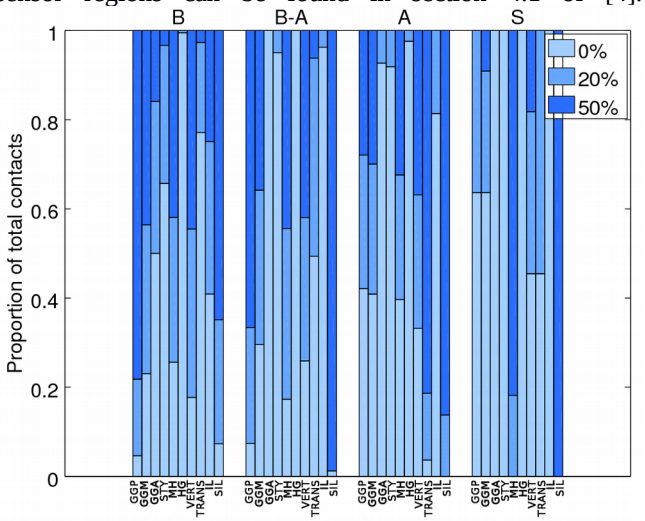


Figure 1: The proportion of the total number of each contact type (B: bilateral, B-A: bilateral-anterior, A: anterior, S: swallowing) with each muscle at 20% and 50% activation.

3 Results

Of the 60,000 different combinations of muscle activations, 1000 resulted in bilateral contact, 247 in anterior contact, 81 in anterior-bilateral contact, 11 in swallowing contact, and 57,829 in other types of contact or no contact. Only about

2% of the total activations matched any of our defined contact types.

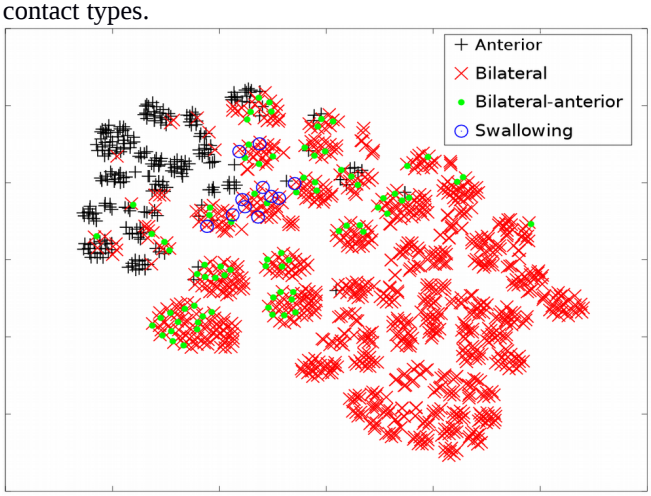


Figure 2: A two-dimensional t-SNE plot of the activation space.

Figure 1 shows the distribution of excitation levels for each muscle in each type of contact. The swallowing contacts are the simplest, consisting primarily of contributions from the SL and MH. This is broadly consistent with what is known about muscle use in the oral phase of swallowing: the MH plays an important role in raising the tongue body, and the SL further elevates the surface of the tongue to make contact with the hard palate. The activations for the bilateral and anterior-bilateral conditions still show a prominent role for these two muscles, but include many other activations necessitated by the more complex tongue shapes required. t-Distributed Stochastic Neighbor Embedding (t-SNE) [8] was used to visualize the distributions of the different contact types in activation space in Figure 2. t-SNE is an iterative algorithm that maps from high-dimensional to low-dimensional space using an optimization function that prioritizes maintaining the Euclidean distance between each point and its neighbours. As a result, this visualization captures the relationships between the points in the original ten-dimensional space. The many clusters in Figure 2 are the result of our sampling methodology: because there were only three activation levels used for each muscle, it resulted in an uneven distribution in the activation space, and this is reflected in the figure.

There is a fairly clear boundary between contact types without bilateral contact (anterior) and contact types with bilateral contact (bilateral, anterior-bilateral, and swallow). There are a large number of redundant solutions for bilateral and, to a lesser degree, anterior-bilateral and swallowing contact, mirroring the redundancy typically observed in biomechanical simulations [5]. However, each cluster of activations resulting in swallowing contact is contiguous with clusters of activations resulting in anterior-bilateral and bilateral contacts, indicating they share similar activations.

4 Discussion

The muscle activations found in the swallowing contacts are a subset of the activations found in the anterior-bilateral and the bilateral conditions: the SL and MH play the most significant role in swallowing contacts, and they play significant roles in anterior-bilateral and bilateral contact as well, but with additional muscle activations needed to produce the more complex tongue shapes for these types of contact, such as depressing the mid-sagittal region of the tongue and raising the tip. This is consistent with other biomechanical studies that have found that simpler, innate movements are subsequently refined to produce the more complex movements used in lip and jaw control in speech [5] and locomotion [9], and indicates that movements used in swallowing are suitable candidates for bootstrapping the learning of subtler speech movements such as bilateral tongue bracing.

The distribution of the swallowing contacts in the activation space also supports this idea. Even within the limited activation space defined by three activation levels per muscle, only a small number of activations result in any of the contact types we are investigating (about 2% of activations). An unstructured search of the full activation space would thus prove costly and inefficient. Although it is impossible to speculate on which activations are favoured by speakers based on this simulation alone, it is telling that each activation resulting in swallowing contact is contiguous with a cluster of bilateral and anterior-bilateral contact activations, and that the cluster containing the majority of swallowing contacts contains no instances of anterior contact. This indicates that swallowing activations could be a useful starting point for learning bilateral and anterior-bilateral speech movements by acting as a heuristic starting point for a search of the activation space.

Acknowledgments

We acknowledge funding from NSERC Discovery Grants to the third and fourth authors.

References

[1] Peter MacNeilage. (2008). The Origin of Speech. Oxford.
[2] M. Studdert-Kennedy and L. Goldstein (2003). Launching language: The gestural origin of discrete infinity. In M. Christiansen and S. Kirby (eds.), *Language Evolution*. Oxford.
[3] J.R. Green, C.A. Moore, M. Higashikawa, and R.W. Steeve. (2000). The Physiologic Development of Speech Motor Control: Lip and Jaw Coordination. *J. Sp., Lang. & Hear. Res.*, 43:239.
[4] B. Gick, B. Allen, F. Roewer-Despres, and I. Stavness (in pr.). Speaking tongues are actively braced. *J. Sp., Lang. Hear. Res.*
[5] Gerald E. Loeb. (2012). Optimal isn't good enough. *Biological Cybernetics*, 106:757.
[6] I. Stavness, J.E. Lloyd, and S.S. Fels (2012). Automatic Prediction of Tongue Muscle Activations Using a Finite Element Model. *J. Biomechanics*, 45:2841.
[7] B. Gick, P. Anderson, H. Chen, C. Chiu, H.B. Kwon, I. Stavness, L. Tsou, and S. Fels (2014). Speech function of the oropharyngeal isthmus: A modeling study. *Compu. Meth. Biomech. & Biomed. Eng.: Imaging & Visualization*, 2:217.
[8] L.J.P. van der Maaten and G.E. Hinton (2008). Visualizing High Dimensional Data Using t-SNE. *J. Mach. Lear. Res.*, 9:2570.
[9] N. Dominici, Y.P. Ivanenko, G. Cappellini, A. d'Avella, V. Mondi, M. Cicchese, A. Fabiano, T. Silei, A. Di Paolo, C. Giannini, R.E. Poppele, and F. Lacquaniti (2011). Locomotor primitives in newborn babies and their development. *Sci.*, 334:997.

PERCEPTUAL LEARNING OF SPEECH SOUNDS: A BIAS FOR 'SIPPER' OVER 'ZIPPER'?

Molly Babel [*1], **Zoe Lawler** [†1], **and Carolyn Norton** [‡1]
[1]Department of Linguistics, University of British Columbia

1 Introduction

Perceptual learning has been proposed as a cognitive mechanism that can account for listeners' adeptness at understanding a wide range of accents and spectral degradations. Our focus is on lexically-guided perceptual learning, whereby a listener's existing phonetic categories are updated with the support of clues from the lexical context [1]. In lexically guided perceptual learning experiments, listeners are presented with, for example, a word like *castle* pronounced not with a canonical /s/, but a fricative that is ambiguous between /s/ and /ʃ/. After exposure to such items, researchers find modifications in the distribution of listeners' /s/ categories, such that more /ʃ/-like sounds are categorized as /s/ or higher word recognition rates for words containing the ambiguous fricative. These changes in behaviour are considered evidence of perceptual learning.

While perceptual learning is a robust phenomenon (see [2] for a review), little is understood about its limits. In this study we test whether listeners show improved learning of phonetically natural shifts in pronunciation. Specifically, we investigate how listeners adapt in the face of a phonetically natural fricative devoicing pattern (e.g., [bɪzi] to *[bɪsi]) compared to a phonetically less-natural fricative voicing pattern (e.g., [mɛdɪsɪn] to *[mɛdɪzɪn]). The aerodynamic constraints of voicing make fricative devoicing patterns more common cross-linguistically.

2 Method

2.1 Subjects

Forty-eight adult volunteers between the ages of 19 - 60 were recruited from the metro Vancouver community ; the majority were students at the University of British Columbia (n = 45). All participants were native speakers of English and reported no speech or hearing impairments.

2.2 Stimulus Materials

Sixty filler sentences and twenty critical sentences containing either the shifted (experimental) or unshifted (control) critical sibilant were determined to be highly-predictable following an online cloze task to assess semantic predictability, and were used as the exposure phase materials. All multisyllabic critical words contained intervocalic alveolar fricatives (/s/ or /z/) and were in sentence-final position. Materials for the lexical decision task consisted of 100 filler words, 60 phonotactically-legal filler nonwords, 20 critical words that were exposed to the listener during the sentence exposure phase, and 40 novel critical words. Efforts were made to ensure that no other sibilant fricatives other than the critical sibilants [s] and [z] were present in any of the materials. However, four words containing an additional sibilant fricative were later identified. With the exception of [h] and [θ], no other fricatives were present in the selected stimuli.

All stimuli were naturally produced by a young adult male speaker of the local Vancouver dialect ; the critical shifted words used in the exposure sentences and in the lexical decision task were not artificially synthesized, following [3]. All materials were recorded using a professional-grade microphone and normalized to 72 dB. No discrepancies in voicing quality were recorded in any of the materials containing the critical sibilant when transcribed by two trained linguists naive to the goals of the study.

2.3 Procedure

Exposure Phase

Half of the /z/ condition participants were assigned to the shifted-/z/ group and exposed to high-predictability sentences containing a sentence-final word that would typically contain [z], (e.g., 'dizzy' [dɪzi]), but was produced with an [s] in place of the [z], rendering [dɪsi] (hereafter referred to as shifted-/z/.) The remaining half of /z/ condition participants were assigned to the control-/z/ group, in which [z] in the critical words in the exposure sentences was produced with the standard pronunciation. The same procedure was used for the shifted-/s/ group and the control-/s/ group. The manipulated variable between subjects was exposure to either the shifted (experimental) or unshifted (control) sibilant.

Eighty randomly ordered sentences (20 critical, 60 filler) were presented consecutively to participants over headphones at a comfortable listening level. Participants were informed to listen carefully to the sentences, but otherwise were not required to do anything while listening to the presented stimuli. Each sentence was separated by a 2000 ms pause.

Lexical Decision Task

A lexical decision task immediately followed the exposure. Participants were presented with a single item over headphones (i.e., nonword, word, exposed shifted-critical word, or novel shifted-critical word) and were asked to classify the item as a 'word' or 'not a word' by using the button box provided. The buttons '1' or '5' for 'word' and 'not a word' were counterbalanced across participants. These response options were visually presented orthographically on a computer monitor. Participants were given up to 1500ms to respond. All items were randomized across participants and the total number of trials (220) corresponded with the total number of word

*. molly.babel@ubc.ca
†. zlawler@alumni.ubc.ca
‡. carolyn.norton@alumni.ubc.ca

and nonword items. All critical words presented during the exposure phase were presented in the lexical decision task. Novel words containing either intervocalic /s/ or /z/ were presented with the shifting pattern (voiced to voiceless, or voiceless to voiced) corresponding to the participant ?s condition. Crucially, critical items were consistently presented in their shifted form during the lexical decision task.

3 Results

A glmer model comparing word endorsement rates in the lexical decision task for the /s/-shifted condition found no differences in critical words relative to control. This is shown in Figure 1. In this figure, filler words, exposure critical words, novel critical words, and filler nonwords are presented with the mean proportion of time these items were responded to as real words. Filler words and filler nonwords receive high and low word endorsement rates, respectively. The critical items pattern between the two.

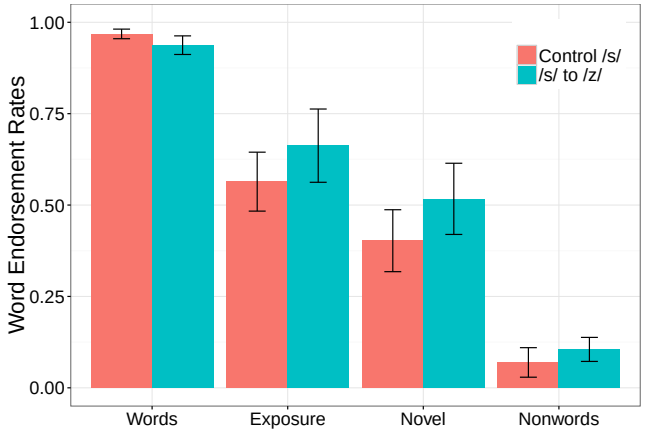


Figure 1: Word endorsement rates by lexical status for shifted /s/-words.

An equivalent model was run for the /z/-shifted condition and found evidence for learning : critical items received higher word endorsement rates in the /z/-shifted condition relative to the control group ($B = 1.31, SE = 0.57, z = 2.3, p = 0.02$). These results are shown in Figure 2. A comparison of performance on exposure words compared to novel words found no significant differences : listeners readily generalized the novel pronunciation shift to items they were not exposed to in the sentence exposure phase.

4 Discussion

Perceptual learning was found in the shifted-/z/ condition, with participants accepting both exposed and novel shifted forms of [z]→[s] items as words after exposure to shifted pronunciations compared to the /z/-shifted control condition. There was no effect of exposure noted for the shifted-/s/ condition, supporting a bias for perceptually learning devoiced fricatives over voiced fricatives. This research offers insight into the limitations of the perceptual learning process.

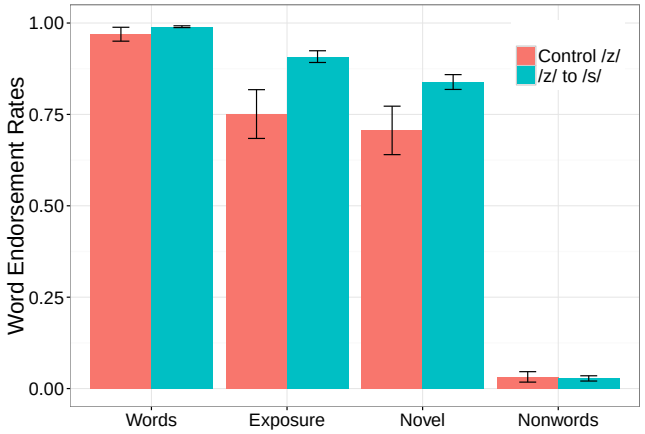


Figure 2: Word endorsement rates by lexical status for shifted /z/-words.

Listeners appear to adapt more readily to phonetically more natural changes, potentially tapping into pre-existing shifts in experienced phonetic distributions. Listeners likely have real-world experience perceiving underlying voiced fricatives as devoiced, as devoicing patterns are common, while the reverse pattern may be less likely ; we intend to confirm this intuition in a corpus study. In the context of this experiment, adaptation to the /z/ to /s/ change may be exploiting those phonetic memory traces.

5 Conclusions

These results suggest that perceptual learning processes may be guided by previous perceptual experiences or phonetic naturalness. Listeners do not learn all ambiguous pronunciations, at least not after a short exposure phase in a lab setting. Understanding the limits and constraints on perceptual learning mechanisms is important in determining which features or aspects of the perceptual learning process are wholly perceptual in nature, as opposed to post-perceptual decision biases. Moreover, understanding which features of the speech signal, elicit more or less robust learning serves as a stepping-stone for researchers to investigate how linguistic factors interact with a multitude of indexical and social cues that broaden or bias our ability to accommodate phonetic variation.

Acknowledgments

We would like to thank the Speech In Context Lab, UBC, and Chase Lawler for your patient dedication to this project.

References

[1] D. Norris, J. M. McQueen, and A. Cutler. Perceptual learning in speech. *Cognitive Psychology.*, 2 :204–238, 2003.

[2] Sven L Mattys, Matthew H Davis, Ann R Bradlow, and Sophie K Scott. Speech recognition in adverse conditions : A review. *Language and Cognitive Processes*, 27(7-8) :953–978, 2012.

[3] Kodi Weatherholtz. *Perceptual learning of systematic cross-category vowel variation*. PhD thesis, The Ohio State University, 2015.

DOES TRAINING TO PERCEIVE L2 ENGLISH VOWELS IN ONE PHONETIC CONTEXT TRANSFER TO OTHER PHONETIC CONTEXTS

Ron I Thomson*
Dept. of Applied Linguistics, Brock University, St. Catharines, ON, Canada, L2S 3A1

1 Introduction

Implicit in most second language (L2) pronunciation teaching materials is a belief that training learners to perceive and produce a target phoneme in one word will generalize to their perception and production of the same sound in other words. This view is at odds with Flege's Speech Learning Model [1], which argues that knowledge of a phonemic category in one phonetic context does not necessarily transfer to other contexts. While some have found evidence that categorical learning can generalize to new words [2, 3], others have demonstrated that there are limits on such transfer, in both perception and production [4-6]. Instead, L2 learners' pronunciation of particular phonemic categories appears to largely emerge word-by-word, rather than category-by-category [7].

Previous research in this area has typically explored a small number of training contexts and, likewise, tested the generalizability of learning to a small number of new contexts. The present study is more extensive in scope, and aims to uncover whether transferability of learning to new contexts is indeed the exception, or relatively commonplace.

2 Method

The data reported here come from a larger study examining the impact of High Variability Phonetic Training (HVPT) on L2 speech. HVPT trains learners to perceive target sounds produced by multiple talkers in multiple phonetic contexts, under the assumption that variability promotes category development (see [4-5] for a detailed overview).

2.1 Participants

Fifteen volunteer participants (five male; ten female) were learning English as a Foreign Language in Bogotá, Colombia (*M* age = 31; range 20-60). All were taking drip-feed English language classes (limited to 1-3 hours per week) and spoke English at a beginner (n=8) or intermediate (n=7) proficiency level. None had previously lived in an English speaking environment. All reported normal hearing.

2.2 Training procedure

Participants were trained to identify ten English vowels (/i/, /ɪ/, /e/, /ɛ/, /æ/, /ɒ/, /ʌ/, /o/, /ʊ/ and /u/ embedded in CV (consonant + vowel) and CVC (consonant + vowel + consonant) syllables, using a customizable researcher mode of the free web-based pronunciation training application, English Accent Coach [8]. This HVPT application employs 30 distinct voices (15 male;15 female) to present each vowel in CV or CVC combinations. Learners must click on the International Phonetic Alphabet (IPA) symbol for the vowel they perceive and are given immediate auditory and visual feedback; when they make an incorrect choice, they hear a buzz and see the correct symbol highlighted in red. They must click on the correct symbol in order to continue (see [8] for details).

Participants completed 40 sessions over a two-month period, but never more than two sessions in a given day (see Table 1 for a detailed breakdown of sessions). The first four sessions introduced learners to IPA symbols, and comprised 50-100 items each. The remaining 36 sessions trained learners to perceive the ten target vowels in CV and CVC syllables, and comprised 200 items each (see [4-6] for a rationale for using open syllables).

Table 1: Training contexts for each of 40 sessions.

Session numbers	Phonetic context in which ten English vowels were embedded
1-4 (phonetic symbol familiarization)	/h + V/
5-9, 39-40	/h + V/
10, 17	/b,d,g + V/
11, 18	/p,t,k + V/
12, 19	/v, ð, z + V/
13, 20	/f, θ, s + V/
14,21	/n, m + V/
15, 22	/ ʃ, tʃ, dʒ + V/
16, 23	/l, ɹ, j, w + V/
24, 27, 30, 33	/b, d, g, p, t, k + VC/
25, 28, 31, 34	/v, ð, z, f, θ, s + VC/
26, 29, 32, 35	/l, ɹ, j, w, m, n + VC/
36, 37, 38	All CVCs

The primary training context was /hV/. Not only was this context trained and tested during the first five sessions, it was also tested again during the final two sessions to determine if the intervening 30 training sessions, which incorporated all English consonantal contexts (with the exception of /ʒV/), resulted in any further improvement in the /hV/ context. All other contexts, grouped by manner and/or voicing features (e.g., voiced stops, followed by voiceless stops, followed by voiced fricatives, etc.), were trained at least twice in order to measure improvement over time, both between and within phonetic contexts.

2.3 Predictions

The extensive quantity of perceptual training in this study, and the incorporation of all English consonantal contexts, allows us to answer the question of whether perceptual learning of L2 English vowels proceeds in a primarily context independent or primarily context dependent manner. If the former, we should expect that the

* rthomson@brocku.ca

perception of English vowels should develop in a relatively linear fashion, largely uninterrupted by changes in the phonetic contexts in which the vowel is taught. Conversely, if learning occurs in a context dependent manner, we should see the learners' vowel identification (ID) scores return to near baseline each time the consonantal contexts changes. Finally, it is possible that category learning progresses somewhere between context independency and dependency, resulting in some weak evidence of transfer of learning as contexts change.

3 Results

Figure 1 provides a clear indication that for vowels in /hV/ syllables, learning is context dependent. While a clear linear improvement in mean ID scores across initial /hV/ sessions 5 to 9 is evident, when a new context was introduced in session 10, the mean ID scores returned to baseline. Furthermore, from sessions 10 to 16, during which time the training contexts changed session-by-session, no impact on ID scores is seen across any contexts.

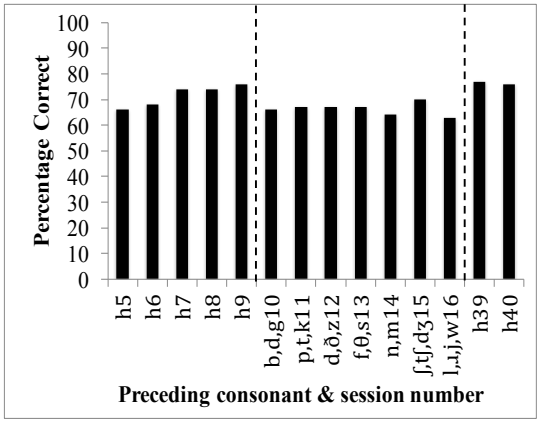


Figure 1: Average correct vowel identification scores for sessions 5-16 and 39-40. Dotted lines indicate change in phonetic context.

Bonferroni-adjusted paired samples t-tests revealed that while mean ID scores for vowels in the primary /hV/ context significantly improved from session 5 to 9, $t(14)= 5.3298$, $p<.001$, no further improvement occurred from session 9 to 39, $t(14)=2.0603$, $p=.06$, despite learners receiving 30 intervening training sessions focussing on the same vowels, albeit in different phonetic environments.

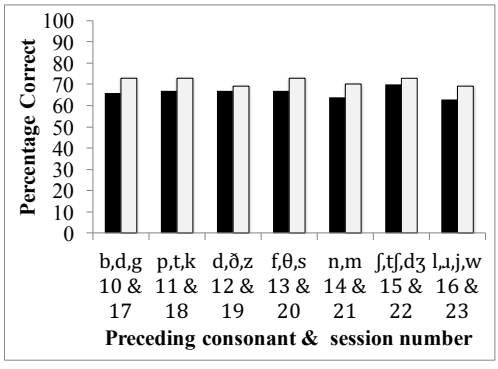


Figure 2: Average correct vowel identification scores for sessions where the same contexts were trained twice, one week apart.

Figure 2 further illustrates context dependency in L2 vowel category learning, comparing mean vowel ID scores within the same contexts over time. For example, ID scores for vowels in /b,d,g +V/ contexts at sessions 10 and 17 are shown to improve over time, and the same pattern is evident for each of the other six contexts illustrated, as well as for those contexts repeated during sessions 24-35 (see Table 1). Only those contexts repeated during sessions 36-38 showed no improvement. For these sessions, vowels were presented in all possible CVC contexts, not just subsets restricted to related phonetic contexts. As such, the overwhelming amount of variation may have made learning difficult.

4 Conclusion

The results of this study provide little support for the belief that L2 vowel category learning proceeds in a context independent manner. Rather, it provides evidence that L2 categories must be learned context-by-context, or perhaps word-by-word (see [7]), with little if any knowledge transfer from one phonetic context to the next. It should be noted that this study trained learners using phonetic contexts that were clustered together in terms of voice and manner properties. It may be the case that training in /bV/ contexts, for example, can transfer to /dV/ or /gV/ contexts, both of which also begin with voiced stops. Since these contexts were all trained simultaneously, this remains unresolved.

Acknowledgments

The author thanks T. Derwing and M. Munro for input on the study's design, the learner volunteers, and the Social Sciences and Humanities Research Council of Canada.

References

[1] J. E. Flege. Second-language speech learning: Theory, findings, and problems. In W. Strange (Ed.) *Speech perception in linguistic experience: Theoretical and methodological issues*. Pp. 229-273. Timonium, MD: York Press, 1995.

[2] X. Wang, and M. J. Munro Computer-based training for learning English vowel contrasts. *System*, 32, 539–552, 2004.

[3] A. Carlet and J. Cebrian. Training Catalan speakers to identify L2 consonants and vowels: A short-term high variability training study. *Concordia Working Papers in App. Ling.*, 5, 85-98, 2014.

[4] R. I. Thomson. Computer Assisted Pronunciation Training: Targeting second language vowel perception improves pronunciation. *CALICO Journal*, 28, 744-765, 2011.

[5] R. I. Thomson. Improving L2 listeners' perception of English vowels: A computer-mediated approach. *Language Learning, 62,* 1231-1258, 2012.

[6] R. I. Thomson, T. M. Nearey, and T. M. Derwing. A modified statistical pattern recognition approach to measuring the crosslinguistic similarity of Mandarin and English vowels. *Journal of the Acoustical Society of America, 126,* 1447-1460, 2009.

[7] R. I. Thomson and T. Isaacs, T. Within-category variation in L2 English vowel learning. *Canadian Acoustics, 37,* (3), 138-139, 2009.

[8] R. I. Thomson. *English Accent Coach* [Computer program]. Version 2.3, 2012 retrieved 27 July 2016 from www.englishaccentcoach.com

BREATHING POSITION AND THE PERCEPTION OF CONSONANTS

Henny Yeung *[1,3], Mark Scott †[2] and Lionel Granjon ‡[3]
[1]Department of Linguistics, Simon Fraser University, Burnaby, BC, Canada
[2]Department of English Literature & Linguistics, Qatar University, Doha, Qatar
[3]Laboratoire Psychologie de la Perception, CNRS & Université Paris Descartes, Paris, France

1 Introduction

Several theories of speech perception and acquisition suggest a role for the motor system in auditory processing. One class of theories suggests that motor information plays an especially important role when the perceived speech is noisy, ambiguous, or otherwise hard to process, and that the motor system can be used to make sensory predictions about the incoming auditory signal to fill in missing information [1] (see also [2]). A source of evidence for these theories comes from studies that require experimental participants to move their vocal-tract articulators in speech-like ways while perceiving ambiguous or noisy speech [3]–[6]: Results show that these movements influence the perception of speech signal in ways that other non-auditory sources of complementary speech information can influence speech perception, like the McGurk effect (i.e., seeing visual speech changes the perception of auditory speech [7]).

It remains unclear what precise aspects of speech planning and production generate sensory predictions, which might influence auditory speech perception. Broadly speaking, previous reports suggest two distinct sources of information. First, speech perception seems to be modulated by the internal rehearsal or planning of speech gestures (inner speech), even if no explicit movements are made [3], [8]. Second, peripheral sensory information about the actual dynamic movements of speech articulators seems to result in a similar effect. For example, when there is an external source eliciting sensory feedback from skin receptors—a robot deforming facial skin in speech-like ways—speech perception is modulated, even if perceivers are not explicitly engaging in any speech rehearsal or planning [9].

Several questions about these sensorimotor influences on perception persist. In the present report, we focus on the specificity of articulatory information in these pathways, asking what kinds of motor activity are sufficient to trigger sensory predictions, and thus influence speech perception. We considered an extreme case in this study, as participants were asked just to maintain a static vocal tract position (i.e., breathing either through the nose or through the mouth) while perceiving speech. This tested the idea that speech-related motor-induced sensory predictions are elicited in an automatic, pervasive, and broad manner, perhaps even when simply breathing in a particular way (i.e., having vocal tract configurations that only loosely match real speech gestures).

* henny_yeung@sfu.ca
† mark.a.j.scott@gmail.com
‡ lionel.granjon@parisdescartes.fr

2 Methods

Experimental participants performed a speech identification task in two conditions. The Experimental Condition involved a classification of an /ada/-/ana/ continuum, while the Control Condition involved classification of a /ada/-/aga/ continuum. In coarse terms, continua endpoints differed either in the position of the velum (up or down for /ada/ and /ana/, respectively), or the position of the tongue tip (raised or lowered for /ada/ and /aga/, respectively).

The critical manipulation involved instructions to the participants to either breathe through the mouth or nose while performing the task. This necessarily changes the position of the velum (up or down), and we predicted that breathing position would have an effect on identifications in the Experimental Condition, but not the Control Condition.

2.1 Participants

Forty-nine native monolingual French speakers participated. All were between the ages of 18-30 with normal hearing.

2.2 Stimuli

A female French native speaker was recorded saying /ada/, /ana/, and /aga/. One token of each disyllable was then used as the endpoints to create the two continua, each with 10,000 steps, using the STRAIGHT software package [10].

2.3 Procedure

Participants were tested using a laptop computer with headphones in a quiet room. In total, there were 8 experimental blocks presented in a counterbalanced order: 2 breathing positions (Nose and Mouth) x 2 conditions (Experimental and Control) x 2 block repetitions.

Within each block, a double staircase procedure was used to sample participants' perceptual boundaries [11]. A "high" staircase started at token 7600, which was closer to either the /ana/ or /aga/ continuum endpoints, while a "low" staircase started at token 2400, which was at the /ada/ endpoints of the continua. Sixteen staircase steps of decreasing sizes were used, and trials alternated randomly between either the high or low staircase.

In each trial one token was presented, after which a button was pressed to indicate which continuum endpoint was heard. A new trial began after a 150ms delay.

3 Results

3.1 Analysis

A mixed-effects logistic regression was used to model responses on each continua separately, with the expectation that there would be an effect of breathing place in the Experimental Condition (the /ada/-/ana/ continuum), but not in the Control Condition (the /ada/-/aga/ continuum).

For fixed effects, breathing place (Mouth or Nose) and continuum token (coded as 0 to 100) were both entered into the model. By-subjects random slopes were entered for breathing place, continuum token, and staircase (four different staircases were run for each condition: high and low staircases in each of the two block repetitions). All analyses were conducted using the *lme4* package in R [12].

3.2 Experimental Condition (/ada/-/ana/)

Results of the mixed effects logistic regression suggested that both fixed factors (breathing place and continuum token) influenced the likelihood of participants' category response. For breathing place, switching from mouth breathing to nose breathing significantly affected response probability ($\chi 2(1) = 8.48$, $p = 0.0036$), and the model estimate (Table 1) suggested an increase in the probability of /ana/ responses when breathing through the nose.

As predicted, when presenting tokens from higher on the continuum (more /ana/-like), responses were also more likely to be perceived as /ana/ ($\chi 2(1) = 83.45$, $p < 0.001$).

	Estimate	S.E.
Intercept	-3.43	.39
Breathing Place	**.34**	**.11**
Continuum Token	**.12**	**.01**

Table 1: Fixed effects in the Experimental Condition (bold font indicates a significant fixed factor at α = .01).

3.3 Control Condition (/ada/-/aga/)

Results of the mixed effects logistic regression suggested that only one of the fixed factors (continuum token) influenced the likelihood of participants' category response. For breathing place, switching from mouth breathing to nose breathing did not significantly affect the response probability ($\chi 2(1) = 1.91$, $p = 0.17$), and as can be seen in Table 1, the standard error of the model's coefficient indicated no consistent pattern of breathing place.

As predicted, when presenting tokens from higher on the continuum (more /aga/-like), responses were also more likely to be perceived as /aga/ ($\chi 2(1) = 24.10$, $p < 0.001$).

	Estimate	S.E.
Intercept	-2.08	.45
Breathing Place	.17	.12
Continuum Token	**.044**	**.008**

Table 2: Fixed effects in the Control Condition (bold font indicates a significant fixed factor at α = .01).

4 Discussion

Results suggested that simply instructing experimental participants to breathe through the nose biased speech identifications towards the /ana/ category on the /ada/-/ana/ continuum in the Experimental Condition. No such effect was found for the /ada/-/aga/ continuum in the Control Condition.

5 Conclusion

These results are striking, as they show the effect of highly subtle motor information on auditory speech identification. In other words, simply instructing participants to breathe through their mouth or nose (and thus changing the position of the velum) can affect perceptual judgments of ambiguous speech sounds (between /ada/ and /ana/). Results extend previous studies in showing that speech identification is influenced by motor planning not linked to auditory imagery, and by motor movements that are not speech-like.

Together, results support speech-processing theories that incorporate sensory predictions from the motor system [1], [2]. The present data suggest that such sensory predictions are highly automatic and pervasive, generated from even very basic information about static positions of oral articulators.

References

[1] J. I. Skipper, V. van Wassenhove, H. C. Nusbaum, and S. L. Small. Hearing lips and seeing voices: How cortical areas supporting speech production mediate audiovisual speech perception. *Cereb. Cortex*., 17 :10, 2007.

[2] M. J. Pickering and S. Garrod. An integrated theory of language production and comprehension. *Behav. Brain Sci.*, 36 :4, 2013.

[3] M. Scott, H. H. Yeung, B. W. Gick, and J. F. Werker. Inner speech captures the perception of external speech. *J. Acoust. Soc. Am.*, 133 :4, 2013.

[4] M. Sato, E. Troille, L. Ménard, M.-A. Cathiard, and V. Gracco. Silent articulation modulates auditory and audiovisual speech perception. *Exp. Brain Res.*, 227, 2013.

[5] M. Sams, R. Möttönen, and T. Sihvonen. Seeing and hearing others and oneself talk. *Cogn. Brain Res.,* 23 :2–3, 2005.

[6] T. Mochida, T. Kimura, S. Hiroya, N. Kitagawa, H. Gomi, and T. Kondo. Speech misperception: speaking and seeing interfere differently with hearing. *PLoS One*, 8 :7, 2013.

[7] H. McGurk and J. MacDonald. Hearing lips and seeing voices. *Nature*, 264 :5588, 1976.

[8] M. Scott. Corollary discharge provides the sensory content of inner speech. *Psychol. Sci.*, 24 :9, 2013.

[9] T. Ito, M. Tiede, and D. J. Ostry. Somatosensory function in speech perception. *Proc. Natl. Acad. Sci. USA*, 106 :4, 2009.

[10] H. Kawahara, T. Irino, and M. Morise. An interference-free representation of instantaneous frequency of periodic signals and its application to F0 extraction. *ICASSP Proceedings*, 2011, pp. 5420–5423.

[11] T. N. Cornsweet. The staircase-method in psychophysics. *Am. J. Psychol*., 75, 1962.

[12] D. Bates, M. Mächler, B. M. Bolker, and S. C. Walker. Fitting linear mixed-effects models using lme4. *J. Stat. Softw.* 67 :1, 2015.

Audio-Aerotactile Integration In Infant Speech Perception

Megan Keough, Padmapriya Kandhadai, Janet F Werker, Bryan Gick

A growing body of literature provides evidence for the importance and integration of sensory modalities other than auditory in speech perception (e.g. vision, McGurk and MacDonald, 1976). Recent work (Gick & Derrick, 2009) has shown that aero-tactile information can influence speech perception. In this study, native English speakers were significantly more likely to perceive a consonant as aspirated when puffs of air were inaudibly applied to participants' skin, even if the token was unaspirated. One possible source of a learned mapping between the auditory and aerotactile modalities in this effect could be the sensation of air puffs produced during one's own speech. To test this possibility, the current study tests whether this effect requires production experience with aspiration, by testing for the effect in prelinguistic (6-8 month-old) English-acquiring infants. If one's own speech production experience is not necessary, the infants should show the same effect as adults. English-acquiring infants participated in an alternating/non-alternating sound presentation task, with a puff of air applied every other syllable to their neck. The tactile stimulus either reinforced or interfered with the aspiration distinction, thereby influencing the infant's perception of whether the trial was alternating. Measures of looking time, reflecting infants' baseline ability to discriminate between two sounds, will be presented. Results provide insight into the developmental trajectory and origins of aero-tactile integration.

SPEECH PRODUCTION - PRODUCTION DE LA PAROLE

Dental Overjet, Articulatory Compensation And Acoustics Of Speech
Lauretta Cheng, Tiffany Doe, Derek Lew, Melissa Henderson, Murray Schellenberg, Bryan Gick 204

Schlieren Study Of External Airflow During The Production Of Nasal And Oral Vowels In French
Jeffrey Rowell, Masaki Noguchi, B. May Bernhardt, Anthony Herdman, Bryan Gick, Murray Schellenberg 206

(Canadian) Raise Your Vowels In Song
Murray Schellenberg, Avery Ozburn 208

Individual Variations And Gradience In English Palatalization
Jae-Hyun Sung 210

Japanese Efl Learners' Production Of Pronouns And Articles In English: Evidence For L2 Prosodic Structures
Noriko Yamane, Noriko Yoshimura, Atsushi Fujimori 212

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 214

DENTAL OVERJET, ARTICULATORY COMPENSATION AND ACOUSTICS OF SPEECH

Lauretta Cheng [*1], Tiffany Doe [1], Derek Lew [1], Melissa Henderson [1], Murray Schellenberg [†1], and Bryan Gick [§1, 2]
[1] Dept. of Linguistics, University of British Columbia, 2613 West Mall, Vancouver, BC, Canada, V6T1Z4
[2] Haskins Laboratories, 300 George St., New Haven, CT, USA 06511

1 Introduction

Dentofacial abnormalities have been linked, with mixed evidence at times, to misarticulations before and after corrective surgery in severe cases [1, 2]. Many individuals, even in surgical populations, appear to achieve typical-sounding productions by using compensatory adaptation of the speech articulators [3]. Overjet, the horizontal overlap of upper maxillary and lower mandibular central incisors, can be both positive (upper protrusion) or negative (lower protrusion), and is one measure of malocclusion. Although a few studies have applied acoustic analysis to supplement perceptual data [2, 4], most previous research relating to malocclusion has been based on judgments of misarticulation rather than objective acoustic measures [5, 6]. As such, this study seeks to investigate the association between degree of overjet in non-surgical populations and speech production using acoustic analysis.

2 Methods

Participants were 20 native English speakers (mean age = 24.7 years, 5 males, 15 females), none of whom reported speech or hearing disorders. Stimuli consisted of 13 target words containing the English vowels /i, u, ɑ/, fricatives /s, z/ and affricates /ʃ, tʃ, dʒ/. Fricatives were placed in both word-initial and -final contexts while vowels were between two consonants. All words were presented in the sentence "Repeat the term _____ often" for a duration of three seconds using PsychoPy (version 1.81.02) software [7]. Participants were instructed to read the full sentence as quickly as they could to control for reading speech rate and to encourage more natural elicitations. The speech samples were recorded in a sound-treated booth with a Samson C03U USB multi-pattern condenser microphone on Audacity 2.1.2 at a sampling rate of 44.1kHz.

Dental overjet was measured by intraoral assessment, considered a reliable technique [8] using disposable metric rulers accurate to 1 mm. All measurements were recorded in millimetres. Other facial measures of mandible width and nose-to-lip distance were taken to prevent participants from learning the purpose of our study. They were asked to comment on their dentofacial history, such as having had braces or major dental surgery, and to fill in a questionnaire on language background information.

Incorrect elicitations were omitted; in total, 3146 tokens were included in the analysis. Target fricatives and vowels were annotated manually in Praat [9]. In the case of fricatives, the duration and four spectral moments (centre of gravity [COG], standard deviation, skewness and kurtosis) were extracted from the portion between one-quarter and three-quarters of the total duration. Measurements of the first three formants were taken from the midpoint of all vowels.

3 Results

Participants were split into four groups for analysis according to the overjet measurements, detailed in Table 1. Two-way Multivariate ANOVAs (MANOVAs) were run in R on formants values (vowels), and duration and spectral moments (fricatives), where the factors were Overjet group and Vowel (/i, u, ɑ/) or Fricative (/s, z, ʃ, tʃ, dʒ/), respectively.

Overjet	*n*	mean age	M	F
1 mm	5	24 years	1	4
2 mm	4	21 years	0	4
3 mm	5	26.4 years	2	3
4 mm	6	26.2 years	2	4

Table 1: Participant details by the four groups of overjet.

Two-way MANOVA results indicate that there was a statistically significant difference between vowels (Pillai's Trace = 0.80426, F(6, 94) = 10.5374, $p < 0.001$) by F1, F2 and F3. There were no statistically significant effects of overjet group on formants, nor for interactions between specific vowels and degree of overjet in the formant values.

There was a statistically significant difference between fricatives (Pillai's Trace = 1.30155, F(20, 316) = 7.6209, $p < 0.001$) by COG, standard deviation, and skewness and duration, but not kurtosis. There was a statistically significant difference between overjet groups across all fricatives (Pillai's Trace = 0.35501, F(15, 234) = 2.0938, p = 0.01096) in standard deviation, kurtosis and duration. Interactions between specific fricatives and overjet group were statistically non-significant in terms of spectral moment and duration values.

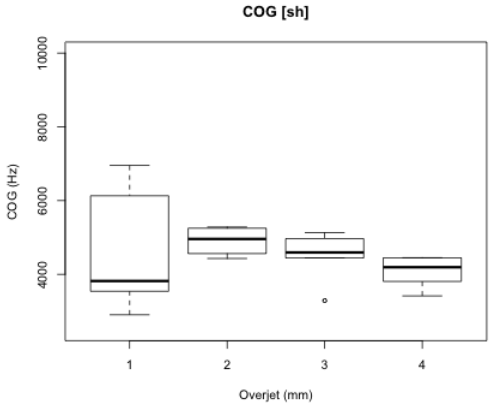


Figure 1: Mean COG values of /ʃ/ for each participant by overjet.

[*] lauretta.cheng@alumni.ubc.ca
[†] mhschellenberg@gmail.com
[§] gick@mail.ubc.ca

Some trends do appear in the means for post-alveolar and alveolar fricatives the spectral moments, shown in Table 2 and 3. In the post-alveolar consonants, trends show the 2-4 mm groups in a linear relationship, but with the 1 mm group reversing the direction. Alveolar fricatives pattern differently where both 1 mm and 4 mm groups appear to reverse the 2-3 mm trend in opposite directions.

Overjet	COG (Hz)	SD	Skewness	Kurtosis
1 mm	4656.3	1683.9	1.4642	4.4680
2 mm	4906.8	1980.4	1.1910	1.6420
3 mm	4668.7	1938.3	1.4143	3.1932
4 mm	4164.0	1864.0	1.8540	4.4530

Table 2: Means of spectral moments across post-alveolar sibilants.

Overjet	COG (Hz)	SD	Skewness	Kurtosis
1 mm	7811.9	1430.1	0.4066	2.5897
2 mm	8250.7	1407.1	0.2918	3.0457
3 mm	7939.0	1619.0	0.4566	2.7988
4 mm	8290.6	1267.6	0.1840	6.1731

Table 3: Means of spectral moments across alveolar fricatives.

4 Discussion

Degree of overjet was compared on acoustic measures of vowels and fricatives in speakers with normal (2-3 mm) and near-normal occlusion [10]. Both the acoustics and perceptions by trained listeners conclude that phonemes were clearly contrasted. Spectral moments for /s/ have been shown to be consistently characterized by higher COG and negative skewness values, representing larger concentration of high frequencies than /ʃ/ [11]. This indicates tongue placement closer to the upper incisors. If there was no compensation present across degrees of overjet, we might expect a linear trend. The small group sample sizes limit the results generalizability, but trends in the data are not linear.

The 1 mm speakers show acoustics with more *low* frequencies than those with normal occlusion; that is, the tongue is moving *away* from an anterior place of articulation. This suggests that speakers with smaller-than-normal overjet may be compensating by pulling their tongue back to create a more typical distance from the teeth for both alveolar and post-alveolar fricatives. This converges with previous findings on articulation with malocclusion, which found tongue compensation of this type [3, 6].

Speakers with 4 mm of overjet pattern differently between places of articulation. Post-alveolar consonants show lower frequencies than speakers with normal occlusion. On the other hand, alveolar fricatives show a trend to higher frequencies, which suggests that speakers with larger-than-normal overjet only compensate by moving the tongue forward for alveolar but not post-alveolar fricatives.

These differences may be explained by the necessity of precise tongue placement for alveolar fricatives [6], which causes 1 mm speakers to move the tongue posteriorly and 4 mm speakers to move anteriorly. In contrast, the place of articulation for post-alveolar production is less dependent on the distance to the teeth, so that 4 mm speakers do not need to compensate at all. The 1 mm speakers do appear to compensate, though this may be in order to clearly differentiate between places of articulation after having adjusted for the alveolars. If this is the case, unlike alveolar fricatives, the post-alveolar compensation may be for contrastive purposes only, as opposed to compensating for articulatory constraints due to dentofacial structure.

5 Conclusion

This experiment adds quantitative acoustic data to the literature on relationships between dental anomalies and speech, as well as on the role of compensation in the case of abnormal overjet. Articulatory compensation appears to be intact in speakers with non-clinical but abnormal overjet, both perceptually and acoustically. Future work with larger sample sizes and a larger range of overjet would help determine whether these trends can be extended further.

Acknowledgments

We thank our RAs Yik Tung Wong, Lisa Vu, and Kaining Xu, as well as Dr. Hannam, Dr. Pliska and Dr. Yen from the UBC Faculty of Dentistry who provided advice and recommendations for the dental measurements. This research is funded by an NSERC Discovery Grant to B. Gick and by NIH Grant DC-02717 to Haskins Laboratories.

References

[1] Ruscello, D. M., Tekieli, M. E., & Van Sickels, J. E. (1985). Speech production before and after orthognathic surgery: A review. *Or. Sur., Or. Med., Or. Path.*, 59, 10-14.

[2] Lee, A. S. Y., Whitehill, T. L., Viocca, V., & Samman, N. (2002). Acoustic and perceptual analysis of the sibilant sound /s/ before and after orthognathic surgery. *J. Or. & Maxillofac. Surg., 60*, 364-372.

[3] Johnson, N. C. & Sandy, J. R. (1999). Tooth position and speech – is there a relationship? *Ang. Ort., 69*, 306-310.

[4] Daniloff, R. G., Wilcox, K., & Stephens, M. I. (1980). An acoustic-articulatory description of children's defective /s/ productions. *J..Commun. Disord., 13*, 347-363.

[5] Laine, T. (1987). Associations between articulatory disorders in speech and occlusal anomalies. *Eur. J. Orthodont., 9*, 144-150.

[6] Vallino, L. D. & Tompson, B. (1993). Perceptual characteristics of consonant errors associated with malocclusion. *J. Or. Maxillofac. Surg., 51*, 850-856.

[7] Peirce, J. W. (2007). PsychoPy—psychophysics software in Python. *J. Neurosc. Meth., 162*, 8-13.

[8] Ovsenik, M. (2007). Assessment of malocclusion in the permanent dentition: Reliability of intraoral measurements. *Eur. J. Orthodont., 29*, 654-659.

[9] Boersma, P. & Weenink, D. (2015). Praat: doing phonetics by computer. Version 5.4.08, Available from http://praat.org/

[10] Darby M. L. & Walsh M. (2015). *Dental Hygiene: Theory and Practice* (4th ed.). St. Louis, MO: Elsevier Saunders.

[11] Jongman, A., Wayland, R. & Wong, S. (2000). Acoustic characteristics of English fricatives. *J. Acoust. Soc. Am, 108*, 1252-1263.

SCHLIEREN STUDY OF EXTERNAL AIRFLOW DURING THE PRODUCTION OF NASAL AND ORAL VOWELS IN FRENCH

Jeffrey Rowell[*1], Masaki Noguchi[2],**

B. May Bernhardt[*1], Anthony T. Herdman[****1], Bryan Gick[†2] and Murray Schellenberg[‡2]**

University of British Columbia ([1]Audiology and Speech Sciences; [2]Department of Linguistics)

1. Introduction

Schlieren imaging enables real-time visualization of airflow through refraction of light. The potential of schlieren imaging for speech research was first proposed nearly 40 years ago [1]. A recent study tested this proposal [2]. Phonetically trained raters coded schlieren recordings of French vowels as oral versus nasal [2] (Figure 1). Although raters discriminated between the oral and nasal vowels ($p < .05$), the study could not fully determine whether the perceptual ratings were based on airflow patterns alone. Thus, the current study was undertaken to investigate optical flow analysis in detecting differences between oral and nasal airflow patterns in the same schlieren recordings. Results show that nasal vowels are produced with a larger magnitude of nasal airflow than oral vowels, with some variation between speakers.

2. Method

2.1. Equipment

The schlieren system was based a single mirror setup used in previous research [3][4]. The system consisted of: (1) a 12-inch diameter spherical mirror with a focal length of 8 feet attached to an optical mirror mount allowing horizontal and vertical adjustments; (2) a battery-powered pinhole LED light source affixed on top of a camera; (3) a video camera capable of recording in high definition at 60 frames per second; and (4) a razor blade attached to an adjustable optics mount which permitted fine tuning of the blade's position. Participants were positioned to minimize distance between the mirror and speech airflow while also preventing accidental contact with the mirror. Participants held a piece of cardboard to redirect background airflow (i.e. heat rising from the participants' chest and legs) away from the mirror.

2.2. Participants and Recordings

Analysis was conducted on the same recordings from the previous perceptual study [2]. Nine native speakers of French were recorded reading a list of oral/nasal minimal pairs adapted from previous research [5] (*pas/paon, paix/pain, pot/pont*). Words were recorded three times. In order to reduce the burden on perceptual raters in the initial study, only one token per word, per participant was selected. Potentially confounding background airflow was minimized by selecting recordings with the greatest number of still frames immediately preceding initiation of /p/.

[*] jeff.rowell@alumni.ubc.ca
[**] mskngch@gmail.com
[***] may.bernhardt@audiospeech.ubc.ca
[****] aherdman@audiospeech.ubc.ca
[†] gick@mail.ubc.ca
[‡] mhschellenberg@gmail.com

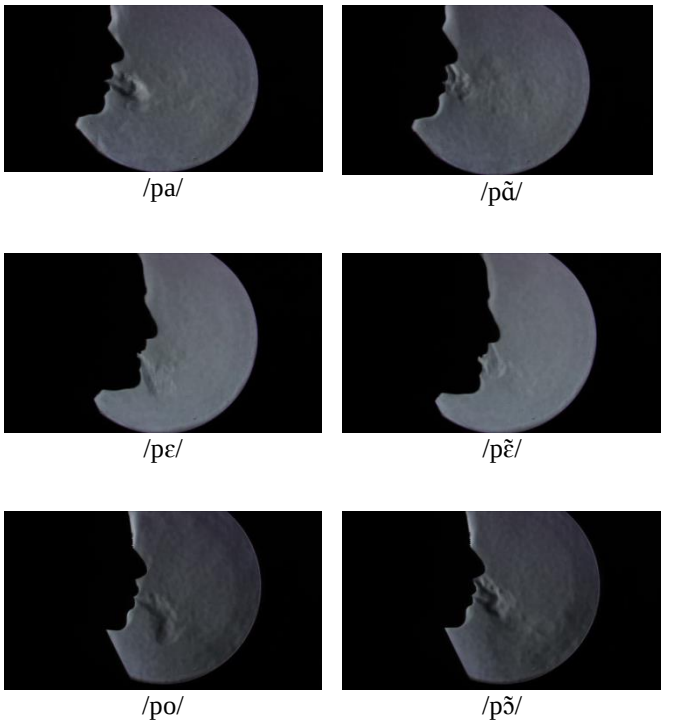


Figure 1: Schlieren recordings demonstrating the visual difference in airflow between oral and nasal vowels. Oral vowels show a single stream of airflow from the mouth and nasal vowels, two streams from both the nose and mouth.

2.3. Analysis

Airflow in the schlieren videos was measured using the Horn and Schunk's optical flow algorithm [6] implemented with the OpenCV [7]. The algorithm takes two consecutive video frames and calculates the direction and magnitude (i.e., speed) of each pixel's movement from one frame to the next (i.e., the displacement vector of the pixel). Displacement vectors are calculated for all pixels within an area of interest (AOI), and the sum of the displacement vectors was used as the measure of optical flow within the AOI.

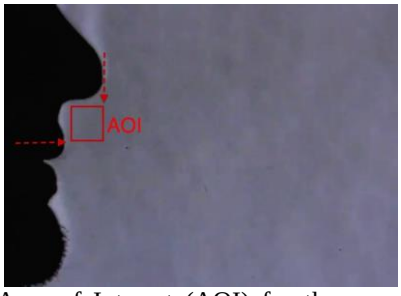


Figure 2. Area of Interest (AOI) for the measurement of nasal airflow

In order to measure nasal airflow, an AOI was defined under the nose (Figure 2): the right side of the AOI was vertically aligned with the tip of the nose; the lower side of the AOI was horizontally aligned with the height of the upper lip.

Airflow was measured during the production of the vowels. Because vowel tokens varied in duration, duration was normalized to percentage points. To compare the nasal airflow patterns of oral and nasal vowels, we obtained the dynamic measurements of the mean magnitude of airflow with 95% CI for oral and nasal vowels by applying a spline smoothing function (both with and across speakers) [8].

3. Results

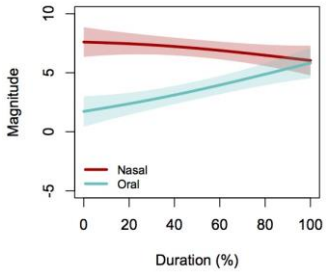


Figure 3. Magnitude of airflow for the oral (aqua) and nasal (red) vowels produced by nine French speakers

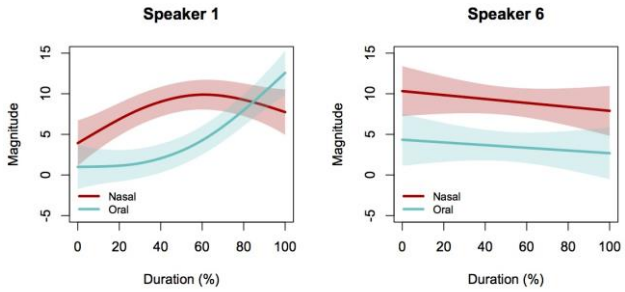


Figure 4. Magnitude of airflow for the oral (aqua) and nasal (red) vowels produced by Speaker 1 and Speaker 6

Figure 3 shows the magnitude of airflow for oral and nasal vowels across speakers. The magnitude of airflow at the onset of vowels was larger for nasal vowels than oral vowels, a consequence of greater nasal airflow in nasal vowels. The magnitude of airflow for nasal vowels became smaller at vowel offset. The magnitude of airflow for oral vowels became larger towards vowel offset. This apparently reflected air turbulence caused by the progression of oral airflow into the AOI rather than an increase in nasal airflow.

Speakers varied somewhat in nasal airflow patterns. For example, Speaker 1's nasal vowels had a gradual increase in nasal airflow across the vowel, whereas Speaker 6's nasal vowels showed notable nasal airflow from the onset of the vowel (Figure 4).

4. Discussion

Optical flow analysis of schlieren recordings shows promise in measurement of external airflow patterns of French oral and nasal vowels; nasal vowels were produced with a larger magnitude of nasal airflow than oral vowels. Airflow patterns varied between speakers, however, which suggests the need for further inquiry into relationships between airflow patterns and acoustic data across speakers.

Schlieren imaging's strength is its ability to visualize changes in the refractive index of transparent media. This is also a limiting factor because changes *within* a stream of air, particularly laminar flow, may be relatively stable compared to the airflow's leading edge. For example, we observed that the magnitude of nasal airflow for nasal vowels became smaller at vowel offset, but this may not have been due to decreased nasal airflow. Once nasal airflow in a small AOI becomes laminar, optical flow analysis may be less sensitive to the flow. Flow visualization may therefore be limited to speech sounds with turbulent flow.

References

[1] Davies, T. P. Schlieren photography: a tool for speech research. *Acoustics Letters* 3 (1979): 73-75.

[2] Rowell, J. Schlieren imaging: visualization of airflow in speech. Master's thesis, University of British Columbia, 2015.

[3] Settles, G. S. *Schlieren and shadowgraph techniques: visualizing phenomena in transparent media*. Springer, 2001.

[4] Tang, J. W., A. D. Nicolle, J. Pantelic, M. Jiang, C. Sekhr, D. K Cheong, and K. W. Tham. Qualitative real-time schlieren and shadowgraph imaging of human exhaled airflows: an aid to aerosol infection control. *PLoS One* 6, (2011): e21392.

[5] Carignan, C. An acoustic and articulatory examination of the "oral" in "nasal": The oral articulations of French nasal vowels are not arbitrary. *Journal of Phonetics* 46 (2014): 23-33.

[6] Horn, B. K. P., and B. G. Schunck. Determining optical flow. *Artificial intelligence* 17, (1981): 185-203.

[7] Barbosa, A. V., H. C. Yehia, and E. Vatikiotis-Bateson. Linguistically valid movement behavior measured non-invasively. In *AVSP-2008*, pp. 173-177. 2008.

[8] Chong, G. Smoothing spline ANOVA models: R package gss. *Journal of Statistical Software* 58, (2014): 1-25.

(CANADIAN) RAISE YOUR VOWELS IN SONG

Murray Schellenberg* **and Avery Ozburn†**
Department of Linguistics, University of British Columbia

1 Introduction

One of the iconic features of Canadian English is the raising of the diphthongs [aj] and [aw] to [ʌj] and [ʌw] before voiceless obstruents [1]. This process, known as Canadian raising (CR), is phonological and may occur regardless of speech rate [2], and it has been claimed that it arises from the shortening effect these consonants have on preceding vowels [3]. It is currently unknown to what extent CR occurs in singing, in which large differences in note duration make it impossible to correct for speech rate. This question is particularly interesting for professional singers, who are trained to modify their vowels. Trained singers have been shown to have a significantly reduced vowel space [4] and to maintain a more open jaw posture [5], which may interfere with raising. In this paper, we investigate to what extent CR exists in the singing of trained Canadian singers, and whether and how it is different from their raising in normal speech. We report on an experiment in which Canadian singers are asked to say and sing passages containing multiple tokens of raising vowels followed by voiceless and voiced consonants. Tokens are embedded in novel lyrics written to fit commonly known melodies. We test the degree of raising in singing compared to speaking as well as within each of these production modes. Results contribute to our understanding of the acoustics of singing and how singer dialect interacts with the acoustic realization of the articulatory settings of trained singing.

2 Methods

To test CR in singing, we wrote new lyrics to two familiar melodies, in order to include a large number of raising diphthongs ([aj] and [aw]) in raising contexts (followed by voiceless obstruents), as well as in non-raising contexts (followed by voiced obstruents) for comparison. In the lyrics, we had 31 tokens of [aj] followed by a voiceless obstruent, 16 of [aj] followed by a voiced obstruent, 15 of [aw] followed by a voiceless obstruent, and 8 of [aw] followed by a voiced obstruent.

We recruited 3 professionally trained singers (1F/2M) who are native speakers of Canadian English. They were seated in a quiet room, in front of a Macintosh desktop computer and Yamaha YST-M40 speakers, and they wore a C520 headset microphone plugged into an M-Audio Fast Track Pro pre-amp. Sound was recorded into Audacity.

The experiment was presented using PowerPoint. The two songs were presented one at a time, first for speaking, then listening to the music, then singing.

In the speaking section, the lyrics were written out on a slide, with one verse per slide. Instructions told participants to say the text aloud three times before moving to the next slide. Following all of the verses, the music slide came up, with embedded sound that participants could click to listen to the music. They were permitted to do so as many times as they wished. When ready, they proceeded to the singing slides. On each slide, the lyrics to one verse were given, together with an embedded sound icon for the music for that verse. They could listen to the music as many times as they wished, and were instructed to sing each verse three times before moving on to the next slide. Note that because the headset microphone was directional, the music playing from the speakers could not be heard in the recording. Following the completion of the first song, the same procedure applied to the second. After both songs, participants completed a short background questionnaire.

3 Results

All diphthongs of interest were segmented manually in Praat [6], and a script was used to extract F1 and F2 values at the 25% point. These values were then used to determine subjects' raising.

We excluded the data from two of the participants (both males) for not raising in the spoken sections. The lack of raising was apparent both auditorily (to trained phoneticians who are native speakers of Canadian English) and through the formants, where F1 was slightly though not significantly higher in raising contexts than in non-raising contexts.[1] The remaining discussion focuses on the remaining (female) participant.

For all speaking/singing and raising/non-raising conditions, the difference in mean F1 between the two vowels [aj] and [aw] (at the 25% point that we measured) were quite small (the largest was less than 40Hz). We concluded that the initial vowel of both diphthongs is essentially equivalent in height for this particular speaker and consequently combined the vowels [aj] and [aw] in further analysis, to get a general picture of this participant's raising. The mean F1 and F2 values for this speaker in each of the contexts and conditions are given in Table 1.

Table 1: Formant values in each condition

Mode of Speech	Raising Context?	F1 (Hz)	F2 (Hz)
Spoken	Yes	566.03	1323.67
Spoken	No	620.44	1306.28
Sung	Yes	518.32	1272.11
Sung	No	563.65	1296.50

* mhschellenberg@gmail.com
† aozburn@gmail.com

As is evident, raising is primarily seen in F1, with F1 values lower for the raising contexts than the non-raising contexts. This is true for both the spoken condition and the sung condition. Further, the F1 values for singing are lower than those of speech for both raising and non-raising contexts.

Despite having only one speaker, we ran a linear model to determine whether any of these effects neared significance, and to look at whether singing and speech have similar degrees of raising. The dependent variable was F1, and the independent variables were mode of speech (sung versus spoken), following consonant voicing (raising versus non-raising context), and their interaction. The reference condition was the spoken mode in the non-raising context.

The factor of raising neared significance ($p = 0.0623$), as did the factor of mode of speech ($p = 0.0918$). We expect that these effects would reach significance with more speakers. In both cases, the estimates and t-values were negative, indicating that raising and singing both (independently) lower F1. The interaction effect did not near significance ($p = 0.8265$). The estimate and t-value for the interaction was positive, indicating a very slight, far-from-significant trend for less raising in singing than in speech.

4 Discussion and Conclusions

The results show substantial F1 differences between raising and non-raising contexts in the singing condition (518.32Hz vs. 563.65Hz), suggesting that this singer maintains CR during singing. This singer has 25 years of training and experience singing and has sung professionally for seven years. Her maintenance of CR in spite of her training speaks to the robustness of this pattern even in singing. In particular, training for singers includes training on English diction. LaBouff [7] is one of the standard texts for this type of training. She instructs singers to sing in "neutral English", which she defines (p. 6) as one of three accents: American Standard, RP, or Mid-Atlantic. (The choice depends on repertoire, with British music in RP, North American music in GA, and anything else in Mid-Atlantic.) The singer reported training of this sort, with a strong emphasis on singing English with a Mid-Atlantic dialect. However, despite this training, CR persists in her singing

Further, raising is maintained to an approximately equivalent degree in singing as in speaking. However, there is a slight (non-significant) decrease in raising magnitude, with the F1 mean in raising sung contexts about 9 Hz higher than we would expect from a simple combination of the F1 decrease from raising alone combined with that from singing alone. We were only able to look at the results of a single singer; further research is needed to investigate how robust this difference is, and whether it becomes significant with additional singers.

For the preliminary analysis, we proceeded with an assumption that [aj] and [aw] start at the same vowel, and we pooled the results across diphthongs. While means suggested this assumption is fairly accurate for this speaker, it may not hold more generally [8], and so future work should look at the diphthongs separately.

This preliminary study did not consider duration, which may also contribute to the presence of CR in singing. Diphthongs set to longer notes are traditionally sung on the initial vowel of the diphthong in Western classical music, which may mean that longer notes show a lesser degree of CR. Preliminary auditory impressions suggest that longer diphthongs may, in fact, have less CR than those set to shorter notes. However, we do not presently have enough data to confirm this impression.

There was an overall lowering of F1 in singing compared to speaking, which neared significance even for the single speaker. As a result, the F1 values for the singer's unraised sung vowels are surprisingly similar to those of her raised spoken vowel values. This F1 lowering may well be a result of the smaller vowel space that results from Western classical singing technique [4]: the singer may be lowering her whole vowel space to compensate for the lowering of higher vowels that is common in singing.

Overall, this study has shown that CR is maintained in sung speech by at least one professional singer who is a native speaker of Canadian English. The results are suggestive of the persistence of singer dialect despite training, and raise a number of possibilities for future research into this phenomenon.

Endnotes

[1.] One of the speakers, though born and raised in Canada, had British parents and a slight British accent. We have no explanation for the lack of raising in the other speaker.

Acknowledgments

We would like to acknowledge the generous support of Bryan Gick and the Interdisciplinary Speech Research Laboratory at the University of British Columbia.

References

[1] J.C.Wells *Accents of English* (Vol. 3). Cambridge University Press, 1982.

[2] P. Boersma & J. Pater. Constructing constraints from language data: The case of Canadian English diphthongs. *Handout, NELS*, *38*, 2007.

[3] J. Myers. Canadian raising and the representation of gradient timing relations. *Studies in the Linguistic Sciences,* 27(1), 1997.

[4] W. Ophaug. *Sangfonetikk: en innføring*. Fagbokforlaget, 2010.

[5] A. Nair, G. Nair & G. Reishofer. The Low Mandible Maneuver and Its Resonential Implications for Elite Singers. *Journal of Voice*, *30*(1), 2016.

[6] P. Boersma, and D. Weenink. Praat: doing phonetics by computer [Computer program]. Version 5.4.08, http://www.praat.org/, 2015.

[7] K. LaBouff, *Singing and communicating in English: a singer's guide to English diction*. Oxford University Press. 2007.

[8] R. E. Hagiwara. Vowel production in Winnipeg. *The Canadian Journal of Linguistics/La revue canadienne de linguistique*, *51*(2), 127-141, 2006.

INDIVIDUAL VARIATIONS AND GRADIENCE IN ENGLISH PALATALIZATION

Jae-Hyun Sung *
University of Alberta

1 Introduction

English palatalization is of two types : lexical palatalization being morphologically governed, in which it applies only in derived environments, before /j/-initial suffixes [1], and post-lexical palatalization resulted as a purely phonetic and coarticulatory process. It is also known that palatalization is optional at the post-lexical level [2]. Depending on individual speakers and speech rate, speakers of English may palatalize target consonants in post-lexical contexts to various degrees, or may not palatalize them at all. For this reason, post-lexical palatalization in English is also known to be gradient coarticulation, while lexical palatalization is a categorical phenomenon [2, 3].

The articulatory properties of lexical and post-lexical palatalization have been reported in numerous previous experimental studies [2, 4–6]. Previous EPG-based findings [2, 4] showed that palatalized /s/s from lexical palatalization in American English, e.g., *pressure*, exhibited categorical patterns, whereas palatalized /s/s from post-lexical palatalization, e.g., *press your*, yield gestural variations across speakers. Given that the findings were based on a small sample, it calls for further investigation whether this holds for other palatalized coronal consonants and other speakers of American English. This study investigates the articulatory aspect of palatalization using ultrasound imaging to verify the previously reported findings in relation to the phonological source of palatalization, i.e., lexical or post-lexical palatalization.

2 Method

2.1 Participants

Twelve native speakers of American English (6 females) from the University of Arizona participated in the production experiment. Age ranges from 20 to 23 with the exception of Speaker 12, who was 40 years old at the time of data collection.

2.2 Stimuli

Forty four English words and phrases were used in the experiment : 32 test items and 12 fillers. Table 1 shows the subset of items used in the experiment.

Table 1: Examples of the stimuli

segment	type	item
/t/	lexical	'spiritual'
	post-lexical	'last year'
/d/	lexical	'procedure'
	post-lexical	'would you'

2.3 Procedure

All the experiment sessions were conducted at the University of Arizona, using a SonoSite TITAN portable ultrasound unit and a C-11/7-4 11-mm broadband curved array transducer. The machine generates 30 frames per second, equivalent to approximately 15-20 ultrasound tongue images for a disyllabic word. The ultrasound images were concurrently recorded with audio, and the visual and audio data were synchronized. During the experiment session, with the ultrasound transducer held in position, speakers were allowed to move their heads freely without any head stabilization device. After the data was collected, raw tongue contours were adjusted based on the palate contour obtained at the beginning of each experiment session.

2.4 Analysis

Image frames corresponding to the palate contours and the test items were identified and extracted based on the corresponding acoustic signals using UltraPraat [7]. Once the image frames of interest were identified, the tongue curves shown in the frames were manually labelled. The data points were statistically analyzed using Smoothing Spline ANOVA (henceforth SSANOVA) [8]. As shown in Figure 1, the sets of tongue contours were considered to be significantly different when the confidence intervals do not overlap.

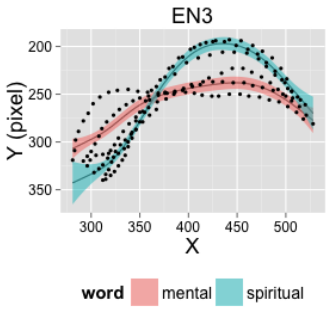


Figure 1: SSANOVA plot of *mental* (plain /t/) and *spiritual* (palatalized /t/) from Speaker 3 (EN3) : Tongue tip is to the right, and shades represent 95% confidence interval. Dots represent the data points.

3 Results

3.1 Plain vs. palatalized

The results are summarized in Table 2. The articulatory gestures from this study show that speakers do articulate the contrast between plain and palatalized coronal obstruents. While the majority of speakers made significantly distinct tongue curves for two conditions, however, no two speakers shared the same exact gestural pattern. For instance, as in Figure 2, Speaker 10 (EN10, female, age 22) only makes the articulatory distinction in the front region of the tongue, whereas Speaker 11 (EN11, male, age 20) makes the articulatory distinction in both tongue tip and back regions. The gestural patterns reported here provide articulatory evidence for inter-speaker variability in palatalization.

*. sung@ualberta.ca

Table 2: A summary of the results : numbers represent how many among 12 speakers showed significantly different tongue gestures.

segment	comparison	results
/t/	plain vs. palatalized	11/12
	lexical vs. post-lexical	9/12
/d/	plain vs. palatalized	12/12
	lexical vs. post-lexical	10/12
/s/	plain vs. palatalized	9/12
	lexical vs. post-lexical	8/12
/z/	plain vs. palatalized	10/12
	lexical vs. post-lexical	6/12

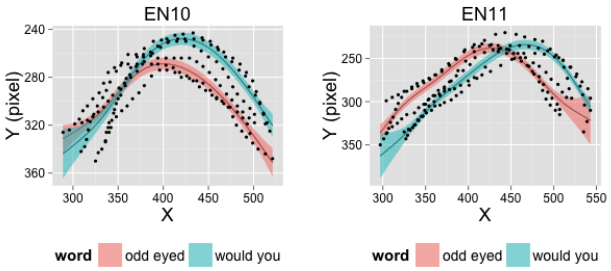


Figure 2: SSANOVA plots of /d/ from *odd eyed* (plain /d/) and *would you* (palatalized /d/) : Speaker 10 (EN10) and Speaker 11 (EN11)

3.2 Lexical vs. post-lexical palatalization

Compared to plain vs. palatalized consonants, the articulatory contrast between lexical and post-lexical palatalization was not always robust (see the results in Table 2). However, the gestural patterns for two different types of palatalization also showed a clear sign of inter-speaker variability, as shown in Figure 3. Speaker 8 (EN8, male, age 20) made a higher tongue dorsum in the post-lexical condition, whereas Speaker 9 (EN9, male, age 21) did so in the lexical condition.

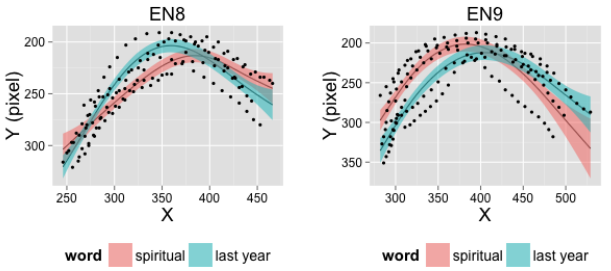


Figure 3: SSANOVA plots of /t/ from *spiritual* (lexical) and *last year* (post-lexical) : Speaker 8 (EN8) and Speaker 9 (EN9)

The articulatory data from 12 speakers confirms that the distinction between plain and palatalized coronal consonants manifests as different tongue gestures, and the way speakers distinguish them is not uniform across speakers. Furthermore, speakers may or may not differentiate palatalized consonants arising from two palatalization processes, also resulting in a substantial amount of individual variations.

4 Discussion

This study shows that speakers of English are aware of various palatalization-inducing contexts, and do articulate contrast among palatalized consonants from different phonological sources. The gestural patterns in this study offer further insight into lexical and post-lexical palatalization in English, in that both types of palatalization exhibit gradient behavior to some extent.

The individual variations in English palatalization suggest that speakers may not have a systematic means of producing palatalization or differing palatalized consonants from lexical and post-lexical processes. Another implication from the idiosyncratic patterns in this study is that speakers may internalize "individualized" palatalization grammar and articulate it. Given that similar individual variations have also been found in other languages [9] and among L2 speakers of English [10], the gestural variations for palatalization merit further cross-linguistic examination.

5 Conclusions

The gestural patterns from this study provide articulatory evidence for different types of palatalization in English and offer new insight into individual variations and gradience. The findings here also add weight to the growing literature on speaker-specific variability in speech production, and call for further cross-linguistic investigation of palatalization.

Acknowledgments

I gratefully acknowledge support from the University of Arizona Social & Behavioral Sciences Research Institute dissertation grant. Special thanks go to Diana Archangeli, Mike Hammond and Brad Story for their helpful comments and technical help on this work. All the remaining errors are my own.

References

[1] Noam Chomsky and Morris Halle. *The Sound Patterns of English*. MIT Press, 1968.

[2] Elizabeth C. Zsiga. An acoustic and electropalatographic study of lexical and postlexical palatalization in American English. In Bruce Connell and Amalia Arvaniti, editors, *Phonology and Phonetic Evidence : Papers in Laboratory Phonology IV*, pages 282–302. Cambridge University Press, 1995.

[3] James Myers. Lexical phonology and the lexicon. National Chung Cheng University, ms, 1999.

[4] Elizabeth C. Zsiga. Phonetic alignment constraints : consonant overlap and palatalization in English and Russian. *Journal of Phonetics*, 28 :69–102, 2000.

[5] Nathan Bush. Frequency effects and word-boundary palatalization in English. In Joan L. Bybee and Paul Hopper, editors, *Frequency and the Emergence of Linguistic Structure*, pages 255–280. John Benjamins, 2001.

[6] Gwanhi Yun. *The interaction between palatalization and coarticulation in Korean and English*. PhD thesis, The University of Arizona, 2006.

[7] Diana Archangeli, Jeff Berry, Paul Boersma, David Ellison, and Gus Hahn-Powell. UltraPraat : Software & database for simultaneous acoustics and articulatory analysis. In *UltraFest VI*, 2013.

[8] Lisa Davidson. Comparing tongue shapes from ultrasound imaging using smoothing spline analysis of variance. *Journal of Acoustical Society of America*, 120 :407–415, 2006.

[9] Jae-Hyun Sung. *A cross-linguistic articulatory analysis of palatalization in Korean, English, and Scottish Gaelic*. PhD thesis, The University of Arizona, 2015.

[10] Gwanhi Yun. Lexical and phonological effects on phonological variation in L2 English palatalization. *Studies in Phonetics, Phonology, and Morphology*, 18 :297–320, 2012.

JAPANESE EFL LEARNERS' PRODUCTION OF PRONOUNS AND ARTICLES IN ENGLISH: EVIDENCE FOR L2 PROSODIC STRUCTURES

Noriko Yamane *[1], Noriko Yoshimura †[2] and Atsushi Fujimori ‡[3]
[1] Kobe University. [1,2] University of Shizuoka. [3] Shizuoka University, Japan

1 Introduction

Experimental studies on Japanese speakers of L2 English (JEL2) have revealed that L1 prosodic transfer occurs in the production of lexical accent, duration, and temporal organization across syllables [4], syllable structures and isochrony [1]. The entire intonation of L2 English may also be influenced by L1 syntax, such as word order [2] and phrasing [3]. But it remains unclear how items unavailable in the L1 are realized in L2 prosody. The current study investigates how morphosyntactic components unavailable in L1 Japanese are prosodically realized in L2 English.

2 Method

2.1 Procedure

Twenty Japanese university students who learn English as a foreign language (intermediate level, 18-22 years old, 14 females, 6 males) (JEL2) and 10 speakers of North American English (20-49 years old, 7 females, 3 males) (EL1) participated in the study. Their task was to read a short passage (1) aloud.

(1) *Some years ago, Mr. Sato had a very kind student in his class. She had a pretty name, Aika. Her classmates liked her very much. Sometimes Mr. Sato saw her at school early in the morning. In her hands she always had very pretty flowers. She picked them from her garden. Everyone in her class loved the colorful flowers.*

2.2 Measurements

English personal pronouns and articles ('her', 'them', and 'the') were selected for the present analysis because their equivalents are not available in Japanese. The duration of these items and phrases containing them (DPs and VPs) was measured with Praat. Then the ratios of the target items to the entire phrases were calculated, and compared with those of EL1. The pitch height and the duration of the pauses were also measured if they became relevant. This paper reports the measurement results of some noun phrases ('the colorful flowers' and 'her garden') and a verb phrase (i.e., 'picked them') recorded.

* noriko.yamane@outlook.com
† yoshimun@u-shizuoka-ken.ac.jp
‡ fujimori.atsushi@shizuoka.ac.jp

3 Results

3.1 Duration

It is well known that L1 English has a significant duration difference between stressed and unstressed syllables. The duration of a stressed vowel to the duration of an unstressed vowel of the same word (e.g., "CONtract") produced by L1 English ranges from 1.6 to 2, but the novice JEL2 tend to have the ratio around 1 [4]. Since pronouns and articles are unstressed in English, some difference should be expected between EL1 and JEL2, in terms of the duration ratio of these items to the phrase containing stressed syllables.

A significant difference in the ratios was found in the noun phrase [her garden], but not in the verb phrase [picked them]. As expected, JEL2 pronounced the pronoun 'her' significantly longer than EL1 in the noun phrase ($p<.0001$). The mean duration ratio of 'her' was 27.8% in EL1, and 41.2% of the phrase in JEL2.

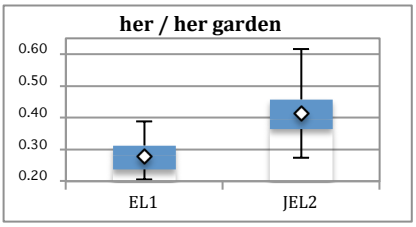


Figure 1: Ratio of 'her' to DP

In the verb phrase, however, the object pronoun 'them' did not show any significant difference in the mean duration ratio between the two groups. (46.4% for EL1, 48.4% for JEL2).

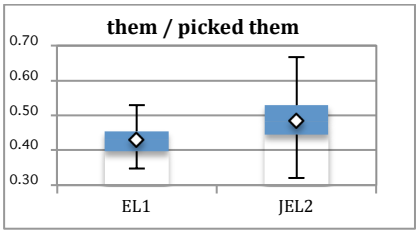


Figure 2: Ratio of 'them' to VP

There was a general tendency for JEL2 to have longer duration in the production of pronouns and articles in DP compared to EL1.

3.2 Pitch

As for the pitch patterns of the target phrases (DP and VP), there was no statistically significant difference between JEL2 and EL1. Both groups showed a great variety of pitch patterns – rising, falling or flat appeared across the speakers,

and it was ungeneralizable. However, the global pitch contours over the sentence level seem somewhat different.

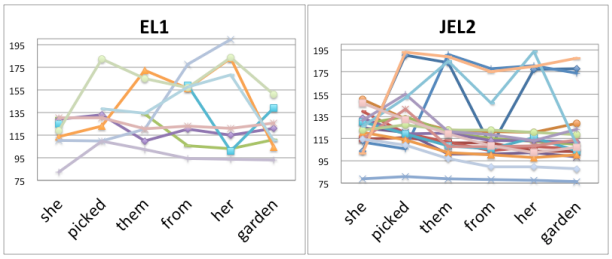


Figure 3: Pitch patterns of the whole sentence

In the sentence *she picked them from her garden*, the subject pronoun 'she' is given whose referent is in the discourse (old information) while the VP is the focus (new information). Then the VP is expected to receive the highest pitch. As seen in Figure 3, the pitch movements vary across EL1, but in the transition from the pronoun 'she' to verb 'picked' (i.e., at the left edge of the VP), pitch rise seems to be a popular direction. As for JEL2, pitch fall in this position seems to be a general tendency, contrary to the expectation. You could also recognize from the graph above that the majority of JEL2 have the highest pitch at the sentence initial position and gradually decline it toward the end of the sentence. This is probably a transfer of Japanese intonation to L2 English [5].

3.3 Pause

Pauses were identified if two ESL instructors perceived them as such. The difference of pausing between EL1 and JEL2 was found most noticeably in the sentence *Everyone in her class loved the colorful flowers.* EL1 typically placed a pause before the determiner (i.e., at the left edge of the DP), but about half of JEL2 paused between 'the' and 'colorful'. The latter pattern was not found in any of EL1. The acoustic measurements were done for the duration of the pauses that those JEL2 placed, and the average was around 74 ms. The duration of the DP *the colorful flowers* that they produced was also measured. For all participants of JEL2, the duration ratio of the pause to the DP utterance was calculated. The results show that the mean ratio of pauses to the DP was 0.03, and there was a significant difference between JEL2 and EL1 ($p<.01$).

4 Discussion

It is interesting to see that there is an initial F0 rise at the sentence-initial position, because the neutral intonation contour of Japanese sentences is similar to the orthography □ showing an initial rise and downsteps [3]. Such a pattern was found in the JEL2's production, suggesing that it is a prosodic transfer at the global intonation level.

Now one question arises: why were pronouns and articles unavailable in Japanese pronounced longer? English pronouns and articles are, in neutral context, realized as phonologically weak (with schwa) which makes a sharp contrast with content words which should have greater prominence. Although pronouns and articles do not exist in Japanese, pronouns like *kare* 'he' or *kanojo* 'she' (direct translations from English) can appear only when they function as referential nouns and have semantic focus in the context. Thus, JEL2 could have unconsciously mapped the L1 focus prominence onto the prosodic prominence of L2 English.

Another question concerns why a pause appears between 'the' and 'colorful' in the JEL2's production. This could be due to a processing problem. Generally speaking, the longer the DP, the longer the time it takes to process it. The JEL2 might have put a pause after the determiner to process the rest of the phrase.

Finally, as shown in section 3.1, EL1 and EJL2 had no significant difference in the duration ratio of the pronoun to VP. It is possible to think that in EL1, the duration of pronouns could behave asymmetrically between DP and VP. Pronouns of English are usually short, but they could be subject to get lengthened due to the VP-final position. This view is compatible with Complement Law which states that 'in a head and complement pair of words, main stress falls on the complement independently of its location' and the Iambic-Trochaic Law which states that OV languages like Japanese is inclined to have trochaic rhythm, and VO languages like English to have iambic rhythm [2]. Further research is required with augmented and controlled data.

5 Conclusion

It was found that i) English pronouns and articles are realized in a longer proportion to the relevant syntactic phrase in JEL2's production, ii) pitch contours of JEL2 have some trace of Japanese, iii) an unexpected break was found inside a phrase. Further study is needed with controlled sentences in order to find whether there is any difference in pronouns and articles between DP and VP.

Acknowledgments

This research has been partially supported by the Japan Society for the Promotion of Science Grant-in-Aid for Scientific Research (C) 26370700. We also thank students at Kobe University and members of the Interdisciplinary Speech Research Laboratory of the University of British Columbia for participating in the experiment.

References

[1] Kubozono, H. (1989). Syntactic and rhythmic effects on downstep in Japanese. *Phonology*, *6*(01), 39-67.

[2] Nespor, M., Shukla, M., van de Vijver, R., Avesani, C., Schraudolf, H., & Donati, C. (2008). Different phrasal prominence realization in VO and OV languages. *Lingue e Linguaggio*, *7*, 1-28.

[3] Selkirk, E., & Tateishi, K. (1991). Syntax and downstep in Japanese. In *Interdisciplinary approaches to language* (pp. 519-543). Springer Netherlands.

[4] Ueyama, M. (2000). *Prosodic transfer: An acoustic study of L2 English vs. L2 Japanese* (Doctoral dissertation, University of California Los Angeles).

[5] Yamane, N., N. Yoshimura, A. and Fujimori (2016). Prosodic transfer from Japanese to English: Pitch in focus marking, *Phonological Studies*, ed. Phonological Society of Japan. 97-104. Kaitakusha.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Affricate Contrasts In Eastern Oromo

Maida Percival

This study has two goals: investigate how laryngeal contrasts in affricates are realized and provide a new, detailed description of affricates in Eastern Oromo, a Cushitic language of Ethiopia. Oromo has a three-way laryngeal contrast between ejective, aspirated, and voiced stops and affricates. Acoustic properties that have been found to distinguish this type of contrast in stops include duration (e.g., closure duration, voice onset time), burst (e.g., centre of gravity, intensity), and vowel measures (e.g., F0 perturbations, rise time) (Kingston 1985, Wright, Hargus & Davis 2002, Vicenik 2010, inter alia). However, these properties have rarely (some not at all) been examined with similar contrasts in affricates. Despite being one of the languages with the most mother-tongue speakers in Africa, Oromo has also been under-documented due to its lack of official status. Therefore, to investigate, 8 native speakers from the Oromia region (residing in Toronto) were asked to read a word list with post-alveolar geminate and singleton affricates, which were in word-initial and word-medial positions (when possible–there are no word-initial geminates and no aspirated singletons). The data was compared to previously analysed stops produced by the same speakers. Preliminary findings suggest similarities in the cues to the laryngeal contrast between stops and affricates in Oromo, but also some differences in the degree to which cues to the three-way laryngeal contrast manifest. For instance, F0 perturbations were not as clear for affricates as for stops, and additional intensity and spectral cues within the frication portion were present for the affricates. The study will discuss these findings with emphasis on articulatory differences between the two manners of sounds, and suggest that while spectral measures are generally referenced as being of importance for place of articulation differences, they may also play a role in distinguishing laryngeal contrasts.

SPEECH TESTING - TESTS DE LA PAROLE

An Ecological Approach To Assess Auditory Perception.
François Bergeron, Aurore Berland, Elizabeth Fitzpatrick, Christophe Vincent, Annie Giasson, Danielle Cloutier, Genevieve Tremblay, Mathieu Hotton, Kevin Leung Kam, Jean-Luc Puel, Jean-Charles Ceccato, Bastien Bouchard 216

Discrimination Of Aspiration Noise In Breathy Vowels
Ilse Labuschagne, Valter Ciocca 218

A Software Platform To Administer The Canadian Digit Triplet Test
Nicolas Ellaham, Christian Giguère, Josée Lagacé, M. Kathleen Pichora-Fuller 220

Intelligibility Of Older Talkers On The Nu6 Test
Huiwen Goy, M. Kathleen Pichora-Fuller, Pascal Van Lieshout 222

A Comprehensive Approach For Assessing The Effectiveness Of Frequency-Lowering Hearing Aids And Electric Acoustic Stimulation (Eas) Cochlear Implant For Treating People With A Severe-To-Profound High-Frequency Hearing Loss.
Mathieu Hotton, François Bergeron 224

Listening To Speech In Noise: Evaluation And Training
Benoît Jutras, Mojgan Owliaey, Lyne Lafontaine, Alexis Pinsonnault-Skvarenina, Jean-Pierre Gagné 226

Intelligibility, Response Time And Listening Efficiency Of Native And Non-Native Listeners In Various Acoustical Conditions
Alice V Lam, Murray Hodgson 228

Speech Auditory Brainstem Responses And Words-In-Noise Recognition Scores: Is There A Link?
Josée Lagacé, Amineh Koravand, Jordon Thompson, Suzanne Lteif 230

What Do Forced Alignment Likelihood Scores Tell Us About The Aligned Speech?
Ayushi Mrigen, Daniel Brenner, Benjamin V. Tucker 232

Effects Of Modality And Linguistic Materials On Memory
Jenna Pattison, M. Kathleen Pichora-Fuller, Sherri L. Smith 234

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 236

AN ECOLOGICAL APPROACH TO ASSESS AUDITORY PERCEPTION

François Bergeron*[1], Aurore Berland[ϕ2], Elizabeth Fitzpatrick[γ3], Christophe Vincent[η4], Annie Giasson[ι5], Danielle Cloutier[φ6], Geneviève Tremblay[κ6], Mathieu Hotton[λ1], Kevin Leung Kam[μ7], Jean-Luc Puel[ν7], Jean-Charles Ceccato[ο7], Bastien Bouchard[π8]

[1]Université Laval, Québec, Canada.
[2]Université de Toulouse II, Toulouse, France
[3]Université d'Ottawa, Ontario, Canada
[4]CHRU de Lille, Lille, France
[5]Hôpital Régional Chaleur Regional Hospital, Nouveau Brunswick, Canada
[6]CIUSSS de la Capitale-Nationale-Installation IRDPQ, Québec, Canada
[7]Université de Montpellier 1, Montpellier, France
[8]Technologies Immersion Inc, Québec, Canada

1 Introduction

Improvement in auditory perception is a major objective of therapeutic interventions offered to hearing impaired children or adults. Numerous tests are proposed to guide these interventions and assess benefits. Optimally, these tests should explore complex abilities such as those encountered by hearing impaired people in daily life.

In fact, available tests are mostly designed to be administered in a controlled environment (such as a sound booth), in quiet or in a standardized speech spectrum noise issued from fixed positions, often with arbitrarily predetermined signal to noise ratios. For example, the Hearing In Noise Test is based on the repetition of sentences against a 65 dBA speech spectrum noise coming from the front, right or left side. The test has been standardized as an adaptive procedure to determine a speech reception threshold (SRT); however, in clinics and in research, it is frequently administered with predetermined signal to noise ratios (+10, +5, 0 dB). While sentences appear as a realistic daily stimulus, one can argue that a speech spectrum noise issued from fixed positions with variable or arbitrarily predetermined signal to noise ratios can be far from real life.

Yet, it is possible nowadays to recreate realistic soundscapes through multichannel audiovisual systems. This project aims to explore this avenue. Specifically, the objectives are to :

1. design a system that can virtually reproduce everyday sound experiences and offer a more realistic testing condition to assess auditory perception in hearing impaired children and adults, namely localisation abilities and speech perception;

2. specify the psychometrics of the system and define norms from normal-hearing individuals.

* francois.bergeron@fmed.ulaval.ca
ϕ aberland@univ-tlse2.fr
γ elizabeth.fitzpatrick@uottawa.ca
η christophe.vincent@chru-lille.fr
ι Annie.Giasson@vitalitenb.ca
φ Danielle.Cloutier@irdpq.qc.ca
κ Genevieve.Tremblay@irdpq.qc.ca
λ mathieu.hotton.1@ulaval.ca
μ leungkam.kevin@gmail.com
ν jean-luc.puel@inserm.fr
ο jean-harles.ceccato@umontpellier.fr
π bbouchard@immersion360.ca

2 Method

The Immersion 360 system has emerged from a pilot project on a prototype system where the feasibility and the relevance of creating virtual environments to assess auditory perception has been confirmed [1]. At this time, experiments showed that altered capacities for speech perception could be observed in a virtual environment despite an excellent performance in a standard clinical setting at a similar signal to noise ratio; more, changing the environment affected the performance, even when signal to noise ratios were kept constant. Therefore, the usual clinical condition did not seem to correctly replicate real life functioning, which is characterized by diverse soundscapes where conversation to noise ratios are in the +3 to +5 dB range in order to maintain intelligibility [2].

From this ground, the Immersion 360 system was developed to virtually reproduce common daily sound experiences in terms of noise type, incoming directions and signal to noise ratios in order to recreate a realistic testing condition to assess auditory perception of hearing impaired persons. As this system can also emulate discrete source positions, localisation capacities can also be assessed with the same equipment. The actualised version of the Immersion 360 system consists of 8 point-source speakers and 8 virtual speakers forming altogether 16 sound sources. These sources are located at equal distance on a 360^0 horizontal arc. The implementation of virtual sources increases the resolution of the system despite a limited number of actual sources, allowing more precise measures of angle errors for auditory localisation in a system based on a limited number of point-source speakers. A meticulous calibration of each source, physical and virtual, ensures an homogeneous loudness across the speaker array. Following suggestions from experienced clinical audiologists, nine common virtual environments were incorporated in the Immersion 360 system to assess speech perception in daily settings (car, garage, cafeteria, restaurant, ball game in a gymnasium, race training in a gymnasium, kindergarten, road traffic, street traffic). These environments were recorded live on locations and reproduced horizontally on 360 degrees. Any other environment needed for clinical or research use can be recorded and integrated in the system.

These two features, that is virtual speakers and integrated environments, distinguishes the Immersion 360 system from similar systems based on an array of speakers [3].

Participants

Eighty normal-hearing adults (18-30 years) were recruited in Canada and in France to assess the psychometrics of the system and define norms for normal-hearing individuals. Hearing thresholds had to be better than 20 dB HL from 250 to 8000 Hz. External and middle ears had to be free of any pathology. All participants were native French speakers.

Procedures

Sentence recognition was assessed with the French-Canadian version of the Hearing in Noise Test or the French adaptation of AzBio [4] in nine virtual environments (car, garage, cafeteria, restaurant, ball game in a gymnasium, race training in a gymnasium, kindergarten, road traffic, and street traffic). These specific environments were selected from a survey completed by clinical audiologists on the most challenging environments in real life reported by hearing impaired people. Speech and noise levels were set at the actual levels collected with a sound level meter on the recording sites. HINT/AzBio lists of 20 sentences were administered in each environment; the score was determined by the percentage of correctly identified words. Presentation order of environment and HINT/AzBio lists were randomly chosen. In order to calculate the test-retest reliability, all tests were repeated one month later with half of the cohort.

3 Results

Averages, standard deviations, and 95% confidence intervals were computed for each test of the Immersion 360 system. These metrics were used to define the normative values for the tests. An analysis of fidelity was used to verify the test-retest reliability for each test of the system and to determine the standard error.

The averages and standard deviations of the speech recognition score in all nine environments were analysed on the perspective of a progressive number of participants, that is 10, 20, 30, 40 and 50 participants. This analysis shows that the variability of these psychometrics remains under the standard error in all tests when 40 to 50 participants are included, and in 91% of the tests between 30 and 50 participants, but fall to 76% of the tests when considering 20 and 10 participants. Stabilization of performance is therefore observed from 30 participants. This confirms that the sample used in the study is sufficient to establish a normative reference.

Table 1 shows that the average performance for speech perception in each environment of the system generally lies between 98 and 100% with a very low standard deviation; the normative lower limit is over 94% (race in a gymnasium). Test-retest showed no significant difference in performance; the maximum standard error is 1%.

Table 1. Psychometrics for the Immersion 360 system

a) AzBio

	Quiet	Car	Street traffic	Garage	Gymnasium Race	Gymnasium Ball	Kinder-garten	Road traffic	Cafeteria	Restaurant
Mean	99,85	99,71	99,71	99,94	95,72	99,06	99,17	98,87	98,43	99,21
SD	0,20	1,05	1,21	1,04	2,83	1,76	0,74	0,91	0,99	0,88
95% interval	99,74	99,16	99,07	99,40	94,23	97,19	99,75	98,48	97,91	98,86
	99,95	99,26	99,34	99,49	97,26	98,97	99,56	99,35	98,94	99,56
p test-retest	0,91	0,32	0,30	0,35	0,43	0,07	0,62	0,73	0,79	0,81
SE	0,11	0,42	0,55	0,74	1,02	0,57	0,41	0,28	0,41	0,49

b) HINT

	Quiet	Car	Street traffic	Garage	Gymnasium Race	Gymnasium Ball	Kinder-garten	Road traffic	Cafeteria	Restaurant
Mean	100,00	99,66	99,81	99,38	99,51	99,88	99,45	99,95	99,76	99,76
SD	0,00	0,78	0,52	1,04	0,72	1,10	0,91	0,14	0,62	0,59
		99,44	99,66	99,01	99,31	99,67	99,29	99,94	99,59	99,59
95% interval		99,87	99,95	99,59	99,70	99,28	99,79	100,00	99,92	99,91
p test-retest		0,45	0,42	0,73	0,50	0,29	0,82	0,33	0,60	0,16
SE		0,25	0,14	0,33	0,17	0,29	0,33	0,04	0,18	0,16

4 Discussion

This study confirms the feasibility of designing a system that can virtually reproduce everyday soundscapes and thus offer realistic testing conditions to assess auditory capacities, such as localisation and speech perception. When applied to a large cohort of normal hearing participants, the Immersion 360 system rendered psychometrics for sentence recognition in the expected range for this population, that is near 100%. This psychometrics can be used as a normative reference for these tests.

The near-perfect performance observed in speech perception in all environments could be anticipated. Indeed, as reflected by the signal-to-noise ratios measured during the recording sessions of each environment, communication partners are always trying to obtain an optimal speech intelligibility regardless of the environmental noise by keeping a positive signal-to-noise ratio. This observation questions the relevance of assessing speech recognition at very low signal-to-noise ratios (0 dB and less) such as frequently done in audiological clinics and in research.

Acknowledgments

This project was partly supported by grants from the OOAQ – REPAR Partnership Program and from Oticon Medical.

References

[1] Bergeron F, Fecteau S, Cloutier D, Hotton M, Bouchard B. Assessment of speech perception in a virtual soundscape. Adult Hearing Screening conference, Como Lake, June 2012.

[2] Smeds K, Wolters F, Rung M. Estimation of Signal-to-Noise Ratios in Realistic Sound Scenarios. Journal of the American Academy of Audiology, Volume 26, Number 2, February 2015, pp. 183-196(14).

[3] Revit LJ, Killion MC, Compton-Conley CL. Developing and testing a laboratory sound system that yields accurate real-world results. Hear Rev. 2007;14:54–62.

[4] Bergeron F, Berland A, Fitzpatrick E, Vincent C, Giasson A, Chafiq W, Fanouillere T. Development and normalisation of a French version of the AzBio sentence test. 14th International Conference on Cochlear Implants and Other Implantable Technologies, Toronto, mai 2016.

DISCRIMINATION OF ASPIRATION NOISE IN BREATHY VOWELS

Ilse Labuschagne [*1] and Valter Ciocca [†1]
[1]School of Audiology and Speech Sciences, The University of British Columbia.

1 Introduction

The presence of aspiration noise (AH) in vowel utterances is one of the key acoustic features that characterizes breathy speech [1, 2]. AH difference limens (DLs) for tokens of the vowel /a/ were measured by Shrivastav and Sapienza [3], and by Kreiman and Gerratt [2], using same-different (SD) and matching tasks, respectively. AH DLs were found to be between 11 and 21 dB in these studies. Shrivastav and Sapienza [3] noticed that DLs depended on the amount of AH present in the standard vowel stimulus: DLs decreased as the amount of AH of the standard stimulus increased. Listeners also exhibited different AH DLs for different stimuli. Since these studies co-varied the amount of AH with changes in other glottal parameters, (e.g., open quotient, spectral tilt, flutter), it is difficult to evaluate the extent to which these changes contributed to the observed AH DLs. The present study investigated DLs for AH by keeping other glottal parameters constant.

There is evidence that changes in the amount of aspiration noise may be more difficult to detect in stimuli with larger amounts of high frequency harmonic energy [3, 4]. Therefore, listeners' sensitivity to the amount of AH in vowels could be affected by spectral differences. In order to test this hypothesis, the present study compared AH DLs for two vowels (/æ/ and /i/).

2 Method

2.1 Stimuli

The implementation of a parallel Klatt synthesizer [5] in the Praat software [6] was used to synthesize six-formant utterances of /æ/ and /i/. These stimuli were modelled after the vowels produced by three male speakers (S08, S30, S44) selected from the Hillenbrand vowel database [7]. The database contains recordings in an h-vowel-d format. The vowel portions of these six utterances (two vowels by three speakers) were synthesized at a sampling rate of 44.1 kHz. This sampling rate ensured that high frequency noise components were present in the stimuli, unlike previous studies which limited the frequency range of their stimuli to 5.5 kHz. Vowels were synthesized to resemble natural utterances by using formant and fundamental frequency estimates at eight time points within each vowel. The amplitude of voicing was set to reproduce the intensity contour of the original excised vowels, and ranged from 67 to 73 dB (average = 70 dB). Glottal waveform parameters for all stimuli included an open quotient of 0.4; all other glottal parameters were set to default values. AH was set to 35 dB for the standard stimuli, with two test conditions differing in AH from the standard stimuli by 2 dB or 4 dB (AH = 37 or 39 dB). Vowel duration ranged from 276 to 302 ms (average = 290 ms).

2.2 Participants

Eight naïve listeners between the ages of 21 and 27 years were screened to ensure that their hearing thresholds were within normal limits (15 dB HL at octave frequencies from 250 to 4000 Hz). Listeners were remunerated for their participation.

2.3 Procedure

First, listeners completed a short training session in order to familiarize themselves with the type of stimuli used in the study. In a subsequent screening session, listeners completed a two-alternative forced choice (2AFC) task in which 16 stimulus pairs with large AH differences (AH = 0 dB for the standard stimulus; AH = 33 dB or 38 dB for the comparison stimuli) were presented. Stimuli were presented binaurally through headphones at a level of 70 dBA in a soundproof booth. Within each trial, listeners heard a sequence of two vowels with different levels of AH. Listeners were asked to select which of the two stimuli sounded "breathier". Feedback was provided after each response. A minimum test score of 12 out of 16 correct identifications was required for a listener to be included in the main experiment. All participants reached this criterion.

The experiment consisted of two sessions, blocked by vowel quality (first /æ/, and then /i/ stimuli). The procedure was similar to that of the familiarization task, except that the two intervals consisted of a standard stimulus with a 35-dB AH level, and one of two comparison stimuli (ΔAH either +2 dB or +4 dB relative to the standard). These differences were chosen on the basis of the results of a pilot study. During each of the two sessions, listeners were presented with 120 trial pairs (3 speakers × 2 AH levels × 2 presentation orders × 10 repetitions). The experiment, including the familiarization and screening sessions, lasted about 1 hour.

3 Results

3.1 Perceptual data

Responses were converted into d' scores [8]. As expected, sensitivity was consistently higher for the 4-dB than for the 2-dB ΔAH stimuli. Figure 1 shows the differences in d' score averages across listeners for the two AH levels and the two vowels.

A three-way ANOVA with AH level, vowel quality and speaker as repeated factors was performed on the d' scores. The main effect of AH level showed a significant difference ($F(1, 86) = 20.8$, $p < 0.01$), with an average d' of 0.56 for the 2-dB and 1.28 for the 4-dB conditions. Main effects of

[*] ilse.labuschagne@alumni.ubc.ca
[†] vciocca@mail.ubc.ca

speaker and vowel quality, and interaction effects did not reach significance.

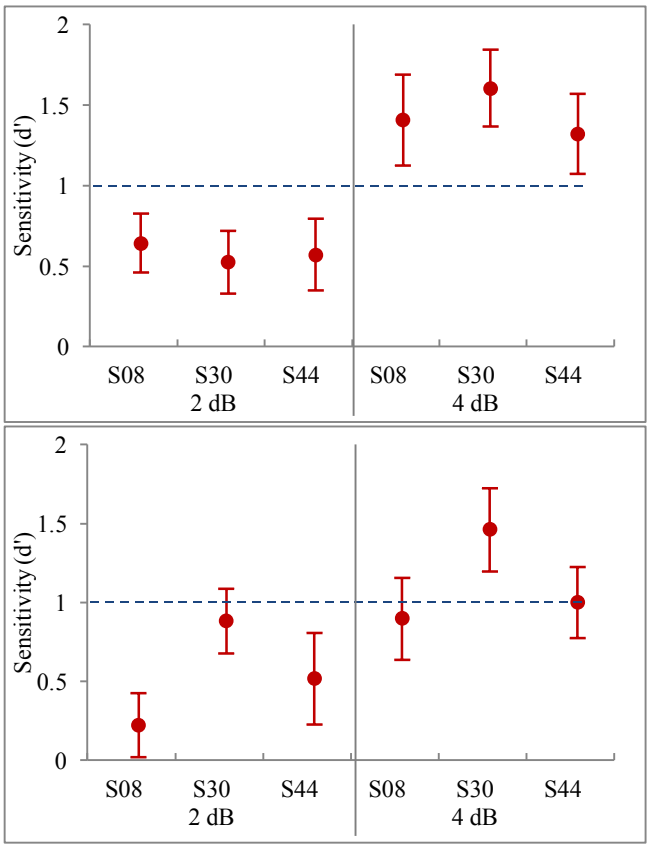


Figure 1: Averages with standard error bars of the sensitivity scores across listeners for the three /æ/ and /i/ vowels -panels (a) and (b), respectively). The empirical threshold level (d' = 1) is indicated with a dotted line.

Although the main effect of vowel quality was not significant, some of the evidence supports the idea that listeners may be more sensitive to AH differences in /æ/ vowels compared to /i/ vowels. For instance, in the 4-dB condition only 4 of the 24 d' scores calculated for the /æ/ vowel (3 vowels × 8 listeners) were smaller than 1, while for the /i/ vowels more than half (14 out of 24) of the d' scores were smaller than 1. While the average d' values for the two vowels in the 2-dB condition were similar (0.58 and 0.54 for /æ/ and /i/, respectively), the difference in the 4-dB condition was larger (1.44 and 1.12 for /æ/ and /i/, respectively). A one-way ANOVA showed that the difference in d' scores between vowels for the 4-dB condition was not significant ($F(1, 46) = 3.02$, $p = 0.089$).

4 Discussion and conclusions

The present results indicate that the AH DLs for the vowels /æ/ and /i/ are between 2 and 4 dB. These values are substantially smaller than those reported in previous studies [2, 3]. Possible reasons for this difference include the differences in the tasks used to measure thresholds, differences in vowel quality, and different definitions of AH DLs. For example, it has been found that an SD task such as the one used by Shrivastav and Sapienza [3] results in poorer performance than a 2AFC task [8, p. 228]. Furthermore, Shrivastav and Sapienza defined the AH DL as the level at which listeners were able to correctly identify a difference in AH in 70.7% of the stimuli. This corresponds to d' = 1 for yes-no tasks, but to d' = 2 in SD tasks. In the current experiment, ΔAH would have to increase above 4 dB to reach a d' value of 2. It is also possible that the higher sampling rate of the stimuli used in the present study, and the omission of low-pass filters in the form of the spectral tilt synthesis parameters, resulted in higher noise energy at higher frequencies; higher noise energy may have resulted in improved discrimination. Differences in vowel quality between the present studies and previous studies may also have contributed to the observed differences in AH DLs. Future studies using the same procedure as the present study with /a/ stimuli will shed light on the contribution of this factor to AH DLs.

The difference in AH DLs between /æ/ and /i/ did not reach statistical significance, probably due to the small sample size (n = 8). It is also possible that the order of presentation of the vowel sessions (/æ/ was always presented first) produced a learning effect that inflated d' scores for the /i/ vowels. Differences in d' scores observed for the two vowels may be due to higher amounts of high frequency formant energy in /i/ compared to /æ/ vowels [3, 4]. Systematic changes to higher formant amplitudes and/or bandwidths, and acoustic analysis measurements may provide insight into the relationship between AH DLs and the acoustic properties of breathy vowels differing in vowel quality.

Acknowledgments

This research was supported by a start-up fund provided to Valter Ciocca by the Faculty of Medicine, University of British Columbia.

References

[1] D. H. Klatt and L. C. Klatt. Analysis, synthesis, and perception of voice quality variations among female and male talkers," *J. Acoust. Soc. Am.*, 87(2), 820–857, 1990.

[2] J. Kreiman, and B. R. Gerratt. Perception of aperiodicity in pathological voice. *J. Acoust. Soc. Am.,* 117(4):2201–2211, 2005.

[3] R. Shrivastav, and C. M. Sapienza, Some difference limens for the perception of breathiness. *J. Acoust. Soc. Am,* 120(1):416-423, 2006.

[4] B. R. Gerrat and J. Kreiman. Measuring vocal quality with speech synthesis. *J. Acoust. Soc. Am.*, 110(5):2560-2566, 2001.

[5] D. H. Klatt. Software for a cascade/parallel formant synthesizer. *J. Acoust. Soc. Am.*, 67(3):971–995, 1980.

[6] Boersma, P., and Weenink, D. Praat: doing phonetics by computer [Computer program]. Version 5.4.19, retrieved 22 September 2015 from http://www.praat.org/, 2015.

[7] J. Hillenbrand, L. A. Getty, M. J. Clark, and K. Wheeler. Acoustic characteristics of American English vowels. *J. Acoust. Soc. Am.,* 97(5):3099-3111, 1995.

[8] N. A. Macmillan, and C. D. Creelman. Detection Theory: A user's guide (2nd ed.). Mahwah, New Jersey: Lawrence Earlbaum Associates, 2005.

A SOFTWARE PLATFORM TO ADMINISTER THE CANADIAN DIGIT TRIPLET TEST

Nicolas Ellaham [*1], Christian Giguère [†1], Josée Lagacé [‡1] and M. Kathleen Pichora-Fuller [§2]
[1]Audiology & Speech-Language Pathology Program, University of Ottawa, 451 Smyth Rd., Ottawa, Ont., Canada K1H 8M5
[2]Department of Psychology, University of Toronto, 3359 Mississauga Rd., Mississauga, Ont., Canada L5L 1C6

1 Introduction

A variety of tests using speech as sound material is available for use in hearing screening, assessment and rehabilitation. Of the materials used for speech tests in audiology, digits have received particular attention as they are highly familiar words and are acquired either at a young age or among the first words when learning a new language. The Digit Triplet Test was originally developed in the Dutch language as an automatic self-screening test [1]. It is a simple test in which the person being tested is presented with a list of digit triplets (*e.g.,* 6-1-8) in a background of noise and is asked to enter the digits heard using a telephone keypad. The test uses an adaptive 1-up-1-down triplet scoring procedure (a correct response is registered only if all three digits are entered correctly) to compute a speech reception threshold (SRT). This SRT corresponds to the signal-to-noise ratio at which the triplets are recognized 50% of the time. The test has been developed in different languages and can now be administered over the phone, the internet or smartphones [2]. Owing to its wide success, and to facilitate comparison between languages, recommendations for constructing such tests have been developed by a working group on multi-lingual speech testing of the International Collegium of Rehabilitative Audiology (ICRA) [3].

This paper describes the testing platform designed to administer the new Canadian Digit Triplet Test (CDTT). First, a summary of the test development is presented. This is followed by a description of the software interface, the supported features, and hardware recommendations. The paper concludes with a brief overview of test applications.

2 Development of the CDTT

Developing the Canadian-English and Canadian-French versions of the test according to the ICRA recommendations involved several steps [3]. First, digit-triplet sequences were recorded in both languages by two fluently bilingual adult talkers (a male and a female). For each language-talker combination, the recordings were segmented to obtain a set of 8 digits at each position in the triplet. Next, a speech-shaped noise signal was generated according to ISO 8253-3:2012 [4] and four test lists were created. The test development includes three testing phases: Phase I (fixed-level, digit scoring) to optimize the digits; Phase II (fixed-level, triplet scoring) to evaluate the test lists; Phase III (adaptive-level, triplet scoring) to establish test norms.

[*] nellaham@uottawa.ca
[†] cgiguere@uottawa.ca
[‡] josee.lagace@uottawa.ca
[§] k.pichora.fuller@utoronto.ca

3 Testing platform

3.1 Overview

The CDTT software is distributed as a cross-platform Java desktop application that can run on any major operating system (*e.g.*, Windows, Mac OS, Linux), with a Java Runtime Environment installed. The main interface provides controls for the test operator to enter participant information, select test parameters, run a test, and save test results (Figure 1). In addition to displaying results at the end of a test, the interface also includes a stimulus-response digits table to update the operator on progress throughout the test as responses are given. A Response Keypad is displayed as the test is running for the test subject to interact with the system (Figure 2). In a test configuration that includes only a single computer monitor, the software can hide the main interface while the test is running to prevent the person being tested from seeing the stimulus digits.

In addition to sound files for the speech test material, the software architecture includes a number of Excel files to store important application settings, list definitions, default test parameters, and calibration values. Test results are also exported into an Excel spreadsheet for each person tested to permit further analysis if needed.

3.2 Features and customizations

The system has been designed as a testing tool both to administer the CDTT in its final form and to carry out the

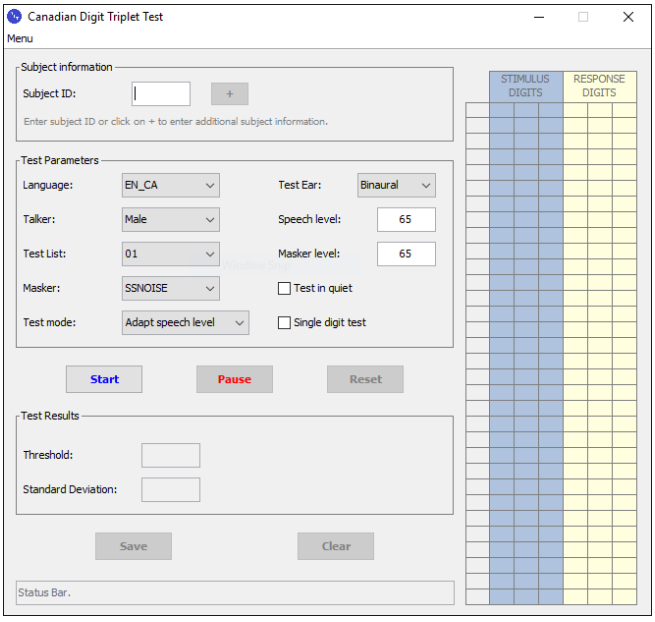


Figure 1: Main interface of the Canadian Digit Triplet Test.

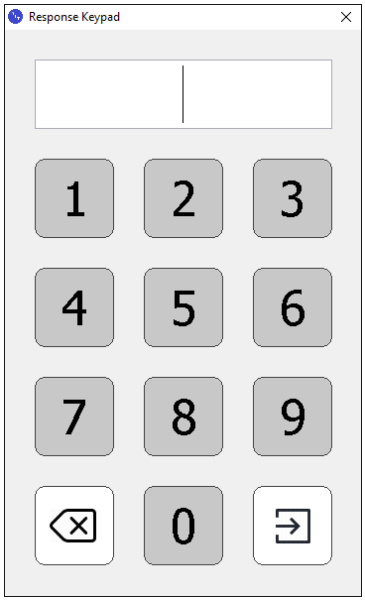


Figure 2: Response Keypad of the Canadian Digit Triplet Test.

different phases of test development. To date, the following features are supported:

- Selection from two test languages and two talkers.
- Selection from four different test lists.
- Testing in monaural or binaural listening conditions.
- Testing in quiet or in speech-shaped noise.
- Testing using a single digit or digit triplets.
- Two test modes: fixed level, or adaptive speech level.
- Triplet and digit scoring (fixed-level mode only).
- Ability to export stimulus-response digits table.
- Input response through software or hardware keypad.
- A calibration interface to calibrate the sound output.
- A detailed settings interface to customize the system.

A wide range of customization options have been built into the application with factory settings defaulting to values consistent with the ICRA recommendations [3]. Moreover, the software implementation has been made flexible in order to permit testing in different use-case scenarios. For example, after the first phase of testing, level adjustment factors are derived to homogenize the intelligibility of all digits at all positions. This requires fitting a psychometric function to achieve a target SRT [5]. Level adjustment factors were derived to align the target SRTs for 50% and 80% intelligibility. Thus, the original recordings used in the first phase are also used in subsequent tests with the level adjustment factors applied in the software at the time of presentation when the digits are resynthesized into triplets.

3.3 Hardware recommendations

The primary result of this speech-in-noise test, the SRT, is not sensitive to absolute sound presentation levels. Thus, it is possible to run the test by connecting a pair of earphones to any computer's audio output. For reliable results, it is recommended to use audiometric earphones, especially if testing in quiet. Moreover, for portability across test sites, it is recommended to use a USB audio interface instead of relying on a computer's internal audio hardware. The calibration values included in the CDTT software have been obtained using pairs of DD45 audiometric earphones and a Creative Sound Blaster X-FI GO! Pro audio interface.

The Response Keypad (Figure 2), allows participants to provide a response using a mouse or touch screen. Its layout corresponds to a phone keypad such that it, like the DTT web interface in different languages, is consistent with versions that are administered over the phone. The number pad on a standard 101-key hardware keyboard or a USB numeric keypad has a different layout. If a hardware keypad is required, the physical keys for 1, 2 and 3 need to be swapped with the keys 7, 8 and 9 in order for the correct keys to be registered. Alternatively, an option is provided in the settings interface to use the physical keypad's layout ("calculator" type). Selecting this option modifies the layout of the software keypad to maintain consistency.

4 Summary and application

The CDTT is presently being developed and evaluated in two languages and using two different talkers, providing a unique set of speech test material. The versatile testing platform presented in this work is designed to administer the test and assist in its development. The test is being deployed in large-scale multi-site studies across Canada on aging: the Canadian Consortium on Neurodegeneration in Aging (http://ccna-ccnv.ca/en/) and the Canadian Longitudinal Study on Aging (https://www.clsa-elcv.ca/). In addition, the test may be a useful tool for lab-based research, audiologic assessment of francophone speakers in minority settings in Canada, studies of English or French as a second language.

Acknowledgments

This work was supported in part by the Natural Sciences and Engineering Research Council (NSERC).

References

[1] C. Smits, T.S. Kapteyn, and T. Houtgast (2004), "Development and validation of an automatic speech-in-noise screening test by telephone," Int. J. Audiology 43(1), 15–28.

[2] M.A. Zokoll, S. Hochmuth, A. Warzybok, K.C. Wagener, M. Buschermöhle, and B. Kollmeier (2013), "Speech-in-Noise tests for multilingual hearing screening and diagnostics," Am. J. Audiology, 22(1), 175–178.

[3] M.A. Akeroyd, S. Arlinger, R.A. Bentler, A. Boothroyd, N. Dillier, W.A. Dreschler, J-P. Gagné, M. Lutman, J. Wouters, L. Wong, and B. Kollmeier (2015), "International Collegium of Rehabilitative Audiology (ICRA) recommendations for the construction of multilingual speech tests," Int. J. Audiology, 54:sup2, 17–22.

[4] ISO 8253-3:2012, "Acoustics - Audiometric test methods - Part 3: Speech audiometry," Geneva: International Organization for Standardization.

[5] S. Jansen, H. Luts, K.C. Wagener, B. Frachet, and J. Wouters (2010), "The French Digit Triplet Test: A hearing screening tool for speech intelligibility in noise," Int. J. Audiology, 49(1), 378–387.

INTELLIGIBILITY OF OLDER TALKERS ON THE NU6 TEST

Huiwen Goy [*1], M. Kathleen Pichora-Fuller [†1,2] and Pascal van Lieshout [‡1,2,3,4]
[1]Department of Psychology, University of Toronto, 3359 Mississauga Rd, Mississauga ON L5L 1C6
[2]Toronto Rehabilitation Institute, 550 University Ave, Toronto ON M5G 2A2
[3]Department of Speech-Language Pathology, University of Toronto, 500 University Ave, Toronto ON M5G 1V7
[4]Rehabilitation Sciences Institute, University of Toronto, 500 University Ave, Toronto ON M5G 1V7

1 Introduction

Most speech intelligibility tests use recordings of younger adult talkers speaking in ideal conditions. However, talker variation is greater in the real world and includes factors such as age and speech changes due to the environment. Older talkers have rarely been recorded, even though it is well known that speech communication in noise is especially challenging for older adults. The first purpose of this study was to investigate the range of intelligibility scores in a sample of older talkers with different perceived speech quality. The second purpose was to determine whether older talkers could increase their intelligibility to a similar extent as younger talkers in past studies, when older talkers were asked to speak in noise or to speak clearly.

2 Method

2.1 Recording of speech materials

Talkers

Talkers were six female and two male adults (mean age = 79.6 years; range 73 to 88). Their hearing thresholds were similar to the 50th percentile for 70-year-olds [1]. Talkers had a variety of speech characteristics and were selected from a previous study in which listeners rated them on their suitability as an audiobook reader [2] (Table 1). Talkers were native Canadian English speakers and were in average to excellent health with no neurological or speech disorders.

Table 1: Perceived audiobook reader quality (Reader Scale of 1 to 5; 5 is best) and acoustic measures of target words in the Normal talking condition.

Talker	Reader	F_0 (Hz)	Duration (ms)
Female 1	1.75	222	744
Female 2	2.25	183	745
Female 3	2.66	180	632
Female 4	3.04	167	610
Female 5	4.19	190	461
Female 6	3.93	208	665
Male 1	3.00	147	623
Male 2	3.75	155	606

Speech material and recording procedure

Talkers recorded sentences from the Northwestern University Auditory Test No. 6 (NU6) [3], which consists of four lists of 50 monosyllabic keywords presented after a standard carrier phrase (e.g., *Say the word boat*).

[*] huiwen.goy@utoronto.ca
[†] k.pichora.fuller@utoronto.ca
[‡] p.vanlieshout@utoronto.ca

There were four recording conditions: speaking "normally, in your most comfortable voice" in quiet (Quiet) and in babble noise (Noise), and speaking "as if to someone with hearing loss" in quiet (HL) and in babble noise (HL+Noise).

Talkers were seated in an IAC sound-attenuating booth with a Sennheiser Linear E825S microphone placed 6 cm from their lips. Speech was recorded using the MS2, PA5 and RP2.1 components of the Tucker-Davis Technologies System III and the Avaaz Time-Frequency Representation program running on a Dell Precision 360 computer. In the noise conditions, multi-talker babble from the WIN test [4] was presented to talkers binaurally through Sennheiser 265 Linear headphones at 78 dB SPL. The same headphones were also worn by talkers in quiet.

Talkers recorded each NU6 list in four talking conditions over two sessions. During each session, talking conditions were always in a fixed order: Quiet, Noise, HL and HL+Noise. The recording order of the four NU6 lists was counterbalanced across talkers and words within each list were randomized. Sentences were re-recorded as needed at the end of the list if there were speech errors or hesitations. Sentences were then spliced out of the raw recordings and RMS-equated to 0.05 Pa using PRAAT [5].

2.2 Listener testing

Listeners

Listeners were 32 younger adults (mean age = 18.4 years, SD = 0.9) who were students enrolled in an introductory psychology course. Listeners had learned English in North America by the age of five years and they had average to excellent general health with no known neurological or speech disorders. They had normal hearing thresholds ($\leq$ 20 dB HL from 250 to 8000 Hz), with no inter-aural differences greater than 15 dB, except for one participant who had an inter-aural difference of 20 dB at 4000 Hz.

Experimental design and test procedure

Each of the four NU6 lists was split in half to create eight half-lists of 25 words each. Every listener heard eight half-lists, each spoken by a different talker. Within those eight half-lists, every listener was exposed to two sets of all four talking conditions (though not every talker was paired with every talking condition for a given listener). The order of the four talking conditions and the order of the eight talkers were rotated through the set of 32 listeners.

Listeners were seated in an IAC sound-attenuating booth. Stimuli were presented to the right ear through Sennheiser 265 Linear headphones using a Dell Precision 360 desktop computer and the RP2.1, PA5, SM5 and HB7 components of the Tucker-Davis Technologies System III.

Babble noise from the WIN test [4] was presented continuously at 70 dB SPL, while speech was presented at 68 dB SPL with a 3-sec interval between sentences. Listeners reported the final word of every sentence and were asked to guess if they were unsure. The experimenter scored responses as they were made and also taped them to enable later confirmation of the scoring.

Data analysis

Percent correct scores were converted into RAU scores for statistical analyses [6]. A linear mixed-effects model was constructed for the effects of Talker and Talking Condition on word recognition accuracy scores, with listeners as the random effect. The model had an unstructured covariance matrix and the degrees of freedom were estimated using the between-within method. Various models were compared using ANOVA to determine whether individual fixed effects and their interaction explained additional variance in word recognition accuracy scores. Main effects were investigated using independent samples *t*-tests (Talker) and paired *t*-tests (Talking Condition) with Holm correction.

3 Results

Comparing a model that included both fixed effects with a model that included both fixed effects and an interaction term, the interaction term did not explain additional variance, $\chi^2(1) = 0.0011$, $p = 0.97$. Comparing a model that included both fixed effects with models that only included either fixed effect, the model with both fixed effects explained more variance than the model with either Talker or Talking Condition alone, $\chi^2(1) = 15.45$, $p < 0.001$ and $\chi^2(1) = 33.45$, $p < 0.001$, respectively. Thus, there was a significant main effect of Talker, $b = -0.62$, SE = 0.11, $t(7) = -5.98$, $p < 0.001$, and a significant main effect of Talking Condition, $b = 0.86$, SE = 0.22, $t(3) = 3.98$, $p = 0.01$, but no significant Talker × Talking Condition interaction.

Word recognition accuracy was best for talker F2 (Female 2), who was more intelligible than any other talker, p's < 0.05. Female talkers with the lowest reader quality scores (F1, F2) were more intelligible than female talkers with the highest scores (F5, F6), p's < 0.005. There was no significant difference between the male talkers (Figure 1).

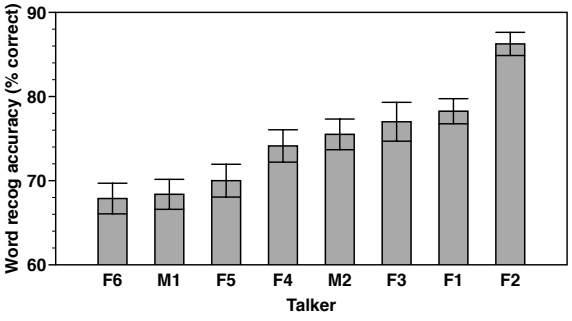


Figure 1: Mean word recognition accuracy scores of eight talkers, with standard error bars. F = Female; M = Male.

Word recognition performance was worse when talkers spoke in the Quiet condition than when they spoke in the Noise or HL+Noise conditions (p's < 0.05). No other pairs of conditions were significantly different (Figure 2).

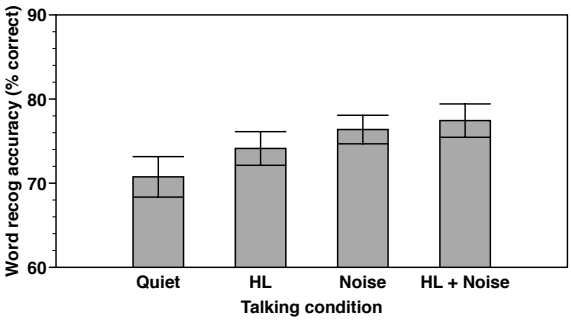


Figure 2: Mean word recognition accuracy scores for four talking conditions, with standard error bars.

4 Discussion

When speaking normally, older talkers showed a wide range of intelligibility scores (61 to 84%), similar to younger talkers in past research (65 to 85%) [7]. The "clear speech" of older talkers was not more intelligible than their normal speech, unlike younger talkers in most studies [8]. However, the benefit from speech produced in noise was similar to that of younger talkers under similar listening conditions [9]. Older talkers whose speech was perceived to be higher quality in quiet were actually less intelligible in noise. Future directions include testing older listeners to determine whether the characteristics of older talkers affect older listeners in a similar way as younger listeners.

Acknowledgements

This study was funded by an NSERC grant (PI: KPF).

References

[1] International Organization for Standardization. *Acoustics: Statistical distribution of hearing thresholds as a function of age (ISO 7029:2000)*. 2000. Geneva, Switzerland: Technical Committee ISO/TC 43 Acoustics.

[2] Goy H, Pichora-Fuller MK, Van Lieshout P. (2016). Effects of age on speech and voice quality ratings. *J Acoust Soc Am*. 2016;139(4):1648-1659. doi:10.1121/1.4945094

[3] Tillman TW, Carhart R. *An expanded test for speech discrimination utilizing CNC monosyllabic words: Northwestern University Auditory Test No. 6 (SAM-TR-66-55)*. 1966. Brooks Air Force Base, TX: USAF School of Aerospace Medicine.

[4] Wilson RH. Development of a speech-in-multitalker-babble paradigm to assess word-recognition performance. *J Am Acad Audiol*. 2003;14(9):453-470.

[5] Boersma P, Weenink D. *Praat: doing phonetics by computer*. [Software] 2016. Available from http://www.praat.org

[6] Studebaker GA. A "rationalized" arcsine transform. *J Speech Hear Res*. 1985;28(3):455-462. doi:10.1044/jshr.2803.455

[7] Bond ZS, Moore TJ. A note on the acoustic-phonetic characteristics of inadvertently clear speech. *Speech Comm*. 1994;14:325-337. doi:10.1016/0167-6393(94)90026-4

[8] Smiljanić R, Bradlow A. Speaking and hearing clearly: Talker and listener factors in speaking style changes. *Lang Linguist Compass*. 2009;3(1):236-264. doi:10.1111/j.1749-818x.2008.00112.x

[9] Pittman AL, Wiley TL. Recognition of speech produced in noise. *J Speech Lang Hear Res*. 2001;44(3):487-496. doi:10.1044/1092-4388(2001/038)

A COMPREHENSIVE APPROACH FOR ASSESSING THE EFFECTIVENESS OF FREQUENCY-LOWERING HEARING AIDS AND ELECTRIC ACOUSTIC STIMULATION (EAS) COCHLEAR IMPLANT FOR TREATING PEOPLE WITH A SEVERE-TO-PROFOUND HIGH-FREQUENCY HEARING LOSS.

Mathieu Hotton [*1,2,3] and François Bergeron[†1,2]

[1]Université Laval, Faculty of Medicine, Department of Rehabilitation, Québec, Canada.
[2]Centre for Interdisciplinary Research in Rehabilitation and Social Integration (CIRRIS), Québec, Canada.
[3]Centre Intégré universitaire de santé et de services sociaux (CIUSSS) de la Capitale-Nationale, Québec, Canada.

1 Introduction

Speech recognition tests have been used for many years to evaluate the effectiveness of hearing technologies and to make comparisons between different hearing aid or cochlear implant fittings. However, the use of speech perception measures alone for such purposes may not be appropriate. First, test settings and materials are not representative of a real-world communication environment. Also, many speech tests are vulnerable to learning and ceiling effects, particularly in the context of repeated measures. Moreover, the benefit from hearing technologies is known to be multidimensional, not only related to speech perception, but also to experienced hearing disabilities, limitations in social participation, quality of life and other personal factors [1].

In this article, results of a project where difficulties in speech measurements were encountered will be reported. The use of a more comprehensive assessment approach to circumvent those difficulties will be discussed. The project objective was to compare the effectiveness of frequency-compression (FC) and frequency-transposition (FT) hearing aids and of electric-acoustic stimulation (EAS) cochlear implant (CI) to improve speech perception for people with a severe-to-profound high-frequency hearing loss (HFHL).

2 Method

2.1 Participants

Ten adults, aged between 52 and 74 years old (6 women and 4 men), participated in the study. They were all candidates for an EAS implant considering their sensorineural severe-to-profound HFHL with aidable residual hearing in low frequencies. Participants had a previous experience with hearing aids of at least seven years.

2.2 Procedures

All participants tested FC and FT hearing aids following an ABAC single-subject design. A four to six weeks' baseline was first completed with their own hearing aids (phase A1), followed by an eight-week trial with one frequency-lowering (FL) device (phase B). Then, a second baseline of four weeks was repeated with their own hearing aids (phase A2). Finally, participants tried a second FL hearing aid for 8 weeks (phase C). Phases B and C were counterbalanced between subjects. After those trials, one participant also received an EAS implant. Consequently, follow-up time ranged from 16 to 32 weeks for all participants.

Speech recognition was measured each week, in free-field, using the Hearing in Noise Test (HINT). Sentences were presented at a fixed level of 63 dBA in quiet and in noise, at +10, +5 and 0 dB SNR, like many CI researches, and as it is currently done to assess CI candidacy in our clinic. Monosyllabic word recognition was also measured in quiet, at a presentation level of 60 dBHL.

The Glasgow Hearing Aid Benefit Profile (GHABP), the Abbreviated Profile of Hearing Aid Benefit (APHAB) and semi-structured interviews were used to collect participants' perspectives on the benefits of each technology. Questionnaires were completed after each trial; interviews were done at the end of the protocol.

Complementary data on EAS implant effectiveness were extracted from our database of EAS users. This data validated that the EAS participant was representative of our EAS users' population.

2.3 Analyses

Speech recognition data were first analyzed individually, using a visual single-subject method. Results of each test session were graphed, separately for each participant. A 95% confidence interval was computed around the mean of the second baseline and was used as a reference for performance comparison between technologies. Secondly, data from all participants were grouped by technology, and a group data analysis was done. Data from questionnaires were also analyzed individually, using published within-subject critical differences (APHAB) and norms (GHABP), and then grouped by technology.

Interviews were transcribed verbatim. Data were divided into meaning units, and then classified by themes and subthemes, using a qualitative inductive approach [2].

3 Results

3.1 Speech recognition

A summary of the individual effects obtained on the speech recognition tasks with the different technologies is shown in Table 1. The EAS implant allowed for the greatest improvement (+43%) when compared to conventional and

[*] mathieu.hotton.1@ulaval.ca
[†] francois.bergeron@rea.ulaval.ca

FL hearing aids; no deleterious effect was noted with this technology. FC and FT hearing aids provided some benefit in comparison with conventional hearing aids (+9 to +10%), but negative effects were also observed with those technologies for 2 and 4 subjects with FC and FT hearing aids respectively.

Table 1: Summary of the individual effects measured with each tested hearing technology.

Technology	Effect vs conventional h/a
EAS implant	Up to +43%
FC hearing aid	-6 to +10%
FT hearing aid	-22 to +9%

Learning and ceiling effects were encountered during speech recognition assessments with most subjects. Data from subject 11 shown in figure 1 was typical. An improvement in performance during the first baseline with her own hearing aids can be observed. This improvement is also present when comparing the performance of the subject between the two baselines. Data from subject 2 showed a clear ceiling effect as his performance in any FL condition varied between 96 and 100%. Those observations suggest that speech materials, or procedure, used in this experiment may have been inappropriate.

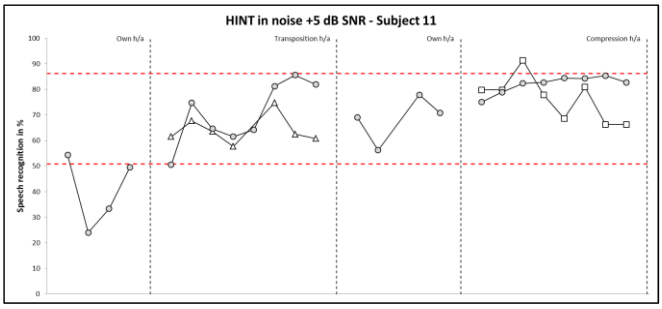


Figure 1: Speech recognition results to the HINT in noise at +5 dB SNR for Subject 11, for the complete 24-week follow-up.

3.2 Questionnaires

Eight and two participants reported a greater benefit with FL in the GHABP and the APHAB questionnaires respectively, in comparison with conventional hearing aids. No negative effect of FL was measured with the questionnaires. The EAS implant participant still reported a greater benefit with this technology. For four participants, results to the questionnaires matched speech recognition results; this was not the case for the other subjects. Five subjects did not get any improvement in speech recognition with FL hearing aids, but reported a significant improvement on the two questionnaires with those technologies.

3.3 Interviews

Nine participants experienced benefits with FL hearing aids. They reported better speech perception and listening comfort in everyday noisy situations (ex. in the restaurant, in the car, in a group conversation), and also improvements in environmental sound detection (ex. doorbell, microwave oven, birds). Those participants preferred FL hearing aids to their own hearing aids and would have kept a trial hearing aid if it were possible. Still, the participant with an EAS implant reported better benefit with this technology.

4 Discussion

Speech recognition results suggest that the EAS implant is the most effective hearing technology when compared to conventional or FL hearing aids. It allowed a greater gain than hearing aids, and no deleterious effect was observed with this technology. The gain provided by FL hearing aids was smaller. In fact, it was so small that it could probably have passed unnoticed in a standard clinical setting. Complementary data collected with standardized questionnaires and semi-structured interviews helped to demonstrate the importance of this effect, as these tools appear more sensitive to the effect of FL in real life, which was often undetected by speech recognition tests. This should be considered when assessing the effectiveness of hearing technologies and making clinical decisions.

Learning and ceiling effects imposed limits on the observable effect in this study. The measurement tools used may have been too repetitive or too easy for the study population, in the context of repeated measurements, or not enough representative of real life; they also may not be sensitive enough to the subtle effects of FL. Other alternatives should be considered for future research.

5 Conclusion

The EAS implant appears as the first indication for treating people with a high-frequency hearing loss. However, FL hearing aids can provide a significant benefit for some individuals. Even if the effect as assessed with traditional speech recognition measures is small, it can be significant from the perspective of the patient. In this context, and considering the potential risks and high costs related to cochlear implantation, trials with FL should be considered on an individual basis prior to implantation.

Results support the use of a more global approach when assessing benefits of hearing technologies. For this purpose, researchers and clinicians should consider not to rely entirely on speech measures, but to also use other data sources, such as standardized questionnaires and interviews.

Acknowledgments

This study was funded by the Fonds de Recherche du Québec en santé (FRQ-S), the CIUSSS de la Capitale-Nationale, and the CIRRIS. Hearing aid loaners were provided by Phonak Canada and Widex Canada.

References

[1] Gatehouse S. (1994). Components and determinants of hearing aid benefit. *Ear Hear*, 15(1), 30-49.

[2] Creswell J. W. (2009). *Research Design, Qualitative, Quantitative, and Mixed Methods Approaches (3rd ed.)*. Thousand Oaks: SAGE Publications, Inc.

LISTENING TO SPEECH IN NOISE: EVALUATION AND TRAINING

Benoît Jutras[*1,2], Mojgan Owliaey[†3], Lyne Lafontaine[‡4], Alexis Pinsonnault-Skvarenina[✦1,2] and Jean-Pierre-Gagné[★1]
[1]École d'orthophonie et d'audiologie, Université de Montréal, 7077 avenue du Parc, Montréal, Québec, H3N 1X7
[2] Centre de recherche, CHU Sainte-Justine, 3175, Chemin de la Côte-Sainte-Catherine, Montréal, Québec, H3T 1C5
[3] Institut Raymond-Dewar, 3 600, rue Berri, Montréal, Québec, H2L 4G9
[4] Regional Support Services in the educational setting – Hearing Disorders,
8700 boul. Champlain, LaSalle, Québec, H8P 3H7

1 Introduction

Having a conversation in noise is a challenge for everyone. This could be a disabling situation for individuals with hearing problems, especially for those with auditory processing disorder (APD). APD "is presumed to originate in the auditory system and is characterized by a persistent limitation in the performance of auditory activities and significant consequences on participation" [1] (p. 13). Close to 60% of children with APD and normal hearing sensitivity, being involved in a rehabilitation program, failed a speech in noise test [2]. This also seen in adults who report difficulty in hearing in noise, despite the fact that they have a normal audiogram [3,4]. Although, this listening difficulty in individuals with APD is well known, there are only a few studies conducted on the efficacy of a listening training programs in noise in children [5,6].

Maggu et Yathiraj (2011) [5] exposed five children suspected of APD to an auditory training program where they listened to short stories in noise at several signal to noise ratios and answered questions, over 15 to 20 training sessions of 25 minutes. Five children with also suspected APD did not participate to the training. Results showed that after the training program, there was a significant improvement on speech in noise tests in the experimental group, but not in the control group. In a study by Loo et al. (2016)'[6], 20 children with APD participated to an auditory training program where they had to identify words presented in noise as well as digits and words in a dichotic listening task. They also listened to stories in noise. The training was over 12 weeks with five 30-minute sessions/week. Nineteen children with APD did not take part to the training as the control group. Results showed, that the experimental group improved significantly their performance on identifying sentences in noise relative to the control group, following the end of the training. No known study was found in the literature related to this topic in adults with APD.

The present document reports results collected in three pilot studies among children with APD or adults complaining of listening difficulties in noise. The objective aimed to examine, in part, the benefits of a desensitization to noise therapy in children and in adults on speech perception in noise.

[*]benoit.jutras@umontreal.ca
[†]mowliaey@raymond-dewar.gouv.qc.ca
[‡]lafontaine.lyne@sympatico.ca
[✦]alexis.pinsonnault-skvarenina@umontreal.ca
[★]jean-pierre.gagne@umontreal.ca

2 Method

2.1 Participants

In the study 1 (S1), ten 8-12 year-old children with APD with normal hearing sensitivity participated in the study. Five were in the experimental group who received the listening training in noise and five in the control group (no training). More children were involved in study 2 (S2), ten took part in the therapy sessions and six did not. In study 3 (S3), nine adults with normal hearing sensitivity, complaining of listening difficulties were involved in the pilot study. Four were exposed to the noise desensitization therapy and five did not receive any therapy.

2.2 Material and Procedure

In the three studies, stimuli were composed of words, sentences and short stories, presented at comfortable level through a CD player in S1 and through headphones connected to a computer for S2 and S3. In S1, the stimuli were supported with pictures while in S2 and S3, they were presented with a software program developed for the experiment (*Logiciel d'écoute dans le bruit – LEB*, meaning Listening in noise software). For S1, children in the experimental group participated in nine 60-minute sessions, one session/week. In S2 and S3, it was two sessions of 30 minutes/week during 9 to 13 weeks. During the sessions, children took part in activities of word discrimination, word identification, sentence identification, comprehension of directions, comprehension of complex sentences and comprehension of short stories. In the three studies, the signal to noise ratio was adapted according to the performance of the participants.

Pre- and post-training measures included speech in noise tests. In S1 and S2, the Hearing In Noise Test (HINT), adapted in French [7], was used. It is composed of sentences presented in quiet and in noise. The sentences were delivered through a speaker standing one meter in front of the participant. In the noise condition, a speaker was also located at 90 degree on each side of the child at one meter distance. The child had to repeat the sentences heard. In S3, the *Test de mots dans le bruit* (TMB, meaning Word in noise test) [8] was utilized. Thirty-five monosyllables were sent through headphones, connected to an Astera audiometer. In each ear, the level was set at 60 dB HL and a babble noise was heard at 55 dB HL simultaneously. The adult participants had to identify the words heard.

3 Results, Discussion and Conclusion

No statistical analyses were performed because of the limited number of participants in the studies. Data observation is reported, comparing the distribution of the participants in each group according to the benefits revealed by the improvement in performance. In S1 and S2, the difference between pre- and post-training performance obtained on the HINT was calculated. This performance corresponded to the signal to noise ratio where 50% of correct sentences were corrected repeated.

In S1, results showed that all participants (100%) in the experimental group improved their performance on the HINT (Figure 1). Indeed, the signal to noise ratio decreased by at least more than 1 dB after the end of the training program compared to the pre-training results. In the control group, an improvement in performance (i.e., lower SNR) was present in two of the five participants (40%). However, these two participants had a greater increase (average -2.1 dB) in performance than the participants in the experimental group (average -1.8).

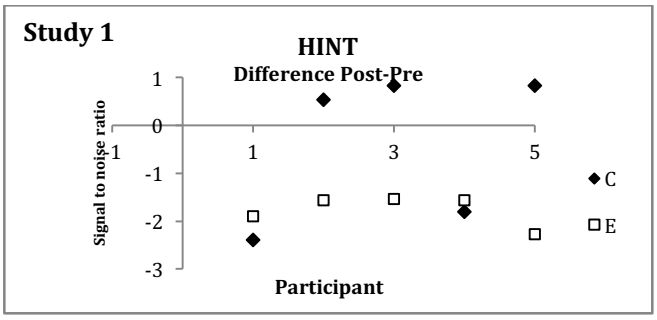


Figure 1. Difference of pre- and post-training HINT threshold measured in five participants in the experimental group (E) and five in the control group (C) for study 1.

Results for S2 are different in terms of percentage of participants showing improvement (Figure 2). In fact, 60% of the participants in the experimental group showed a decrease in SNR of at least 1 dB while 33% reached this level in the control group. However, as in S1, the average of the signal to noise ratio decrease was greater in the control group (-2.45) than in the experimental group (-2.1).

The S1 and S2 results suggest that following the listening in noise training program, children with APD are more likely to show improvement in their performance when listening to speech in noise, but with a limited magnitude. The positive changes cannot be attributed only to the benefits of the treatment because children in the control group can reach equal or greater performance compared to the one in the experimental group over time.

Based on these data, the TMB was used in the experiment with adults to examine if this test could me more sensitive to the effect of noise desensitization therapy on speech in noise test performance than the HINT.

Result of the S3 revealed that one of four adults (25%) in the experimental group and also one in the control group showed an increase of the percentage of correct responses of at least 5% on the TMB (Figure 3). This was an increase of 5.5% and 7% respectively.

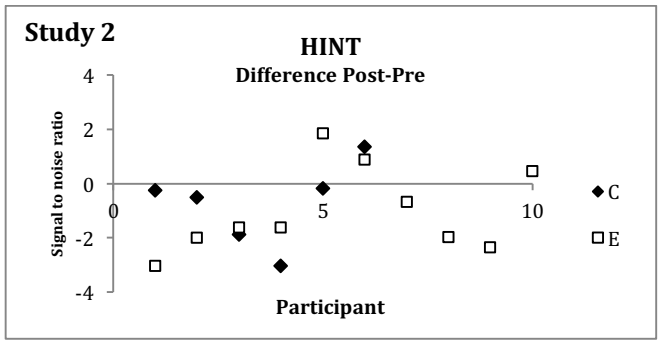


Figure 2. Difference of pre- and post-training HINT threshold measured in ten participants in the experimental group (E) and six in the control group (C) for study 2.

These preliminary results could suggest that the TMB is not a good test to capture the benefits of this therapy type. However, the fact that the majority of the S3 participants did not fail this test by having a score within normal limits could influence the conclusion. In other words, adults who fail the TMB might be better candidates to show improvement on this test.

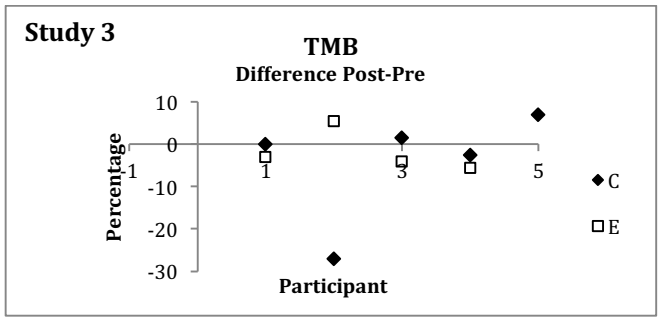


Figure 3. Difference of pre- and post-training TMB percentage of correct responses measured in four participants in the experimental group (E) and five in the control group (C) for study 3.

Acknowledgments

S1 was funded in part by the Speech-Language and Audiology Canada, S2 by the Ministry of Education, Recreation and Sports of Quebec, Canada and S3 by The Hearing Foundation of Canada.

References

[1] CSGSLPA. http://sac-oac.ca/sites/default/files/resources/Canadian-Guidelines-on-Auditory-Processing-Disorder-in-Children-and-Adults-English-2012.pdf, 2012.
[2] ASHA. http://www.asha.org/ members/deskref-journals/deskref/default, 2005.
[3] Saunders, G. H., et al. (1989). Ear & Hearing, 10, 200-208.
[4] Saunders, G. H., et al. (1992). Ear & Hearing, 13, 241-254.
[5] Maggu, A. R. et al. (2011). Canadian Journal of Speech-Language Pathology and Audiology,, 35, 56-.
[6] Loo, J. H, et al. Ear & Hearing, 37, 38-47.
[7] Vallancourt, V. et al. (2008). 29, 453-466.
[8]Lagacé, J. (2010). Revue canadienne d'orthophonie et d'audiologie, 32, 19-30.

INTELLIGIBILITY, RESPONSE TIME, AND LISTENING EFFICIENCY OF NATIVE AND NON-NATIVE LISTENERS IN VARIOUS ACOUSTICAL CONDITIONS

Alice Lam *[1] **and Murray Hodgson** †[1]
[1] Department of Mechanical Engineering, University of British Columbia, Vancouver, BC, Canada

1 Introduction

This study evaluates the speech perception of English as a Second Language (ESL) and native English-speaking students and in various acoustical conditions. Three different metrics for speech intelligibility are compared: the intelligibility score (IS), response time, and listening efficiency (DE).

Speech intelligibility scores are based on the accuracy of participants' responses to listening tests [1]. The response times of participants can be used as a more sensitive measure, to distinguish between conditions that appear equally intelligible [2].

Prodi *et al.* [3] introduced a combined metric, the "direct listening efficiency" or "listening efficiency" (DE), which combines IS and response time to evaluate the amount of effort required from a listener to understand speech. The development and evaluation of this metric has so far been in the Italian language. This project is an initial evaluation of the DE metric in English, for both native and non-native (ESL) speakers.

2 Method

2.1 Test Material

The Diagnostic Rhyme Test (DRT) [1] was used for this study. The test material was split into 16 lists of 6 word pairs, each recorded by one of five native speakers (2 male, 3 female) in an anechoic, low-noise environment (Condition A). These recordings were auralized into conditions B (reverberation dominant) and C (some reverberation plus "babble and activity" noise [4]). T_{mid} = 1.08 s and 0.78 s for conditions B and C, respectively, These conditions corresponded to an STI (Speech Transmission Index) of 0.78 for Condition A, and an STI of 0.575 for Conditions B and C.

2.2 Test Procedure

During the test, participants listened to the recorded word with carrier phrase, and were then asked to select the word that they had heard from one of three options (the two words in the word pair, or "none of the above"), as shown in Figure 1. This process would be repeated for each pair in the list, making up one test. Each participant completed one test from condition A as practice, followed by one test each from conditions A, B and C.

All tests took place in an anechoic environment. Playback of the test material (all monaural) was done using a single KRK Rokit 5 G2 loudspeaker 1.5m away from the seated participant, at head height. Playback was calibrated to 66 dBA, corresponding to "raised" vocal effort [1].

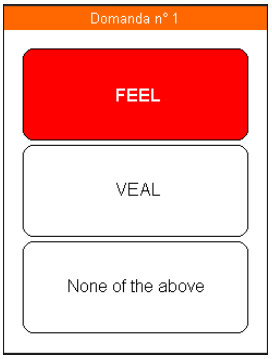


Figure 1: Sample Intelligo test screen.

The Intelligo system [5] was used to facilitate data collection and synchronization with the playback system. Participants used a touchscreen handset to enter responses in real time. Following the tests, participants were asked to fill out a brief questionnaire regarding their proficiency in English, their country of origin, and other languages spoken.

2.3 Participants

All 37 participants were university students aged 18 to 32, with no reported hearing loss. 25 individuals (15 male, 10 female) identified as ESL, and 12 identifying as non-ESL (native speakers of English). We defined an individual as ESL if they were not exposed to the English language until after 3 years of age. The amount of time in which the ESL participants had lived in a primarily-English speaking environment ranged from 6 months to 15 years.

3 Results

The IS and DE were calculated for each test using the method in [3], as follows:

$$Intelligibility\ Score\ (IS) = (average\ raw\ score + 1) * 0.5$$

(raw scores: correct = +1, incorrect = -1, "none of the above" = -.05)

$$Listening\ Efficiency\ (DE, s^{-1}) = \frac{Intelligibility\ Score}{Mean\ Response\ Time\ (s)}$$

We compare the effectiveness of these three metrics in performing two different tasks: differentiating between sound fields (conditions A, B, C) and differentiating language ability (ESL and non-ESL). For the former, the Kruskal-Wallis test was used. Where applicable, the Wilcoxon rank sum test was performed *post hoc* in order to determine which pair of conditions showed significant differences. For the latter, Wilcoxon Rank Sum tests were performed.

* avl@alumni.ubc.ca
† murray.hodgson@ubc.ca

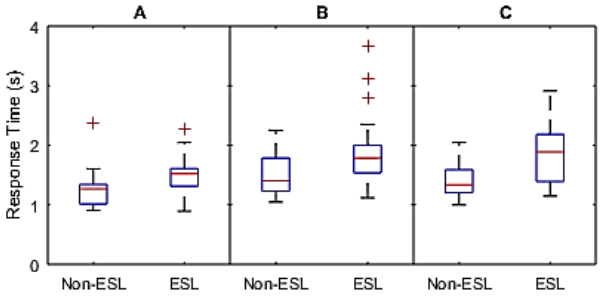


Figure 2: Response times for all cases.

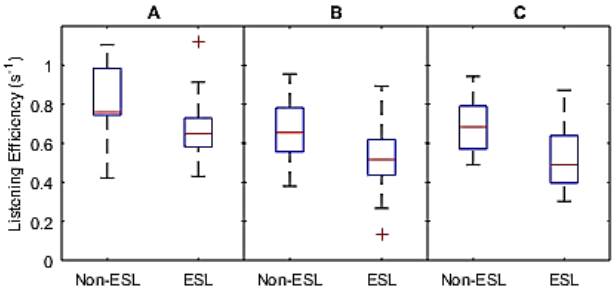


Figure 3: Listening efficiency for all cases.

3.1 Differentiating Sound Fields

Using Intelligibility Score

There was no significant difference between conditions A, B and C in either the ESL or non-ESL case ($p = .3826$ and $p = .1714$, respectively).

Using Response Time

As shown in Figure 2, there was no significant difference between conditions A, B and C for the non-ESL case. In the ESL case, response time was lower (i.e. faster) in A than it was in B or C ($p = .0154$ and .0127, respectively).

Using Listening Efficiency

As shown in Figure 3, there was no significant difference between conditions A, B and C for the non-ESL case. In the ESL case, DE was higher in A than it was in B or C ($p = .0101$ and .0091, respectively).

3.2 Differentiating Language Ability

Using Intelligibility Score

None of the three conditions showed any statistically significant differences ($p > 0.15$ in all cases) between non-ESL and ESL participants' intelligibility scores.

Using Response Time

As shown in Figure 2, all three conditions showed statistically significant differences between non-ESL and ESL participants' average response times ($p = .0249$, .0270 and .0087 for conditions A, B and C, respectively).

Using Listening Efficiency

As shown in Figure 3, all three conditions showed statistically significant differences between non-ESL and ESL participants' average DE ($p = .0178$, .0163 and .0079 for conditions A, B and C, respectively).

4 Discussion

The relative ineffectiveness of using IS alone could be due to the fact that conditions A, B and C were generally similar: in all cases participants could understand what was being said after some thought, and there were no cases where speech was genuinely unintelligible.

When differentiating between sound fields, only the ESL case showed significant differences. This suggests that ESL students have a disadvantage in acoustically non-ideal conditions compared to their non-ESL peers, a result that aligns with [6]. The smaller sample size of non-ESL may also have been a factor.

Comparing non-ESL and ESL results, ESL participants showed a higher average response time and lower average DE. This result aligns with the idea that these two metrics give a picture of "listening effort" as posited in [3].

Though the qualitative results for DE and response time were similar in all cases, in every case the DE results were more significant, showed fewer statistical outliers, and less variance.

5 Conclusion

This study compared the effectiveness of using IS, response time, or DE to differentiate between either acoustical conditions or language ability. It was found in all cases that using DE was the most effective option, with response time being a close second. Since incorporating the intelligibility score in a listening test that already measures response time is relatively trivial, using DE instead of response time alone is the best option to evaluate listening effort in addition to speech intelligibility.

Acknowledgments

We thank Nicola Prodi for providing the Intelligo test system as well as his guidance and advice, and Katie O'Brien for her assistance in organizing and conducting the listening tests.

References

[1] *ANSI S3.2-2009 Method for Measuring the Intelligibility of Speech over Communication Systems*, 2009.

[2] M. H. L. Hecker, K. N. Stevens and C. E. Williams, "Measurements of Reaction Time in Intelligibility Tests," *J. Acoust. Soc. Am.*, vol. 39, no. 6, pp. 1188-1189, 1966.

[3] N. Prodi, C. Visentin and A. Farnetani, "Intelligibility, listening difficulty and listening efficiency in auralized classrooms," *J. Acoust. Soc. Am.*, vol. 128, no. 1, pp. 172-181, 2010.

[4] N. Prodi, C. Visentin and A. Feletti, "On the perception of speech in primary school classrooms: Ranking of noise interference and of age influence," *J. Acoust. Soc. Am.*, vol. 133, no. 1, pp. 255-268, 2013.

[5] N. Prodi, C. Visentin and C. Belletini, "Listening efficiency testing," in *45th AES Conference Proceedings*, Helsinki, 2012.

[6] S. J. van Wijngaarden, H. J. M. Steeneken and T. Houtgast, "Quantifying the intelligibility of speech in noise for non-native listeners," *J. Acoust. Soc. Am.*, vol. 111, no. 4, pp. 1906-1916, 2002.

SPEECH AUDITORY BRAINSTEM RESPONSES AND WORDS-IN-NOISE RECOGNITION SCORES: IS THERE A LINK?

Josée Lagacé*, Amineh Koravand, Jordon Thompson et Suzanne Lteif
Audiology and Speech-Language Pathology Program, Health Sciences Faculty, University of Ottawa, Ottawa, Canada

1 Introduction

Normally developing children obtain lower scores on speech recognition in noise tasks compared to adolescents and adults.[1] However, many children with school-based learning problems have more difficulties recognizing speech in the presence of background noise compared to peers of similar age.[2] Although there are effective interventions to cope for these listening problems, early identification of speech recognition problems in noise has been an ongoing challenge. Because of immature attention, language and hearing skills, it is not possible to accurately measure speech recognition in noise abilities before the age of six years old.

Some authors have suggested that measuring the electrophysiological response of the brainstem following the presentation of speech stimuli (cABR) may provide an indirect measure of proficiency in listening to speech in noise.[3] cABRs show a high degree of transparency to the stimulus waveform and have a high test-retest reliability.[3]As such, cABRs are being considered for evaluating speech recognition in noise ability in young children as they do not require patient response. In the meantime however, available data showing correlation between behavioral scores on speech recognition in noise tasks and electrophysiological measures are scarce[4].

The aim of the present study was to investigate possible relationship between variables of cABR and the performance measured at a word-in-noise recognition task on adults. This is part of a larger study aiming at developing a screening tool for listening difficulties in noise for young children.

2 Method

This study had been previously approved by the Ethic board of the institution and was conducted at the *Laboratoire de recherche en audition* of the University of Ottawa (Ottawa, Ontario).

2.1 Participants

Forty-three normal hearing adults from 18 to 30 years old participated in this study. They had no history of hearing problems or language development difficulties. They all had attended francophone schools, up to high school. All participants had a hearing screening at 20 dB HL from 250 to 8000 Hz prior to experimental measures.

josee.lagace@uottawa.ca
amineh.koravand@uottawa.ca

2.2 Procedure

Experimental measures included monosyllabic word recognition tasks presented at 60 dB HL, with and without ipsilateral competing white noise in the right ear with insert earphones. Electrophysiological measures with a pre-recorded 40 ms syllable /da/ presented at 60 dB HL, with and without competing ipsilateral competing white noise in the right ear with insert earphones were following. The electrophysiological responses were collected using a vertical electrode montage (active Cz, forehead ground, ipsilateral mastoid bone reference) using Ag–AgCl electrodes.

3 Results

The sample was divided in three groups as following:

- Group 1 (n=14) was exposed to SNR of + 5 dB for the word recognition task in noise and the cABR measured with noise;
- Group 2 (n=15) was exposed to SNR of + 0 dB;
- Group 3 (n=14) was exposed to SNR of -5 dB.

3.1 Word-in-noise recognition scores

As shown in Table1, close to perfect average score was obtained by each group at the word recognition task (35 items) without competing noise (quiet condition). There was no statistical difference in the mean scores between the groups in quiet condition ($F_{(2,42)=}.753$, p =.48).

Table 1. Mean correct word recognition score for each group with standard deviation.

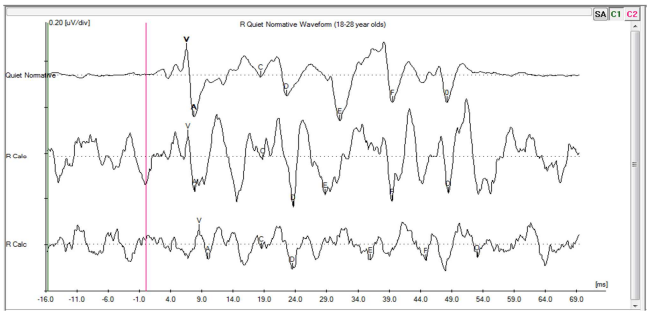
	Quiet condition	Noise condition
Group 1	93.47% (±4.811)	91.02% (±3.56)
Group 2	94.48% (±4.64)	84.29% (±5.18)
Group 3	95.71% (±5.10)	76.73% (±9.87)

The average score was lower when words were presented with an ipsilateral white noise (noise condition). As expected, the Group 3 who was exposed to a higher noise level (i.e.: SNR of +5 dB) had the lowest average correct score. There was a statistically significant in the correct recognition scores for the three groups ($F_{(2,42)}$ =15.82, p =.00). Post hoc comparisons using the Tukey HSD test indicated that the mean score for Group 1 was significantly different from Group 2, as well as Group 3. The Group 2 did also differ significantly from Group 3.

3.2 Speech evoked ABR

An example of a cABR waveform is presented in Figure 1. For the purpose of this paper, only the average amplitude and latency of wave V and A was computed as they are more commonly recognized by clinicians.

Figure 1. Example of a cABR waveform. Middle: cABR waveform in quiet condition. Lower: cABR waveform in noise condition.



The mean latency values of wave V and A are presented in Table 2. As expected, there was a significant (p< .00) increase of wave V latency from the quiet to the noise condition, as well as for wave A latency. There was no effect of SNR on the latency value of wave V or wave A. The difference in the mean latency of wave V ($F_{(2,42)}$ =0.24, p =.79) and wave A ($F_{(2,42)}$ =0.09, p =.92) between the groups did not reach statistical significance.

Table 2. Mean latency (in ms) of wave V and A with and without noise (standard deviation).

	Wave V No noise	With noise	Wave A No noise	With noise
Group 1	6.77 (±0.32)	9.87 (±2.47)	8.22 (±0.78)	12.23 (±2.84)
Group 2	6.53 (±0.54)	9.26 (±2.51)	7.70 (±0.38)	12.04 (±3.02)
Group 3	6.37 (±0.33)	9.83 (±2.79)	7.58 (±0.64)	11.79 (±2.96)

The mean amplitude values of wave V and A are presented in Table 3. As expected, there was decrease of the wave V amplitude from the quiet to the noise condition, as well as for wave A. That decrease of amplitude between quiet and noise condition only reached statistical significance for wave A amplitude in Group 1 and 3.

There was no significant effect of SNR on the amplitude value of wave V or wave A. The difference in the mean amplitude of wave V ($F_{(2,42)}$ = 1.24, p =.30) between the groups did not reach statistical significance, nor of wave A ($F_{(2,42)}$ = 1.80, p =.18). As opposed to behavioral measures in noise, no linear trend can be identified with the latency or the amplitude of wave V and A with decreasing SNR.

3.3 Word recognition scores in noise and speech evoked ABR

Correlations between speech recognition scores in noise and cABR variables (latency of wave V and B measured with and without competitive noise, as well as amplitude of wave V and B measured with and without noise) were computed for each group. Only the correlation between the word recognition score and the amplitude of wave V reached the statistical significance (p=.03).

Table 3. Group average amplitude value (in mV) of wave V and A with and without white noise (standard deviation).

	Wave V No noise	With noise	Wave A No noise	With noise
Group 1	.10 (±.05)	.08 (±.03)	-.18 (±.07)	-.08 (±.04)
Group 2	.15 (±.06)	.10 (±.06)	-.20 (±.05)	-.14 (±.16)
Group 3	.10 (±.06)	.08 (±.04)	-.21 (±.08)	-.08 (±.05)

4. Discussion and Conclusion

Similarly to studies investigating effects of white noise on cABR,[5] there was an increase of latency of wave V and A from quiet to noise condition, as well as reduction of their amplitude. However, there was no effect of SNR on the latency of wave V or A, neither on the amplitude. For example, the mean latency of wave V measured in noise condition was not significantly different between the groups. The presence of background noise appears to affect the processing of the stimulus at the subcortical level, but there is no linear trend that can be observed with the increase of the noise level, as can be observed with word recognition tasks in noise. The absence of SNR effect may be due to the fact that the three groups were not exposed to the three SNRs.

Only one correlation between speech recognition score in noise with one cABR variable at only one SNR condition, which is not sufficient at this point to suggest any link between cABR and hearing in noise. Analyses of other cABR components, such as transition wave (C), Frequency Following Responses (D, E, F) and the offset wave (O) could bring some insights about the possibility to use cABR as a clinical tool to identify speech in noise problems.

Acknowledgments

The authors wish to thank all the participants.

References

[1] R.H. Wilson, N.M. Farmer, A. Gandhi, E. Shelburne, E., & J. Weaverby. Normative data for the Words-in-Noise Test for 6- to 12-year-old children. *JSLHR*, 53: 1111, 2010.

[2] C.M. Warrier, K.L. Johnson, E.A. Hayes, T. Nicol, & N. Kraus, Learning impaired children exhibit timing deficits and training-related improvements in auditory cortical responses to speech in noise. *Exp. Brain. Res,* 157: 431, 2004.

[3] S. Anderson and N. Kraus. Objective neural indices of speech-in-noise perception. *Trends Amplif*, 14:73, 2010.

[4] S. Anderson, A. Parbery-Clarke, T.White-Schwoch and N. Kraus. Auditory brainstem response to complexe sounds predics self-reported speech-in-noise performance. *JSLHR*, 56:31, 2013

[5] F. Prévost, M. Laroche, A.M. Marcoux & H.R. Dajani, H. R. Objective measurement of physiological signal-to-noise gain in the brainstem response to a synthetic vowel. *Clin Neurophysiol*, 124:52, 2013.

WHAT DO FORCED ALIGNMENT LIKELIHOOD SCORES TELL US ABOUT THE ALIGNED SPEECH?

Ayushi Mrigen [*1], Daniel Brenner [†2] & Benjamin V. Tucker [‡2]
[1]Department of Mathematics, Indian Institute of Technology, Kharagpur
[2] Department of Linguistics, University of Alberta, Edmonton, Canada

1 Introduction

Forced alignment is an automatic speech recognition procedure frequently employed in the speech sciences. It is typically used to time-align a string of phones with a speech recording on the basis of some operationalization of acoustic similarities to averaged representations of the phones over a corpus. Ordinarily these similarity measures are not of interest to researchers, who simply want to know where in time the phones are. This study investigates the potential usefulness of these alignment scores for phonetic research, considering the phonetic properties which are and are not represented in the scores. The hypothesis is that these scores can be used to infer acoustic relationships along some dimensions between the aligned speech and the corpus on which the models were trained.

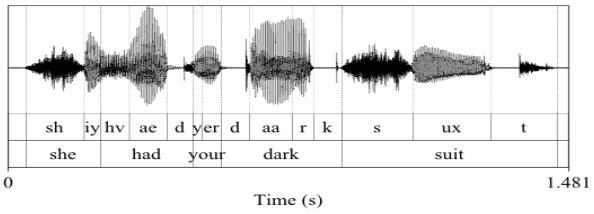


Figure 1: Sample waveform for a portion of a TIMIT sentence shown with the phone sequence aligned to it.

2 Method

2.1 What do alignment scores mean?

Forced alignment is essentially a system of optimized matching. The models of the system are trained on a corpus to provide an averaged acoustic representation of each phone in the corpus. During alignment, the system is given a string of phones and a sound recording, and it apportions continuous sections of the sound recording to the phones to maximize the similarity of the aligned portions to the averaged models. The typical forced aligner (like the Penn Aligner) is implemented using Hidden Markov Models [1]. This approach to symbol-sound mapping has a variety of practical applications.

For the problem of isolated word recognition, if each spoken word is represented by a sequence of speech vectors or observations O, defined as; $O = o_1, o_2, \ldots o_T$. The isolated word recognition problem can then be regarded as that of computing; arg max $\{P(w_i|O)\}$, where w_i is the i'th vocabulary word. However, since this probability is not computable directly we use the Bayes' rule which gives us: $P(w_i|O) = (P(O|w_i)P(w_i))/P(O)$

2.2 Modification in the typical forced aligner

A typical forced aligner, like the Penn Forced Aligner [4] takes two inputs, the audio file which contains the speech, and the text file with the transcription of the speech. The aligner produces a PRAAT [2] TextGrid file as output, which contains the alignment in typically two tiers: (1) the word level, and (2) the phone level (Figure 1). This output contains the name of the word or the phone, and the start and end time of occurrence of the same within the speech file. Within the alignment script, the Penn Aligner calls the function HVite. The original output of HVite is in the form of a MLF file, which contains, among other things the probability score which the Viterbi algorithm computes while aligning the file. This output, from the Hidden Markov model is of interest to us. In order to investigate more about the score, the code to convert the MLF file to the TextGrid was modified to simultaneously print these probability scores alongside the name of the phone/word.

2.3 Measuring distances between vowels

In order to test the hypothesis that the log likelihood scores obtained from forced aligners can be employed to tell us about phonetic distances, these scores need to be compared with an existing measure of distance. A straightforward perceptually relevant geometric distance can be computed for vowels in the F1 & F2 space, so we selected this metric for comparison. A set of eight vowel phones was selected and the distance between every two pairs was measured using the standard F1-F2 technique, and by the forced aligner. In order to measure the distance between each pair, every speech file was aligned with the text file for every vowel (tokens of [i] were aligned with the [ɪ] phone, the [e] phone, the [ɛ] phone, etc.). It was noticed that for each vowel ranking obtained in the descending order of scores matched approximately with the increasing order of F1-F2 distances.

2.4 Training on HTK

Different kinds of training have an effect on the pattern of alignment scores obtained. As a result, two sets of training models were created after training on different sets of speech data. The first model was trained on the TIMIT corpus [5], which is composed of read sentences. The other model was trained using the Buckeye corpus [3], which contains interview recordings. The two corpora were chosen

[*] ayushimrigen11@gmail.com
[†] brenner@ualberta.ca
[‡] benjamin.tucker@ualberta.ca

to compare the differences in the patterns of scores from the read speech and the conversational speech models.

2.5 Aligning larger corpora

For testing purposes a holdout section was selected from both TIMIT and Buckeye. While the TIMIT corpus already has a test section in the corpus, a bootstrapped holdout section was created for the Buckeye corpus. For both these sections, they were first aligned with their original transcription, using the models trained on data from the same corpus. From the TextGrid files obtained by this alignment, a list of files containing each vowel was created. The original vowel in the file was substituted one by one with every other vowel in the list ([i] was substituted with the [ɪ] phone, the [e] phone, the [ɛ] phone, etc. one by one). The scores of each such alignment was recorded. In this way we obtained the cross comparison scores between each vowel pair.

3 Results

A ranking of this cross comparison is shown in Table 1. The log likelihood scores were compared to the F1-F2 phonetic distances. The result of a regression for each vowel is shown in Table 1.

Table 1: LogLik ranking & correlations by TIMIT test vowel.

Vowel	Log Likelihood ranking	Correlation
aa	[aa, ow, ah, ae, ey, uw, ih, iy]	-0.847
ae	[ae, ey, ah, ow, aa, ih, uw, iy]	-0.61
ah	[ow, aa, ah, ae, ih, ey, uw, iy]	-0.761
ey	[ey, iy, ae, uw, ih, ow, ah, aa]	-0.800
ih	[ih, ae, iy, uw, ey, ow, ah, aa]	-0.761
iy	[iy, ey, uw, ih, ae, ow, ah, aa]	-0.846
ow	[ow, aa, ah, ae, uw, ey, ih, iy]	-0.884
uw	[ow, uw, iy, ih, ey, ah, ae, aa]	-0.429

These results show a negative correlation with the F1-F2 distances measured manually. However, when a similar analysis was done with Buckeye aligned on the two sets of models, and TIMIT aligned on Buckeye, the negative correlation appears weaker (Table 2).

Table 2: Correlation coefficients comparing data from the different models.

Vowel	Buck-Buck	Buck-TIMIT	TIMIT-TIMIT	TIMIT-Buck
aa	0.015	-0.102	-0.848	0.036
ae	-0.222	-0.312	-0.613	-0.428
ah	-0.021	-0.090	-0.761	0.100
ey	-0.576	0.522	-0.800	0.155
ih	0.056	0.480	-0.762	0.101
iy	-0.074	0.163	-0.846	0.164
ow	-0.109	-0.127	-0.884	-0.167
uw	0.054	0.108	-0.430	0.047

When comparing the Buckeye corpus model we find that the correlation coefficients between distances and scores for models trained on both corpus is low. With the TIMIT corpus the correlation coefficients are high in general when TIMIT is aligned with TIMIT models, except for 'uw' and the coefficient is low when it is aligned on Buckeye models. Figure 2 illustrates the distribution of scores for the diphthong [aɪ] for the two different holdout datasets when fit using the models from the two different corpora. This example illustrates the very different results that can be obtained by using different models and data.

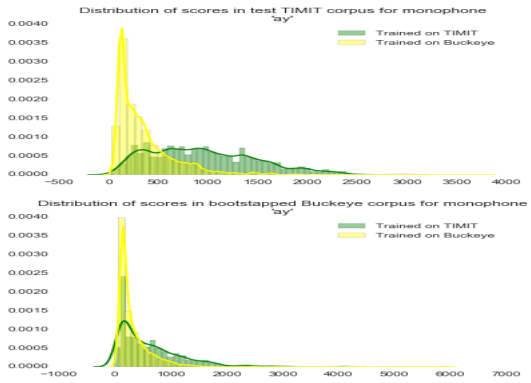


Figure 2: Distributions of alignment scores for the same phone on TIMIT and Buckeye, trained on the two corpora.

4 Conclusion

These probability measures seem to show a relationship with some acoustic characteristics of the segments. It is clear that these scores behave differently with different training, and this relationship may be exploitable to assess typicality of sounds within a given corpus context.

Acknowledgments

Funding for this project was provided by the University of Alberta Research Experience Program and Social Sciences and Humanities Research Council: #435-2014-0678.

References

[1] Baum, L. E.; Petrie, T. (1966). "Statistical Inference for Probabilistic Functions of Finite State Markov Chains". *The Annals of Mathematical Statistics* 37 (6): 1554–1563.

[2] Boersma, Paul & Weenink, David (2016). Praat: doing phonetics by computer [Computer program]. Version 6.0.19, retrieved 13 June 2016 from http://www.praat.org/

[3] Pitt, M. A., Johnson, K., Hume, E., Kiesling, S., & Raymond, W. 2005. The Buckeye corpus of conversational speech: labeling conventions and a test of transcriber reliability. *Speech Communication* 45, 89-95.

[4] Yuan, J. & Liberman, M. 2008. Speaker identification on the SCOTUS corpus. *Proceedings of Acoustics 2008*.

[5] Zue, V. & Seneff, S. Transcription and Alignment of the TIMIT Database. *Proceedings of the 2nd Meeting on Advanced Man -- Machine Interface through Spoken Language* 1988, 11.1-11.10.

EFFECTS OF MODALITY AND LINGUISTIC MATERIALS ON MEMORY

Jenna Pattison*[1], M. Kathleen Pichora-Fuller†[1,2,3] and Sherri L. Smith‡[4,5]
[1]Department of Psychology, University of Toronto, Mississauga, Ontario
[2]Toronto Rehabilitation Institute, University Health Network, Toronto, Ontario
[3]Rotman Research Institute, Baycrest Hospital, Toronto, Ontario
[4]Audiologic Rehabilitation Laboratory, Auditory Vestibular Research Enhancement Award Program, Veterans Affairs Medical Center, Mountain Home, Tennessee
[5]Department of Audiology and Speech Language Pathology, East Tennessee State University, Johnson City, Tennessee

1 Introduction

Audiologists can test memory with auditory or visual stimuli. On the one hand, auditory factors such as the level of presentation or the signal-to-noise ratio can affect recall [1,2]. Visual tests can be useful for measuring memory because they are not vulnerable to the effects of hearing loss. On the other hand, auditory speech tests may be more ecologically relevant for audiological rehabilitation than reading working memory measures because speech understanding involves auditory and cognitive processing [3]. In addition, a test that yields a wider range of scores may be more useful to audiologists because there is greater potential for distinguishing the abilities of listeners. In a previous study, a listening memory test using words yielded a greater range of working memory scores than a test based on reading sentences [3]. However, because the linguistic properties of the materials were not matched across tests, the difference in the range of scores could be attributed to either modality or linguistic factors.

In the current study, we compared auditory and visual tests with matched word-level and sentence-level materials in order to tease apart the effects of modality and linguistic factors on recall scores. The results of the present study will inform the ongoing design of clinically feasible tests of working memory for use in audiology.

2 Method

2.1 Participants

Participants were 32 young adults (27 female, 5 male) in good or excellent health, who were native English speakers (mean age = 19.9 years, *SD* = 1.8, range 18-26) and had at least some post-secondary education (mean years of education = 14.1, *SD* = 1.3). All participants had normal hearing thresholds (≤ 25 dB HL) from 250-8000 Hz, except for two participants who had thresholds of 30 dB HL at 8000 Hz in the left ear.

2.2 Materials

The complex auditory and visual stimuli were sentences taken from the Revised Speech Perception in Noise (R-SPIN) Test materials [4]. The simple auditory and visual stimuli were taken from the Word Auditory Recognition and Recall Measure (WARRM) materials [5].

2.3 Procedures

Participants completed each of the four tests (2 modalities x 2 linguistic levels): simple (word-level materials) auditory, complex (sentence-level materials) auditory, simple reading, complex reading. In simple conditions, participants repeated a target word introduced by a carrier phrase. In complex conditions, every target word was introduced by a different sentence. All auditory conditions were completed in quiet. The order of conditions was counter-balanced. In each test, 100 items were presented, with five trials in each of five setsizes (2, 3, 4, 5, 6). The number of words correctly recognized, judged and recalled was measured (similar to the protocol of Daneman & Carpenter [6,7]).

3 Results

There was a significant main effect of linguistic complexity ($F(1,31) = 81.0$, $p < .001$, $\eta p2 = .72$), with higher recall for words than sentences. There was also a main effect of modality ($F(1,31) = 57.1$, $p < .001$, $\eta p2 = .65$), with higher recall for auditory than visual stimuli. In addition, there was a main effect of setsize, ($F(4,124) = 336.5$, $p < .001$, $\eta p2 = .92$), with recall decreasing with increasing setsize. There were significant interactions between linguistic complexity and setsize ($F(4,124) = 9.3$, $p < .001$, $\eta p2 = .23$) and modality and setsize ($F(4, 124) = 14.4$, $p < .001$, $\eta p2 = .32$). As setsize increased, the differences in recall due to complexity and modality became more pronounced (see Figure 1). There was no three-way interaction. We conducted a Bonferroni post-hoc, corrected for multiple comparisons. 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 8.1%. Collapsed across linguistic complexity, recall was better for auditory than for visual materials by 9.1%.

* jenna.pattison@mail.utoronto.ca
† k.pichora-fuller@utoronto.ca
‡ Sherri,Smith@va.gov

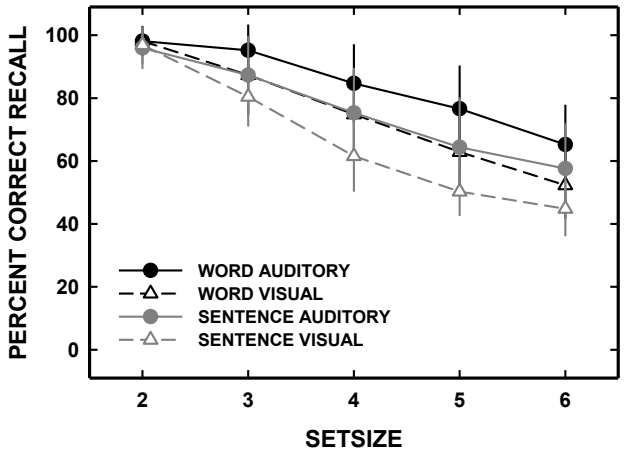


Figure 1: Percent correct recall by setsize for target words in word-level and sentence-level materials during listening and reading tests. Error bars represent SDs.

Using the total scores, recall was better for word- than for sentence-level materials by 8.9% and recall was better for auditory than for visual materials by 10.8%. There was a significant main effect of linguistic complexity ($F(1,31)$ = 83.1, p <.001, ηp2 = .73) and modality ($F(1,31)$ = 67.2, p <.001, ηp2 = .68), but no significant interaction (see Figure 2).

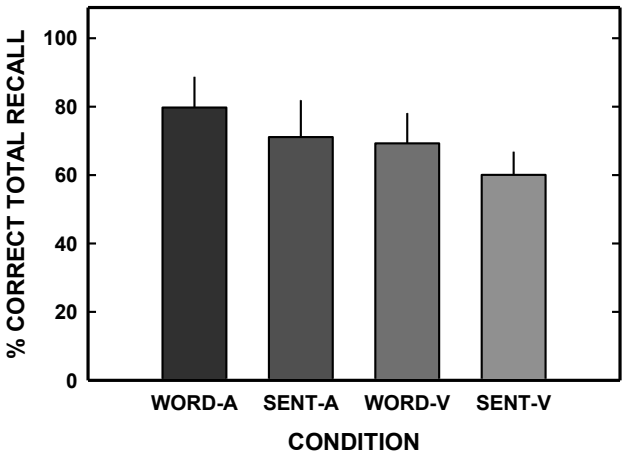


Figure 2: Percent correct total recall for target words in word-level and sentence-level materials during listening and reading tests. Error bars represent SDs.

4 Discussion

Our results clarify the findings of Smith and Pichora-Fuller (2015) by showing that auditory stimuli are easier to recall than visual stimuli even when the linguistic complexity of the stimuli is controlled. Ruchkin et al. [8] suggested 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. Another interpretation is that reading is cognitively more demanding than listening.

There was a significant correlation between modalities for the sentence-level conditions (r = .39, p = .03), and a stronger correlation for word-level materials (r = .58, p < .001). Possibly the lexical cues are more common across modalities, whereas the modality-specific aspects of more complex sentence-level processing differ to a greater extent. As in the previous study, the range of scores was greater for the easiest test of listening to words than for the hardest test of reading sentences; however, both modality and linguistic complexity contributed to the range in scores.

5 Conclusion

These findings may or may not apply to older individuals or people with hearing loss or vision loss. Further research will have to be conducted in order to generalize these findings.

Future research also could explore if the listening memory test is sensitive to intra-individual differences in performance to varying listening condition (e.g., listening is aided vs. unaided or in quiet vs. in noise).

References

[1] Pichora-Fuller, M.K., Schneider, B.A., & Daneman, M. (1995). How young and old adults listen to and remember speech in noise. *Journal of the Acoustical Society of America, 97*(1), 593 608.doi:http://dx.doi.org.myaccess.library.utoronto.ca/10.1121/1.412282

[2] Baldwin, C.L. & Ash, I.K. (2011). Impact of sensory acuity on auditory working memory span in young and older adults. *Psychology and Aging, 26*(1), 85-91. doi: 10.1037/a0020360

[3] Smith, S. L., & Pichora-Fuller, M. K. (2015). Associations between speech understanding and auditory and visual tests of verbal working memory: effects of linguistic complexity, task, age, and hearing loss. *Frontiers in Psychology (Auditory Cognitive Neuroscience Section), 6,* 1394. doi: 10.3389/fpsyg.2015.01394

[4] Bilger, R.C., Neutzel, M.J., Rabinowitz, W.M., Rzeczkowski, C. (1984). Standardization of a test of speech perception in noise. *Journal of Speech and Hearing Research, 27(1)*, 32- 48. doi: 10.1121/1.2017541

[5] Smith, S., Pichora-Fuller, M.K., & Alexander, G. (2016). Development of the Word Auditory Recognition and Recall Measure (WARRM): A working memory test for use in rehabilitative audiology. *Ear and Hearing.* PAP July 19, 2016. doi: 10.1097/AUD.0000000000000329

[6] Daneman, M., & Carpenter, P.A. (1980). Individual differences in working memory and reading. *Journal of Verbal Learning and Verbal Behaviour, 19(4)*, 450-466. doi:10.1016/S0022-5371(80)90312-6

[7] Daneman, M. & Carpenter, P.A. (1983). Individual differences in integrating information between and within sentences. *Journal of Experimental Psychology: Learning, Memory, and Cognition, 9(4*), 561-584. doi: http://dx.doi.org/10.1037/0278-7393.9.4.561

[8] Ruchkin, D.S., Berndt, R.S., Johnson, R., Ritter, W., Grafman, J., & Canoune, H.L. (1997). Modality-specific processing streams in verbal working memory: evidence from spatio- temporal patterns of brain activity. *Cognitive Brain Research, 6(2*), 95-113. doi: 10/1016/S0926-6410(97)00021-9

Laboratory Assessment Of Daily-Life Speech Understanding

Sridhar Kalluri, Jing Xia, Ervin Hafter

Current laboratory tests of speech understanding that are in common use in audiology do not incorporate important elements of daily listening that engage the cognitive elements of listening (e.g., attention, working memory). Given that cognitive processing takes on a particularly important role in sub-optimal listening scenarios, it is not surprising that traditional speech tests have had limited success in accounting for the complexities of daily listening caused by complex acoustic environments, listening goals that may change from moment to moment, hearing impairment, distortions caused by signal processing, and the auditory processing effects of aging. With the goal of developing better tools for assessing the impact of hearing impairment and sensory and cognitive interventions (hearing technology, auditory and cognitive training) on speech communication, we are developing a new speech test and measurement paradigm that incorporates important elements of daily listening that engage cognitive aspects of auditory processing. The new test focuses on understanding the meaning of speech rather than just hearing and reporting the phonetic elements of speech. It also incorporates important demands of daily listening such as the need for listeners to operate within the context of a continuous flow of speech information and with the presence of competing speech. This presentation will review how the test is constructed to achieve these elements of listening, and what impact these elements have on performance when compared to traditional tests of speech reception. Such research should lead to tests that are better than traditional tests at predicting the effects of treatments on real-life experiences of listening-instrument users. As such, they may form a part of a toolkit that clinicians deploy for determining intervention plans for their patients.

SUSTAINABLE BUILDING ACOUSTICS - ACOUSTIQUE DU BATIMENT DURABLE

Investigation Of Airflow And Sound Transmission Through Vented-Acoustical Panels For Natural Ventilation
James Higgins, Vivek Shankar, Murray Hodgson 238

Acoustical Investigation Of Laneway House Facades
Rosa Lin, Maureen Connelly 240

Increasing Sound Insulation Of Window With Add-On Systems & Various Insulating Glass Unit Configurations
Joyce Mak, Maureen Connelly 242

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 244

INVESTIGATION OF AIRFLOW AND SOUND TRANSMISSION THROUGH VENTED-ACOUSTICAL PANELS FOR NATURAL VENTILATION

James Higgins [*1], Vivek Shankar [†2], and Murray Hodgson [‡2]
[1]VanAir Design Inc.
[2]University of British Columbia, Acoustics and Noise Research Group

1 Introduction

Commonly, airflow pathways through partitions are simple holes partially blocked by a louver or grille. However, by diverting airflow through a partition, there is an opportunity to silence noise transmission. This work studies airflow and sound transmission through vented thin panel partitions. The focus application was ventilation openings for naturally ventilated buildings. As such, low pressure and low velocity airflow were investigated in detail.

The research was conducted in labs built for directly testing acoustical transmission loss and airflow on the same specimen installation, in full-size. First, four simple partition channel shapes were tested and evaluated; they were named 'L', 'U', 'straight', and 'Z' and their cross sections are shown in Figure 1. These tests provided justification for continuing detailed research into the Z shape. By varying the dimensions of the Z channel and the arrangement of sound absorbing materials, many permutations were tested. The performance data was then organized into a test inventory spreadsheet, which can be used to design a vented partition for size, sound transmission, and ventilation requirements.

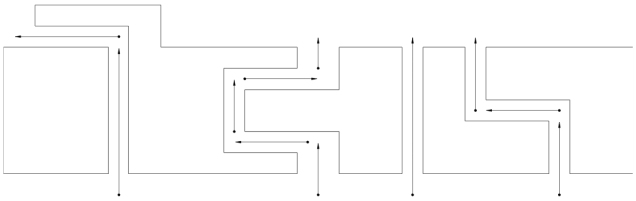


Figure 1: Section view of channel shapes L, U, Straight, and Z.

The wings of the Center for Interactive Research on Sustainability (CIRS) building are naturally and mechanically ventilated, and therefore very low pressures drive airflow. In the original design, large open areas were left above the movable wall partitions for airflow, but these openings transmit sound which causes a distracting work environment. To increase the transmission loss of the partitions while allowing for adequate ventilation, the operating conditions of these wings were studied, and six finish-quality vented partitions were designed and installed in the openings of four meeting rooms. Finally, in situ measurements were taken to find the installed performance of the vented partitions and the new operating conditions of the meeting rooms.

[*] james@vanairdesign.com
[†] vivekvs1@gmail.com
[‡] hodgson@mech.ubc.ca

2 Lab Experiments and Characterizations

The first part of this work was centered on the laboratory testing of channel configurations for sound transmission loss and airflow performance. The laboratories used have two active test rooms and a test partition between them, in which the sample is mounted. Both ventilation and acoustic test procedures were adapted from ASTM standards (ASTM E779-10 'Standard Test Method for Determining Air Leakage Rate by Fan Pressurization' [1] and ASTM E336-11 'Standard Test Method for Measurement of Airborne Sound Attenuation between Rooms in Buildings' [2]).

2.1 Ventilation Experiments

Using a calibrated measurement blower fan and differential pressure gauges, one room is pressurized. The pressure across the test partition and the flow through the fan were read from the gauges. Measurements of steady-state flow were taken at a number of fan speeds for pressures from 0.2 to 50 Pa. The leakage for each lab's pressurized source room was also measured across this range, and the leakage flow is subtracted from the flow through the fan to get the flow through the test sample.

The flow is characterized by its equivalent open area for flow, EOA_f. This compares the flow through a ventilator to the flow through a hole in flat plate. For each measurement point, this is done by comparing the sample's discharge coefficient to the discharge coefficient of an opening in a flat plate of 0.61:

$$EOA_f = \frac{Q}{0.61}\sqrt{\frac{\rho}{2\Delta P}}$$

2.2 Acoustical Experiments

To measure the acoustical transmission loss (TL), white noise was sounded in the source room and the noise reduction across the partition, as well as receiver room reverberation time, were measured in third octave bands at multiple points in the rooms. Frequency dependant transmission loss is calculated using averages as:

$$TL = Leq_s - Leq_r + 10\log\left(\frac{A_v T_{60}}{0.161 V_r}\right)$$

where Leq_s is the sound level in the source room, Leq_r is the sound level in the receiver room, A_v is the area of the sample, T_{60} is the reverberation time, and V_r is the volume of the receiver room.

Sound Transmission Class (STC) was used as a single number characterization for the TL curve, and is computed as in ASTM E 413-10, Classification for Rating Sound Insulation [3].

The equivalent open area characterization can also be adopted for speech sound transmission. The difference of a

normalized speech spectrum and the TL spectrum is summed logarithmically from 100 to 10,000 Hz, then multiplied by the area of the sample to get:

$$EOA_{s_speech} = A_v \sum_{f=100}^{10,000} 10^{(L_{speech}-TL)/10}$$

3 Evaluation of Simple Channel Shapes

Four channel shapes were evaluated, 'L', 'U', 'straight', and 'Z'. The dimensions of each channel were chosen to provide the best performance comparison and to fit applicable architectural size constraints. For each configuration, the channel depth was 4" and, except for the Straight channel, all spanned 20" of the partitions width. There were no acoustical linings applied to the channels.

Table 1 gives single number ratings for the airflow and acoustical performance. For both airflow and acoustical transmission loss, the L, U, and Z shapes are in the same range of performance. The Straight channel has a much higher airflow capacity but much lower transmission loss.

Table 1: Single number performance ratings for 4 channel shapes.

Channel Shape	EOAf	STC
L	210	7.4
U	238	9.1
Z	207	8.6
Straight	415	-1.7

Due to the geometry of a flat partition, the Z shape was chosen as the primary channel to study. For the same partition thickness, the U shape must have half the channel width of the Z and L shape. So, by applying a partition thickness constraint, the airflow capacity of the U shape would be reduced to possibly half of the tested value above.

4 Iterative Testing of Z-Channels

Many permutations of the Z-channel were systematically tested in the laboratory, varying channel dimensions and the placement and material of acoustical linings, so results could be used to design and predict the performance of vented panels for the acoustical, ventilation, and architectural requirements of any application, including CIRS.

Each Z-channel configuration was described by seven parameters, and the tested values are listed for each paramter are listed in Table 2. Figure 2 shows a section view of the Z-channel and labels the parametric dimensions and absorption placement; dimension H extends into/out of the page.

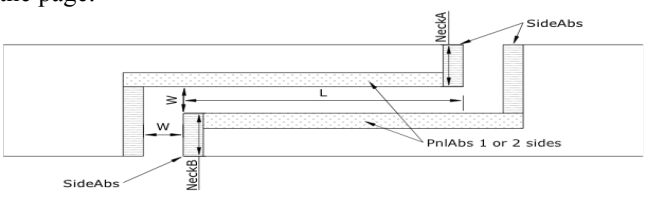


Figure 2: Parametric dimensions and absorption for the Z-channel, H extends into/out of the page.

Table 2: Lists of values tested for dimensions and absorption.

W [in]	H [in]	L [in]	NeckA [in]	NeckB [in]	Fibreglass Absorption *Panel*	Fibreglass Absorption *Sides*
1.5	22	6	0.75	0.75	None	None
2	44	11.75	1.25	1.25	Only 1 panel	1" thickness
2.5	72	18	1.75	1.75	Both panels	
3		28.75	2.25	2.25	1/2" thickness	
4			4.25	4.25	1" thickness	
6			5.25	5.25		

In total, 45 channel configurations were tested. The single number characterization results (STC, EOA_{S_Speech}, and EOA_f) from these tests were compiled into a test inventory spreadsheet that lets the user easily compare the performance of channel configurations. Some general findings were:

- long necks produce good sound absorption and airflow
- bends after a short length cause poor pressure relief and have high flow resistance
- airflow scales linearly with H but not with others
- transmission loss is independent of H
- cotton lined fiberglass (Knauff Plenum Board) outperforms high density acoustical foam

5 Ventilator Panel Design for CIRS

The final phase of this work was to design, build, install, and test vented panel partitions at CIRS. By investigating the building's operating conditions and using this study's laboratory results, six panels for four meeting rooms were designed to improve sound isolation and provide adequate ventilation. CIRS operates by a hybrid of mechanical displacement and natural cross-ventilation strategies, both at very low pressures. Using pressures estimated from tracer gas experiments and minimum ventilation rates dictated by ASHRAE, a requirement was set for EOA_f. Given the area of the openings at CIRS, the channel dimensions were chosen to meet this requirement, and provide STC ratings similar to the rest of the existing partition.

6 Conclusion

Through iterative lab testing, performance ranges were determined for innovative panels that provide silenced airflow. When designing a Z channel vented-acoustical panel, the test inventory spreadsheet is a useful reference tool for performance data and ranking dimensional choices.

References

[1] Standard test method for determining air leakage rate by fan pressurization, ASTM Standard E779-10, 2010.

[2] Standard test method for laboratory measurement of airborne sound transmission loss of building partitions and elements, ASTM E90-11, 2011.

[3] Classification for Rating Sound Insulation, ASTM E413-10, 2010.

ACOUSTICAL INVESTIGATION OF LANEWAY HOUSE FACADES

Rosa Lin[1] and Maureen Connelly[2]
Centre for Architectural Ecology, British Columbia Institute of Technology
NE03 - 3700 Willingdon Avenue, Burnaby, BC, V5G 3H2, Canada

1. INTRODUCTION

Façade transmission loss (TL) of six laneway houses (LWH) in Vancouver were investigated through measurement and modelling. Laneway houses are a popular type of small residence [3] that increase housing stock and effective density throughout Vancouver's single family residential zones. Noise levels in Vancouver's residential laneway environment may be high, exceeding 65LAeq [1]. LWH have many size-, mass- and design-related disadvantages that require acoustical attention.

Design-related challenges include limited distance from noise sources, relatively large glazing-to-façade and façade-to-floor-area ratios, and small interior volumes. Common design in the LWH include: non-absorptive interior surfaces and minimal furnishing, open-plan living spaces, double-height ceilings, and outdoor-oriented design with generous use of operable glazing components and patio space. All wall assemblies evaluated are wood-frame construction, with common cladding systems (wood, fibre cement planks, and stucco). A summary of the case studies' characteristics is presented in Table 1. Case studies B, C and D were fully-furnished at time of investigation.

Despite high-risk noise concerns, municipal LWH design publications do not address exterior noise nor specify care in locating noise-sensitive spaces, and suggest window placement and setback limits incompatible with sound design principles outlined in Vancouver's Noise Control Manual. Thus, LWH may have insufficient exterior noise insulation for residential health and comfort, warranting this investigation, in which assessment tools were also explored.

2. METHOD

Façade TL data and performance metrics of the case studies were calculated from measured and modelled data. In-situ measurements followed the ASTM E966 and C423 standards and the modelling tools used were the Canadian Housing and Mortgage Corporation (CMHC) traffic noise model [2] and the LWH model. The CMHC traffic noise model uniquely predicts road traffic noise levels at the exterior of each LWH façade and the required façade sound insulation for each designated space to meet residential room criteria, and includes room and construction considerations.

The LWH model is the numerical area-weighted composite TL of a multi-component façade. The wall portion was modelled in AFMG's Soundflow software based on known or best-estimated assembly construction details. Standard input values include material properties, dimensions, and layer sequence. Lateral airgaps through a typical cladding type, such as lap joints of horizontal fibre cement plank siding, were accounted for. Adding miniscule air gaps between each layer of solid panel material as would exist in real construction improved Soundflow output conformance to laboratory-measured data of similar wall construction. Laboratory test data from the NRC [3] was used for fenestration units.

Results from all methods were evaluated against CMHC criteria and analyzed. Procedures in the CMHC, ASTM E1332 and ASTM E413 were used to calculate the insulation rating metrics of acoustical insulation factor (AIF), outdoor-indoor transmission class (OITC) and sound transmission class (STC).

3. RESULTS

3.1 CMHC criteria and CMHC model results

Four out of the six LWH were estimated to have excessive road noise exposure up to 64LAeq (Table 1). LWH A and B were estimated to each have three of four sides exposed to road traffic noise above 55LAeq. Façades B, C and E have lower criteria than the others (Figure 1). A has the highest insulation requirement due to the highest outdoor level and most sensitive indoor use. B has the lowest criteria due to lower outdoor level and a kitchen interior more forgiving of noise. A, C and E did not meet component AIF requirements.

Table 1: Case studies noise exposure, facade and room information

Case Study ID	A	B	C	D	E	F
Estimated noise exposure (LAeq)	61	54-59	50	54	54	44
Test facade size (m^2)	7.6	7.8	28.8	13.9	10.1	20.0
Fenestration/facade ratio	12%	24%	22%	0%	46%	19%
Test room floor area (m^2) (approx. ±1 m^2)	7.2	19.0	30.0	20.7	19.5	27.0
Façade/floor area ratio	106%	41%	96%	67%	52%	74%
Test room volume (m^3) (approx. ±2.5m^3)	17.5	42.9	122.0	44.4	65.1	57.5
Test room use	S/B	LK	LDK	LD	LDK	LDK
Room condition	E	F	F	F	E	E
Architectural features		O	OH^2		O H^2LB	OH^2
Room absorption @500Hz (Sabines, m^2)	2.8	14.3	30.4	13.6	12.8	12.9

S=study, B=bedroom, K=kitchen, D=dining, L=living room; E=empty, F=furnished, O=open plan, H^2=double height ceiling, LB=lofted open bedroom

3.2 Empirical results

Façade TL ratings calculated from measured 1/3 octave band TL data indicate that half of the case studies, A, E, and F, did not meet criteria, and that C only marginally passed (Figure 1). STC values range from 26 to 35 (Figure 1) and OITC values range from 24 to 30 (Figure 2). B and D are the highest performers in terms of STC at 35 and 36,

[1] rlin35@bcit.ca; rosa.lin@cal.berkeley.edu
[2] Maureen_Connelly@bcit.ca
[3] Under 900sq.ft.(84m^2), averaging 650sq.ft.(60m^2), under 1.5 stories.

respectively. In terms of OITC, B is highest at 30; A, C, D and F perform similarly at 26 to 27.

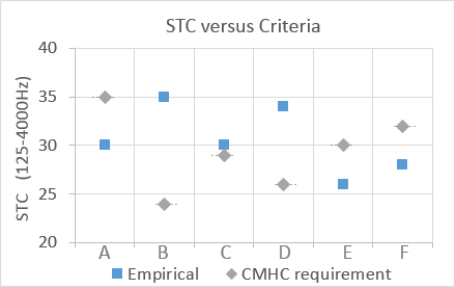


Figure 1. STC (field) versus criteria

3.3 LWH model

Outdoor-indoor TL (OITL) by the LWH model was compared to empirical data. In B, modelled façade data closely aligns with field-measured data from 125Hz to about 2000Hz (Figure 2). In the other five of six case studies, conformance between field and model data is seen in the frequency range of 200Hz to 1250Hz. The largest discrepancies between the LWH model and field-measured data are found in the higher frequencies around 1600Hz and above.

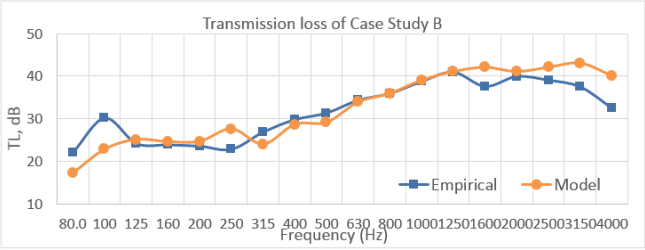


Figure 2: Modelled vs. measured OITL

Figure 3 compares field to model OITC for each test façade. Conformance between field and model values is seen in two cases studies, B and C. In the other four of the six case studies, modelled OITC values are higher than measured by 3 points.

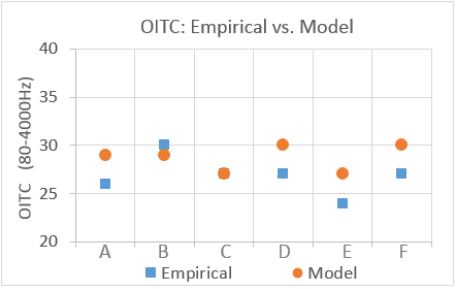


Figure 3: OITC: measured vs modelled

4 DISCUSSION

Empirical and model TL data both indicate that the case study LWH need acoustical improvement, especially in A, E and F. Façade insulation ratings are low across the board, and indoor absorption (Table 1) may be too low in many cases to reduce indoor levels. Based on outdoor noise level estimated by CMHC, A, B and E have outdoor living areas that require additional noise protection.

CMHC-estimated component AIF deficiencies are attributed to, in A: the highest façade/floor area ratio, and the highest outdoor levels coupled with the most sensitive space; in C and E: extensive use of glazed doors.

The results reflect a correlation between architectural characteristics of the LWH and lower acoustical insulation. A, having the highest facade/floor ratio at 106%, has the second lowest measured OITC of 26; B, having the lowest façade/floor area ratio at 41%, has the highest measured insulation ratings OITC 30 and STC 35. E, having the highest fenestration/facade ratio at 46%, has the lowest measured facade insulation of OITC 24 and STC 26. D, with the lowest fenestration/façade ratio at 0%, has the highest model OITC 35 and the second highest empirical STC 34.

In all cases, the LWH model, particularly in Soundflow data, estimated significantly higher TL around frequencies of 1600Hz and above than in measured data. This may be due to additional air leaks existing in real construction, such as at inaccessible details like joints, nails and screws, and service penetrations. Other general discrepancies between model and measured data are primarily attributed to flanking, as in the case of E, where the room has a source-facing roof/ceiling, an adjacent metal garage door, and an adjoining wall with a window near the test façade.

Although the LWH model produced some data that closely approximated measured data, in most cases it overestimated façade performance and thus should not be relied upon for criteria compliance without improvement or additional consideration for discrepancies.

5 CONCLUSION

This case study investigation of Vancouver LWH revealed excessive outdoor noise levels, insufficient façade insulation, and limited indoor absorption. A number of case studies demonstrated a strong correlation between LWH characteristics and low façade insulation. Acoustical education and regulatory drivers are called for in the design and construction of LWH to protect resident health and comfort. Tools to aid this process, such as the CMHC and the LWH models, should be further explored. LWH exposed to high traffic noise need acoustical improvement. Practical steps include installing acoustical fenestration (including skylights), treating air leaks, adding sound-attenuating façade systems like brick cladding or living walls, and taking care in designing sleeping areas and open plans to protect sensitive spaces from noise.

6 REFERENCES

[1] Lin, “Sound Living in Vancouver’s Laneway Housing” (Master of Applied Science thesis), BCIT, 2014.
[2] Canadian Mortgage and Housing Corporation, “Road and Rail Noise: Effects on Housing,” 1981.
[3] Bradley & Birta, “Laboratory Measrements of the Sound Insulation of Building Façade Elements,” IRC/NRC, 2000.

INCREASING SOUND ISOLATION OF WINDOW WITH ADD-ON SYSTEMS & VARIOUS INSULATING GLASS UNIT CONFIGURATIONS

Joyce Mak [*1] and Maureen Connelly [†2]

School of Construction and the Environment, British Columbia Institute of Technology, Burnaby, BC V5G 3H2 Canada

1 Introduction

Windows represent the most significant weak noise isolation component of the building envelope. Although it is more efficient and cost effective to incorporate acoustical consideration during the design and construction process, remedial work to the window may be required to improve the acoustic comfort in existing residential buildings. In situation where the need to increase transmission loss of the existing windows is required, commercially available add-on systems can be incorporated to either the exterior or interior side of the windows to increase sound isolation. This research evaluates the potential improvement of incorporating five different types of add-on systems to a typical residential window to reduce outdoor traffic noise into the interior space.

The five add-on systems can be subcategorized by installation location: exterior and interior. Exterior add-on systems include an exterior roller shutter and an exterior shutter. Where it is not desirable or feasible to alter the building's exterior aesthetic, especially in multi-unit residence or at heritage or historic buildings, interior add-on systems may be required as alternative solutions. Interior add-on systems investigated include a high-density closed-cell (HDCC) foam window plug, a mass loaded vinyl curtain, and an interior acrylic glazed window.

2 Methodology

Sound intensity measurements were conducted to determine the in-situ sound insulation of the window assembly plus the add-on systems. All sound transmission measurements were made in accordance with ISO 15186-2 "Acoustics – Measurement of Sound Insulation in Buildings and of Building Elements Using Sound Intensity – Part 2 Field Measurements". Measurements were made between 100 to 5000 Hz at one-third octave intervals.

[*]jmak.joyce@gmail.ca
[†]Maureen_Connelly@bcit.ca

To study the potential acoustic benefits of add-on systems on windows, baseline measurements of the installed window with various IGU configurations were conducted. The thicknesses of the double and triple glazing used in the tests are as follow: 4(22)6L, 4(16)4 and 6L(11)5(11)6L. The value between parentheses denotes the depth of air-filled glazing space and the thickness of the glass panes, in mm. The type of glass used in the tests is clear float glass. The letter "L" is used to represent laminated safety glass.

A base tilt-and-turn window with unplasticised Polyvinyl Chloride (uPVC) frame was mounted in a high-mass wood-framed wall panel. The window was installed as per a typical residential window installation with sealed joints around the window perimeters. The high-mass wall panel was 82" high and 70" width. The window rough opening was 48" high, 36" wide and 7" deep. The window sill was at a height of 20" above the floor of the testing facility.

3 Results & Discussion

A total of fifteen field sound transmission loss tests were conducted, including three baseline IGU window tests. The interior storm window was tested at depth of 3" and 6" from the interior face of glazing.

The baseline IGU window test results are consistent with trends from published transmission loss data. As expected, when comparing between the two double-pane IGUs, the 4(22)6L IGU exhibits consistently higher transmission loss across all frequencies. The triple glazed IGU has acoustic advantages over the 4(16)4 IGU above 160 Hz. However, the triple pane has almost identical transmission loss data to the 4(22)6L IGU, only showing marginal benefits around 200 to 315 Hz and above 3.15 kHz. Thus, the add-on systems window was studied only on the two double-paned IGU configurations.

3.1 Exterior Add-on Systems

In the case of 4(22)6L IGU, the two exterior add-on systems transmission loss curves are almost identical,

providing increase of sound transmission loss at 700 Hz and above. The resonance effect is emphasized more with the exterior shutters than with the exterior roller shutters at 125 Hz. Above 1000 Hz, the exterior roller shutters exhibit slightly higher sound transmission loss by 5 dB.

In the case of 4(16)4 IGU, the exterior shutter shows higher overall performance than the exterior roller shutters. The difference is most prominent at the frequency range from 200 to 800 Hz. At 250 Hz, the exterior shutters illustrate higher sound transmission loss of 15 dB than the exterior roller shutters. At higher frequencies, the exterior shutters demonstrate slightly enhanced acoustic performance.

3.2 Interior Add-on Systems

In the case of 4(22)6L IGU, the vinyl curtain provides the highest acoustic performance when compared amongst the rest of the interior add-on systems. The vinyl curtain and interior storm window have similar transmission loss characteristics across all frequencies. The HDCC foam has the least sound insulation benefits and shows a pronounce resonance dip at 125 Hz. All three interior systems provide enhanced acoustic performance beyond 1.25k Hz.

In the case of 4(16)4 IGU, the interior storm window offers the highest overall acoustic performance when compared to the remainder of the interior systems. While the vinyl curtain is comparable to the interior storm window at 125 to 400 Hz, the HDCC foam is comparable to the interior storm window at 1.25 kHz and above.

In general, when add-on systems are installed on either side of the double pane windows, the entire assembly may be considered a triple pane IGU system with an additional air chamber created. Mass-air-mass-air-mass resonance may be found at low frequencies. Exception should be given to the HDCC foam; since the HDCC foam is non-rigid and is installed directly against the window, the air layer is not created.

4 Conclusions

An upward transmission loss trend is clear for all add-on systems against the baseline IGUs. Acoustic benefits of add-on systems are most significant on the low acoustic performance double pane IGU with similar glass thickness, as shown in Figure 1. Exterior shutters have better acoustic performance than exterior roller shutters. Interior storm window provides appreciable noise reduction when compared amongst other interior add-on systems.

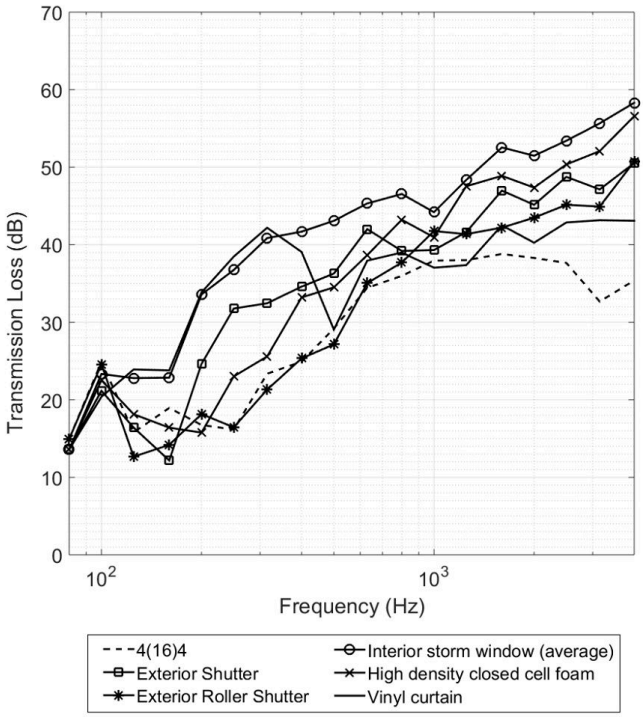


Figure 1: Sound transmission loss data for exterior and interior add-on systems against double pane 4(16)4 IGU

The STC and OITC ratings do not adequately numerate the performance of add-on systems, especially when noise reduction is dependent on the source spectrum and that traffic noise is low frequencies dominant.

References

[1] Diaz, C., & Pedrero, A. An experimental study on the effect of rolling shutters and shutter boxes on the airborne sound insulation of windows. *Applied Acoustics* , 369-377, 2007.

[2] Morin, M., & Marshall, S. Noise isolation provided by windows in residential projects. *Canada Mortgage and Housing Corporation*, 1997.

[3] Quirt, J. D. Measurements of the sound transmission loss of windows. *Building Research Note, 1981*.

[4] Tadeu, A. J., & Mateus, D. M. Sound transmission through single, double and triple glazing. Experimental evaluation. *Applied Acoustics , 62*, 307-325, 2001.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Sustainability In Acoustic Design

Forest Borch, Brigette Martin

Sustainability continues to be an important consideration for new building projects with incentives being offered from all levels of government. While the construction industry has made great advances when it comes to developing and improving sustainable methods of construction, acoustic consultancy has largely lagged behind. Acoustics has primarily been considered under the "social" pillar of sustainability rather than the "environmental" or "economic" pillars, with green building frameworks such as LEED awarding points for achieving specific internal noise levels. Therefore, when acoustics is discussed in conjunction with sustainability, the focus is typically on the conflict between simultaneously achieving acoustic criteria while implementing other important elements of the sustainable building design, such as passive ventilation systems. This paper looks at alternative ways to incorporate sustainability into acoustic design. It will provide an overview of current worldwide practices as well as investigate how acoustical consultants can use existing "green" certifications to provide more information relating to sustainability for acoustic design projects.

Sound Absorption And Scattering Properties Of Living Walls: Applications To Room Acoustics

Maureen Connelly, Daver Bolbolan, Mahsa Akbarnejad, Sepideh Daneshpanah

This work presents the research framework and the investigations of the acoustical characteristics of interior living walls and their impact on room acoustics. The overall goal is to determine how living walls can be used to positively benefit room acoustics by reducing reverberation and ambient noise, and affect speech intelligibility by creating sonic subzones within rooms. A method of evaluation at one-third scale was developed to measure the scattering effects of substrates and plant species (Bolbolan). A series of measurement were carried out to quantify the absorption of living walls with consideration of the parameters of carrier type, substrate, moisture content and plant species variants and to determine which parameters most significantly impact the absorption and scattering of living walls (Akbarnejad). The research on the verification of effect of interior living walls on room acoustics includes field measurement experiments of permutations of living walls installed in a 132 m2 theater designed for interactive workshops (Daneshpanah); classroom acoustic criteria such as background A-weighted noise, noise criteria (NC) curves, reverberation time, and speech intelligibility are measured and modelled for a 60 m2 seminar classroom with varying living wall installations. Measurement determined that the scattering coefficient of living wall substrate meets the ISO 17497-1 criteria for the turning table base plate. This allowed six living wall plant species, English Ivy, Fern, Golden Pothos, Pilea, Spider plant, Ficus pumila, to be further evaluated in substrate. The function of 'leaf area index X mass' was found to be more relevant than plant foliage characteristics (height, stem diameter, mass, leaf thickness, leaf length, leaf area) in predicting absorption; overall coverage is relevant to high-frequency scattering. Living walls reduced reverberation time and may affect speech intelligibility in close proximity to the living wall as measured and modelled in the classroom application.

Achieving Better Sound Insulation Of Clt Partitions

Ivan Sabourin, Christoph Hoeller, Berndt Zeitler

During the last 5 or more years, the direct airborne and impact sound insulation of over 100 different cross-laminated timber (CLT) floor and wall assemblies have been measured at the National Research

Council Canada. The installation of a lining such as a floor topping, a ceiling lining or a wall lining on CLT elements is often necessary to achieve good sound insulation. A large data set of airborne, impact and vibration test results of CLT assemblies with various linings were analyzed to determine sound insulation improvements and similarities. The airborne sound transmission loss was measured according to ASTM E90; the light impact sound insulation was measured according to ASTM E492; and the heavy impact sound insulation was measured using the heavy/soft impact source. The structural reverberation times and the wavenumbers of select CLT floor assemblies were also considered. As the 2015 edition of the National Building Code of Canada now has a requirement for the Apparent Sound Transmission Class (ASTC) which includes flanking sound transmission, the improvement of flanking sound insulation due to the addition of linings on CLT elements will be discussed.

Sound and Vibration Instrumentation

Scantek, Inc.



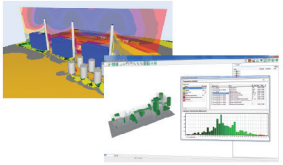
Sound Level Meters

Selection of sound level meters for simple noise level measurements or advanced acoustical analysis



Vibration Meters

Vibration meters for measuring overall vibration levels, simple to advanced FFT analysis and human exposure to vibration



Prediction Software

Software for prediction of environmental noise, building insulation and room acoustics using the latest standards



Building Acoustics

Systems for airborne sound transmission, impact insulation, STIPA, reverberation and other room acoustics measurements



Sound Localization

Near-field or far-field sound localization and identification using Norsonic's state of the art acoustic camera



Monitoring

Temporary or permanent remote monitoring of noise or vibration levels with notifications of exceeded limits

Scantek, Inc. Sales, Rental, Calibration

www.ScantekInc.com 800-224-3813

www.pcbpiezotronics.ca/acoustics(en-CA)

When You Need to Take a
Sound Measurement



PCB's Broad Range of Acoustic Products
High Quality ▪ Unbeatable Prices ▪ Fast Delivery ▪ Best Warranty

To Learn More Visit: www.pcbpiezotronics.ca/acoustics(en-CA)
Toll Free 888-424-0033 ▪ **E-mail** info@dalimar.ca

PCB PIEZOTRONICS INC. Dalimar instruments A PCB GROUP COMPANY LARSON DAVIS A PCB PIEZOTRONICS DIV.

Your acoustic measurement experts providing a full line of quality microphones, meters, and noise monitoring solutions

VIBROACOUSTICS - VIBROACOUSTIQUE

Understanding Sources And Sound Radiation Of A Snowmobile Track
Stéphane Beuvelet, Raymond Panneton, Alain Desrochers, Remy Oddo 248

Laboratory And Field Measurements Of Nail Guns' Noise Emission
Marc-André Gaudreau, Frédéric Laville, Pierre Marcotte, Jérôme Boutin 250

Comparison Of Two Methods Of Transfer Path Analysis Applied To Snowmobile For Noise Source Identification
Nassardin Nassar Guenfoud, Olivier Robin, Raymond Panneton, Alain Desrochers, Walid Belgacem 252

A Data Processing Method For Noise Measurements Of Snowmobiles And Their Sub-Systems On Test Bench
Jason Méjane, Raymond Panneton, Alain Desrochers, Rémy Oddo 254

Optimization Study And Panel Parameter Study For Noise Radiation Reduction Of An Aircraft Panel Excited By Turbulent Flow
Steven Albert James Sonnenberg, Joana Rocha 256

Abstracts for Presentations without Proceedings Paper - Résumés des communications sans article 258

UNDERSTANDING SOURCES AND SOUND RADIATION OF A SNOWMOBILE TRACK

Stéphane Beuvelet[*1], **Raymond Panneton**[†1], **Alain Desrochers**[‡1], **and Rémy Oddo**[§1]
[1]Department of Mechanical Engineering, GAUS, Université de Sherbrooke, Sherbrooke, Qc, Canada, J1K 2R1.

1 Introduction

Maximum permissible sound levels are fixed by the Snowmobile Safety and Certification Committee standard. Classical acoustical solutions are difficult to use in snowmobile because of weight and space requirements. It requires understanding noise generation in order to act at the source. This paper investigates the noise emitted by this system.

Previous studies on snowmobile noise [1, 2] show that the system composed of the track, the suspension and the tunnel is deemed to be an important source. Moreover, the track noise is unpleasant [2]. Kleinendorst [2] have used Finite Element Analysis (FEA) to compute the stress transmitted to the tunnel by the impacts between the rods and guiding wheels. The noise was then reconstructed using measured Frequency Response Function between the tunnel and microphones. Results were however found to be inconsistent with the measurements.

Arz [3] shows that the noise of the system is generated at passing frequency of the track elements. From this, it was deduced that the noise was caused by the impacts between the fiberglass stiffener rods and other elements of the suspension. According to Kleinendorst [2] the noise is caused by the impact between the rods and the guiding wheels.

2 Method

2.1 Test bench description

Due to fast changing snow conditions, it is difficult to make repeatible track noise measurement [4]. To circumvent this problem, a test bench was developed. On this test bench, the track is driven by a silent electric motor and slips on a steel plate lubricated with water. The speed of the track is automatically controlled and the load applied on the suspension is controlled and continuously recorded. This test bench allows to isolate the noise of the track and to make repeatable measurements. Two successive tests show a maximum noise difference of only 1 dB.

2.2 Sources of vibration

There are many potential sources on the track, suspension and tunnel system. All of them are at the same frequency, which makes noise mapping difficult [5]. To understand which sources of vibration were important for noise radiation, 22 parameters were tested in two Plackett–Burman (P-B) design of experiments. P-B design allows obtaining the effect of many parameters with few tests. A variance analysis was also made to obtain the probability that the observed effects were indeed caused by the various design alternatives. By example, figure 1 shows the effect of parameter A and its degree of confidence in function of the speed of the track. It shows that above 80 km/h, parameter 1 will decrease the noise of almost 1 dB with a probability higher than 0.9. The efficiency of each parameter varies depending on the speed.

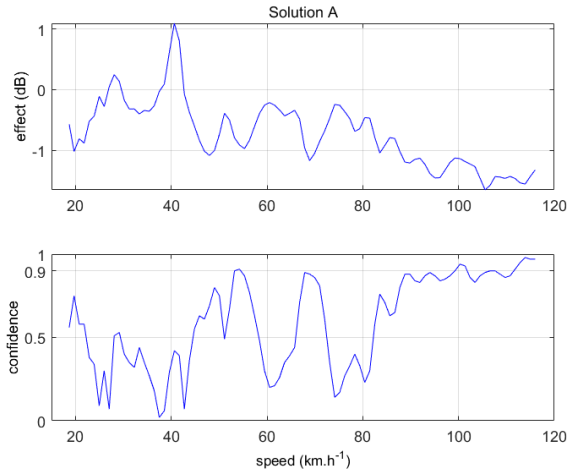


Figure 1: Effect of solution A.

2.3 Track and tunnel contribution

Mote [6] shows that the vibration of a moving continua becomes unstable as its speed and mass increase when compared to its tension. The track of the snowmobile has a high speed and mass and a low tension, so its vibrations can be nonlinear. High speed videos of the track were made between 20 to 120 km.h^{-1} but no instabilities have been reported.

A test bench has been designed to reproduce the environment and the tension of a track in use (figure 2). Impact testing were made on two track sections to understand their noise radiation in different tension conditions.

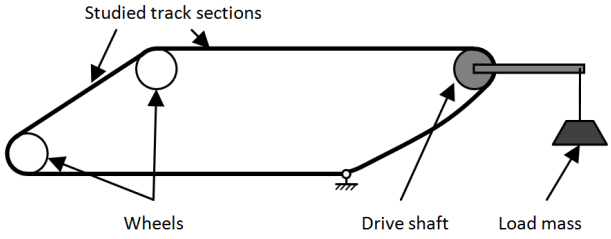


Figure 2: Static test bench

A rig for making tests without a tunnel was also fabricated. Hence, the noise of the system could be measured with and without a tunnel, so as to understand its contribution.

*Stephane.Beuvelet@USherbrooke.ca
†Raymond.Panneton@USherbrooke.ca
‡Alain.Desrochers@USherbrooke.ca
§Remy.Oddo@USherbrooke.ca

3 Results

3.1 The track generates vibrations, not noise

The most effective solutions were found to be those that decrease the elevation of the wheel when it passes over the lugs. This shows that this is the most important source in the system and it explains why the system is noisier on ice than on snow.

Static tests on tracks show that they radiate under the 1^{st} and 3^{rd} bending modes of the rods. Figure 3 shows the noise radiated by two tracks with different rod 1^{st} and 3^{rd} modes. There is no correlation between the rod modal frequencies and the radiated noise.

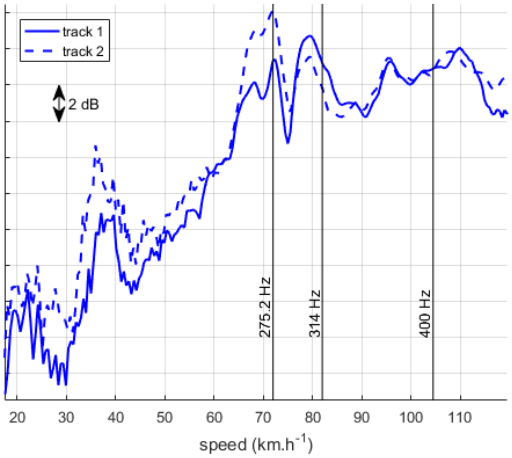


Figure 3: Noise of two tracks. Their third mode is respectively at 314 Hz and 400 Hz.

3.2 The noise is radiated by the suspension and the tunnel

Under 90 km.h^{-1}, the suspension without the tunnel made a noise of up to 5 dB less than that with the tunnel (figure 4). Above 90 km.h^{-1}, the noise of the two configurations is the same. It means that, in a standard configuration, the noise is radiated by the suspension under 90 km.h^{-1}, and by the tunnel after. This conclusion may change depending on the geometry of the tunnel and the suspension. Nassardin Guenfoud [7] made a Transfer Path Analysis improved of the system in order to understand how the energy is transmitted from the wheels to the radiating elements.

4 Conclusions

Owing to the presented work, the understanding of the noise generation by a snowmobile track and suspension has been improved. The most important excitation source was found to be the passage of the guiding wheels over the lugs. This excitation propagates in the suspension on which the tunnel of the snowmobile is attached. The sound radiation and the dynamics of the track have not been identified as important contributors in the system. Finally, depending on the track speed, the system has two preferred ways to generate noise:

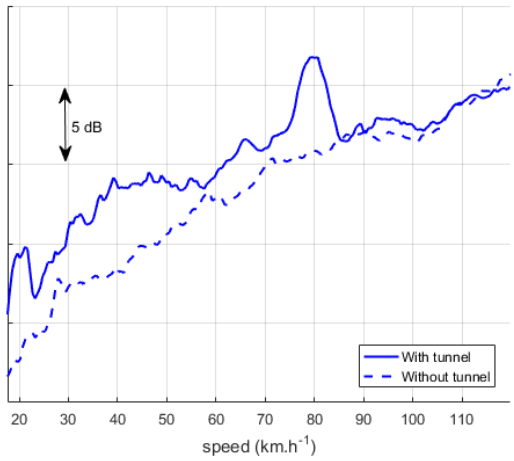


Figure 4: Noise of a track with and without tunnel.

airborne noise from the suspension itself and structural sound radiation from the tunnel.

Further studies are still needed to find out how the suspension and tunnel radiate noise.

Acknowledgments

This project is the result of a collaboration between BRP (Bombardier Recreational Products), the CTA (Centre de Technologies Avancées) and the GAUS (Groupe d'Acoustique de l'Université de Sherbrooke). This research was supported by BRP and NSERC (Natural Sciences and Engineering Research Council of Canada). The test bench was designed by Walid Belgacem research professional at Université de Sherbrooke.

References

[1] M.D. Osborne. Development of a Kit to Reduce to Noise Level of the MOST Vehicle. Technical report, Michigan Technological University, Houghton, 1991.

[2] J.L. Kleinendorst. *Understanding chassis inputs from the rear suspension of a snowmobile*. PhD thesis, Michigan Technological University, 2014.

[3] J.P. Arz. *Developpement d'une méthode de réduction du bruit d'impact des structures par ajout d'une couche mince d'élastomère*. Thèse de doctorat, École de Technologie Supérieure, 2010.

[4] B.J. Dilworth. *Identification of ground and environmental effects to the pass-by noise testing of snowmobiles*. PhD thesis, Michigan Technological University, 2009.

[5] L. Lefebvre. *Identification et caractérisation des sources de bruit impulsionnelles, répétitives et synchronisées, dans un système mécanique complexe*. PhD thesis, École de Technologie Supérieure, 2006.

[6] C.D. Mote. On the Nonlinear Oscillation of an Axially Moving String. *Journal of Applied Mechanics*, 33(2):463–465, 1966.

[7] N Guenfoud. Comparison of two methods of transfer path analysis applied to snowmobile for noise source identification. In *to be published at the Proceedings of the Acoustics Week in Canada*, Vancouver, 2016.

LABORATORY AND FIELD MEASUREMENTS OF NAIL GUNS' NOISE EMISSION

MESURES LABORATOIRE ET TERRAIN DE L'ÉMISSION SONORE DES CLOUEUSES

Marc-André Gaudreau[*1], Frédéric Laville[†1], Pierre Marcotte[‡2], Jérôme Boutin[2]**
[1] ÉTS, 1100 Notre-Dame Ouest, Montréal, QC, H3C 1K3, CANADA
[2] IRSST, 505 Boul. de Maisonneuve Ouest, Montréal, QC, H3A 3C2, CANADA

1 Introduction

Nail guns are heavily used in the construction industry by carpenters and roofers. These hand-held power tools expose their users to impulse noise and hand-arm vibration that are difficult to mitigate due to the user's proximity to the tool. Thus, the choice of low noise and low vibration power tools seems to be a good strategy in order to reduce workers' exposure to such noise and vibration. However, among the very few studies on the noise and vibration levels of nail guns found in the literature, only one made by the Health and Safety Executive in England is quite exhaustive. It is presented in two reports [1, 2]. In the present study, 10 nail guns were tested (8 framing and 2 roofing nailers). Both noise and vibration were studied, but only the noise aspect will be discussed in this paper (vibration results can be found in a paper presented at the 6th American conference on human vibration [3]). The goal of this paper is to present field measurements and laboratory measurements using EN 12549 [4].

2 Field Measurements

Six of the eight framing nailers were first tested on a construction site with two carpenters operating each of the six nailers for the construction of a building wall (see Fig. 1). The two roofing nailers were tested on asphalt shingle installation and were also operated by two roofers each (see Fig. 1).



Figure 1. Field measurement for framing nailers (left) and for roofing nailers (right).

The measurement rig consisted of a 6 input Wi-Fi B&K acquisition card that was relaying to a base PC: a triax 5000g accelerometer (i, ii, iii) fixed on the handle of the gun, the static pressure at the inlet of the gun (iv) and the acoustic pressure at the worker's ears (v, iv), via a microphone fixed to the center of each earmuff. The rig was held in a small comfortable backpack and even if the sensors were wired to the acquisition card, the worker could work without obstruction. Between 8 and 12 trials of 10 impacts were recorded for each nailer/worker combination.

3 Laboratory Measurements

Eight framing nailers and two roofing nailers were tested in the laboratory under controlled conditions as per the EN 12549 standard. From this standard, three operators were required to perform five trials of 10 nails each, with each trial lasting a period of 30 seconds (1 nail each 3 second). The nails were sunk in a standardized piece of wood (sawn pine wood, clear of knots) buried in a sandbox (see Fig. 2). The measures were performed in a semi anechoic room where both the sound power and the sound pressure level at the worker's ear were measured. The sound power was measured using a 9 microphones cubic grid as per ISO 3744 [5], embedded in EN 12549.



Figure 2. Laboratory measurement rig.

4 Results

For each measurement (both lab and field measurements), a one second equivalent sound pressure level (SPL), A-weighed, of a single impact was calculated using the following formula :

$$L_{pA,1s} = L_{pA_{eq}T} + 10\log_{10}\left(\frac{T}{T_0}\right) - 10\log_{10}(N) \quad \text{, Eq. 1}$$

[*] marc-andre.gaudreau@cegepdrummond.ca
[†] frederic.laville@etsmtl.ca
[‡] pierre.marcotte@irsst.qc.ca
[**] jerome.boutin@irsst.qc.ca

where $L_{pA_{eq}T}$ is the equivalent pressure level for a given period (T), T_0 is 1 second and N is the number of impacts during the period T. The sound power $L_{wA_{eq}T}$ (SWL) was only available in laboratory measurements and the equivalent acoustic power level, A-weighed, of a single impact $L_{wA,1s}$ was calculated using :

$$L_{wA,1s} = L_{wA_{eq}T} + 10\log_{10}\left(\frac{T}{T_0}\right) - 10\log_{10}(N) \quad \text{, Eq. 2}$$

Even if EN 12549 requires a microphone measurement at the average worker's position to measure $L_{pA,1s}$, the acoustic power $L_{wA,1s}$ was preferred to compare the noise level of the nailers since the position of the test subject and the position of the exhaust on each nailer were found to cause major uncertainties in $L_{pA,1s}$ values. In order to compare the laboratory and the field measurements, an estimation of the pressure level ($\overline{L_{p,A,1s}}$) at the ear of the subject, made out of the SWL and the distance (r) between the nailer and the ear of the subject is made using :

$$\overline{L_{p,A,1s}} = L_{w,A,1s} - 10\log_{10}(2\,\Pi\, r^2) \quad \text{, Eq. 3}$$

Table 1: Framing nailer ranking using Laboratory (left) or Field (right) measurements.

Laboratory measurements			
Rank	Tool	SWL dB(A)	estimated SPL dB(A)
#1	S6**	95.6	92.1
#2	R2	101.7	98.2
#3	S2	102.6	99.1
#4	S3	103.0	99.5
#5	R1	105.1	101.6
#6	S5*	107.1	103.6

Field measurements		
Rank	Tool	SPL dB(A)
#1	S6**	91.7
#2	R1	98.2
#3	R2	98.4
#4	S3	98.7
#5	S5*	99.7
#6	S2	100.1

Table 1 presents the laboratory ranking of the framing nailers with $L_{wA,1s}$ and its estimated sound pressure $\overline{L_{p,A,1s}}$, using r = 0.6m (see figure 1). The nailers R1 and R2 are using coils while nailers S1 to S6 are using strips (S1 and S4 are not presented as they were not evaluated at the workplace). All nailers were pneumatic, except nailer S5 (identified by *) which is gas operated and nailer S6 (identified by **) which is battery operated.

Table 2: Roofing nailer ranking using Laboratory (left) or Field (right) measurements.

Laboratory measurements			
Rank	Tool	SWL dB(A)	estimated SPL dB(A)
#1	B2	97.9	97.9
#2	B1	99.9	99.9

Field measurements		
Rank	Tool	SPL dB(A)
#1	B2	97.7
#2	B1	99.5

Table 2 presents the ranking of the 2 roofing nailers for both the laboratory and the field measurements (same ranking). The lab estimated SPL values were calculated using r = 0.4m (see figure 1). The 2 roofing nailers are labeled B1 and B2.

5 Discussion

Firstly, concerning the field results, since the lab bench prescribed in EN 12549 is designed to reduce the acoustic radiation from the nailed piece of wood, it was expected that field results would be higher than those measured in lab. This has not been validated by this study. For framing nailers, the field results are, on the contrary, lower by 3 dB or more than the estimated SPL ($\overline{L_{p,A,1s}}$) from the lab SWL. For the roofing nailers, the field results are at the same level as the lab estimated SPL. Secondly, concerning the lab and field rankings, they are very similar. Thirdly the battery operated nailer (S6**) is considerably less noisy than all other tested nailers in both lab and field measurements.

6 Conclusion

Concerning the EN 12549 standard, it seems appropriate in order to perform representative workplace ranking of nailers following their sound power level values. Concerning the reduction of workers' noise exposure, the battery operated nailer (S6**) stands out as the best choice as its level is at least 6 dBs lower than any other tested nailer in both lab and field measurements.

Acknowledgements

The authors are grateful to IRSST for providing the financial support of this research project.

References

[1] E. Shanks, *Noise emission from fastener driving tools*, HSE, England, (2008).

[2] R. Heaton et al., *Correlation between vibration emission and vibration during real use*, HSE, England, (2008).

[3] P. Marcotte et al., Vibration emission values of nail guns measured at the workplace and in the laboratory as by the ISO 8662-11, proceedings of the 6th ACHV, Miliwaukee, WI (2016)

[4] AFNOR. (2008). Acoustique -- Code d'essai acoustique pour les machines à enfoncer les fixations -- Méthode d'expertise. Association Française de Normalisation, NF EN 12549+A1:2008

[5] ISO 3744 (2010), Acoustics — Determination of sound power levels and sound energy levels of noise sources using sound pressure — Engineering methods for an essentially free field over a reflecting plane

COMPARISON OF TWO METHODS OF TRANSFER PATH ANALYSIS APPLIED TO SNOWMOBILE FOR NOISE SOURCE IDENTIFICATION

Nassardin Guenfoud [1], Olivier Robin [1], Raymond Panneton [1], Alain Desrochers [1], Walid Belgacem [2]
[1]Université de Sherbrooke, 2500 boul. de l'Université, Sherbrooke, J1K2R1, Québec, Canada
[2]Centre des Technologies Avancées, 3000 boul. de l'Université, Sherbrooke, J1K 0A5, Québec, Canada

1 Introduction

The problems related to snowmobiles noise are important issues for the recreational vehicle industry. When a snowmobile is operating, forces are injected in its rear suspension by the rotating track and will generate vibrations and also possibly acoustic radiation of the drive system (track - suspension - tunnel). To identify the related vibroacoustic transfer paths and main sources, the methodology of TPA (Transfer Path Analysis) and OTPA (Operational Transfer Path Analysis) were used. A comparison between these two methods was reported in [1] for a panel excited by two shakers and one speaker in laboratory conditions. This paper compares these two methods for an industrial case in terms of reconstruction of radiated pressure and transfer path identification.

2 TPA and OTPA

For the TPA, the matrix equations are the following:

$$\begin{cases} \{A\} = [H_1]\{F\} \\ \{P\} = [H_2]\{F\} \end{cases} \quad (1)$$

where F, A and P are vector quantities that represent the injected forces, corresponding accelerations and radiated pressure, respectively. H_1 and H_2 are matrices containing 'acceleration upon force' and 'pressure upon force' transfer functions, respectively (obtained using static measurements). A will be measured in dynamic conditions. To determine F, the matrix H_1 has to be inverted and the SVD (singular values decomposition) will be applied.

For the OTPA, we have the following equations:

$$\begin{cases} \{A\} = [H_1]\ \{F\} & (2.1) \\ \{P\} = [H_3]\ \{A\} & (2.2) \end{cases}$$

The unknowns are F and H_3. The matrix H_3 is now calculated by taking several operational measurements to obtain a sufficient number of equations to express the radiated pressure at each speed by a linear combination of the measured accelerations [2].

3 Experimental characterization

3.1 Measurements

To perform the measurements, a test workbench (Figure 1) for the drive system was designed. It allows controlling the operating speed of the drive shaft and set the track in realistic rotating conditions. Other sources of noise and vibrations are eliminated since they do not operate (engine, CVT). Realistic load conditions (weight of a driver, …) are achieved by putting the tunnel under constraint using a hydraulic actuator.

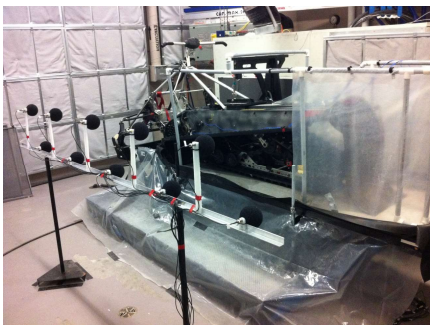


Figure 1 : Test workbench for the measurements.

This study was focused on the noise linked to the excitation corresponding to the engagement of the drive shaft on the track by the sprocket teeth (composed of eight teeth). The corresponding frequency depends on the vehicle speed, and at a given rotational speed (rpm), it equals eight times the rpm divided by 60 (then called order 8).

Twelve 3D accelerometers (named A1-A12) were placed on the suspension, and three target microphones were positioned in the room in front of the suspension (see figure 1). Measurements were made on a 0-2048 Hz frequency range with a 2 Hz frequency resolution (the upper considered frequency is 600 Hz), and separated in two static and dynamic steps.

Static measurements:

1. H_1 transfer functions were determined using impact hammer testing at each accelerometer locations (3D measurements). The size of H_1 is 36x36x1024.
2. H_2 transfer functions were reciprocally determined, using a volume velocity source located at each microphone position. This volume source has low frequency limitations under a frequency of 200 Hz. The size of H_2 is 3x36x1024.

Operational measurements:

3. Operational accelerations and pressures were measured during three consecutive run-ups of the drive system from 20 to 120 km/h (then discretized in 147 speeds). The sizes of A and P are 36x147x1024 and 3x147x1024.

3.2 Assumptions

Since measurements could be made only on one side of the suspension, it was verified by preliminary tests that acceleration levels and frequency contents were similar on

[1] nassardin.guenfoud@gmail.com
[1] Olivier.Robin@usherbrooke.ca
[1] Raymond.Panneton@usherbrooke.ca
[1] Alain.Desrochers@usherbrooke.ca
[2] walid.belgacem@cta-brp-udes.com

both sides of the suspension so that the problem is considered symmetric. Two important assumptions were also made:
1. For both methods: the transfer functions are independent of the vehicle speed.
2. For OTPA only: the measured sound pressure level can be expressed as a linear combination of the acceleration measurements of each points.

4 Results

For the order 8, results of TPA and OTPA modelling in terms of radiated pressure compared with the measured one are shown in figure 2.

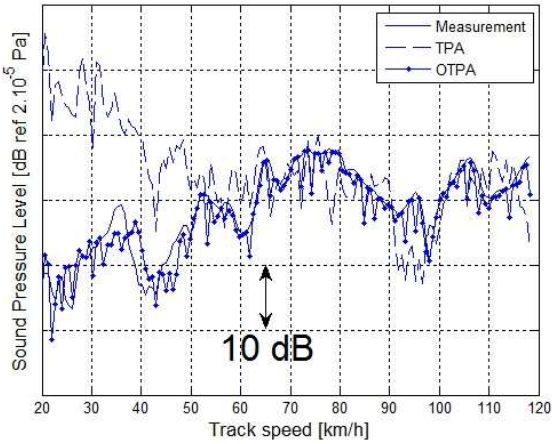


Figure 2 : Comparison of the TPA and OTPA modelling results with a dynamic pressure measurement (order 8).

The reconstruction results are very similar for both methods. However the TPA clearly overestimates the pressure level below a speed of 50 km/h, which is linked to the low frequency limitation of the volume source. This limitation is absent in the case of OTPA which remains valid at all frequencies.

5 Identification of transfer paths

For the TPA, Eq. (1) is used to calculate the contribution (A1 to A12) as following.

$$\{P\} = [H_2][H_1]^{-1}\{A\} \quad (3)$$

For the OTPA, Eq. (2.2) has to be used after having calculated the matrix H_3 using operational data only.
Figure 3 shows the respective contribution in terms of noise level for the 36 considered transfer paths as a function of the track speed. The main transfer paths can be easily identified and ranked in order to target which component should be modified in order to obtain possible reduction of the noise generated by the drive system. Note that in the case of the TPA, main transfer paths are better distinguished.
Four major contributors are easily identified from the results shown in figure 3: A3, A8, A11 and A12. A new experimental test was made for testing the effect of structural modifications at these points. The results showed a significant reduction in the noise of the drive system. This confirms that the two models were able to well identify the main transfer paths and sources of the considered snowmobile rear suspension.

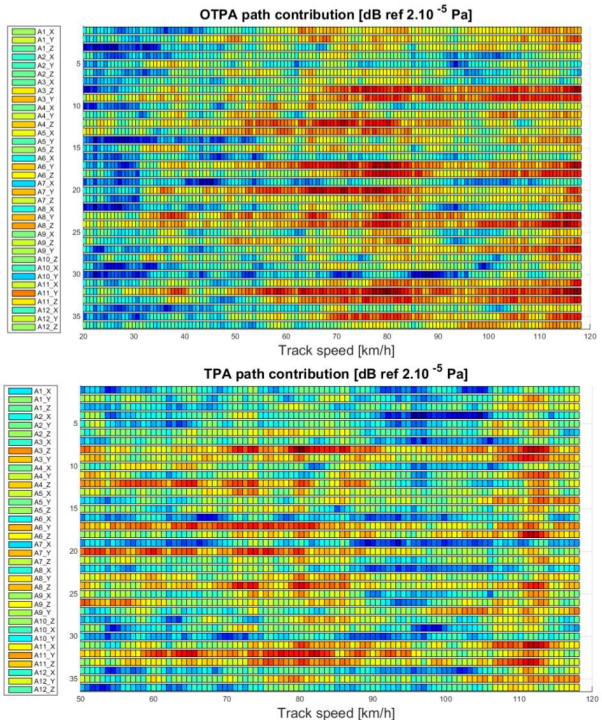


Figure 3 : Contribution of transfer paths in the case of OTPA and TPA, for the order 8 (TPA results are given from 50 km/h, whereas OTPA results start from 20 km/h).

6 Conclusions

The methodologies of TPA and OTPA were applied to the rear suspension of the snowmobile. The OTPA is quicker to implement since measurements of H_2 are no longer needed, and it avoids limitations linked to bandwidth-limited transfer functions (here low frequency limitations of the volume velocity source). However, it relies on a strong hypothesis (pressure is a linear combination…) that can have for consequence overestimations in the matrix H_3 if other sources than those included in the model strongly influence the measured sound pressure level. The TPA generally provides more precise predictions; however both methods provided a satisfactory identification of transfer paths. This was confirmed by noise reduction results obtained with additional tests on a snowmobile including modifications of the identified main paths and noise contributors.

Acknowledgements

This research was funded by NSERC (Natural Sciences and Engineering Research council of Canada).

References

[1] C. Sandier, Q. Leclere, N. B. Roozen. Operational transfer path analysis: theoretical aspects and experimental validation. *Acoustics 2012*, Nantes, France, 2012.
[2] H. G. Moura, A. Lenzi. An alternative formulation of transfer path analysis applied to the force identification problem. *20th Int. Congr. Mech. Eng.,* Brazil, 2009.

A DATA PROCESSING METHOD FOR NOISE MEASUREMENTS OF SNOWMOBILES AND THEIR SUB-SYSTEMS ON TEST BENCH

Jason Méjane [*], **Raymond Panneton** [†], **Alain Desrochers** [‡] **and Rémy Oddo** [§]
Department of Mechanical Engineering, GAUS, Université de Sherbrooke, Sherbrooke, Qc, Canada, J1K 2R1

1 Introduction

Snowmobile sound emission is qualified using standards SAE J1161 and J192. They set the noise level limit during Pass-by test, test depending on environmental conditions [1]. As the objective is to compare two models of CVT driving pulleys (responsible of noise) and to avoid this dependency, use of a test bench is a solution. Maximum engine speed has to be reached and maintained on test bench to reproduce Pass-by condition. But speed and load controls can be hard to achieve and comparison of pulleys (using time averaging of the recorded signal) led to results different from Pass-by comparison (time averaging is applied on the signal for maximum engine speed on test bench). This is because engine speed can vary up to 20% even with wide open throttle. Engine speed has to be accounted when using a test bench to predict noise level reduction in Pass-by test between two snowmobile CVT, as noise depends on it [2, 3]. Test bench measurements also need a processing method, this one taking into account engine speed. Including statistical indicators improves the analysis [4]. This paper describes and discusses a data processing method based on bench tests measurement, allowing reliable noise comparison between two different models of a snowmobile CVT, and predictions on Pass-by noise reduction.

2 Method

2.1 Measurement

The test bench used is a functional snowmobile, fixed to a plate using the two front skis. The track is not driven. A water-brake torque load is attached to the driven pulley, attempting to reproduce normal use torque. All noise sources but the CVT driving pulley are isolated. An antenna of four microphones faces the CVT within 2.2 meters (8 times the minimal considered wave length, i.e. 1200 Hz). Time signals of the microphones and the engine speed are recorded. One measurement lasts 8 seconds, during which wide open throttle acceleration is maintained. Because of speed variations and to make statistical indicators more robust, a minimal number of 10 measurements are made. Two models of CVT are compared : model 'A' (noisy) and model 'B' (silent).

2.2 Calculation

Data processing algorithm

To take noise level dependency to engine speed into account, data will be processed and ranked in relation to the engine speed. The data is processed as following :

• Step 1 - For each pulley model :

First, for each of the 10 measurements, the algorithm slices the signals into FFT blocks with overlap. Then, it computes, for each block, the frequency spectrum of every microphone and the mean rpm of the block. As the speed range was cut into steps, the mean rpm will match a rpm step. So for each rpm step increment, it averages the microphones' frequency spectra. If, for the duration of the measurement, several time intervals match the same rpm step, it finally averages those spectra.

Secondly, for each rpm step, the algorithm averages the frequency spectra of every measurement (having a spectrum matching the rpm step). This provides the mean spectrum of the model and the number of contributing measurements for each rpm step. Thanks to this, variance is finally computed.

• Step 2 - Comparison between two models :

To compare two models, the algorithm computes the statistical indicators for each rpm step and each frequency band.

Statistical indicators

Confidence intervals : This indicator depends on the standard deviation, calculated as the square root of the unbiased variance estimator. To compare the results of two CVT models, we look at the overlap of their confidence intervals (CI) for the 'A' and 'B' models as in (1) :

$$\bar{x}_A > \bar{x}_B \rightarrow \begin{cases} CI_A & = \bar{x}_A - t_{1-\alpha/2}^{n_A-1} \times (\sigma_A/\sqrt{n_A}) \\ CI_B & = \bar{x}_B + t_{1-\alpha/2}^{n_B-1} \times (\sigma_B/\sqrt{n_B}) \end{cases} \quad (1)$$

where $t_{1-\alpha/2}^{n-1}$ is the quantile from the t-distribution table depending on the degrees of freedom $n-1$ and the confidence level α (Student's law), and $\bar{x}_A$ and $\bar{x}_B$ are the models' mean values. The computation starts with a large α (small interval entails small confidence). If the two intervals do not overlap, the value of α is decreased and the previous test is repeated, and so on until the intervals overlap. The last value of α for which there is no overlap is kept, giving the confidence $P(\%) = 100(1-\alpha)$. This indicator can not be used to state on the reality of the difference, as the true means may be very close to each other.

Significance of differences coefficient : The significance of the differences coefficient (SDC thereafter) is introduced in this study to answer the need to better understand and determine the noise differences between two models. This coefficient is based on the mean difference between the models 'A' and 'B'. First, the difference between unbiased variances

*. jason.mejane@usherbrooke.ca
†. raymond.panneton@usherbrooke.ca
‡. alain.desrochers@usherbrooke.ca
§. remy.oddo@usherbrooke.ca

is calculated as $\Delta_\sigma = |\Delta_m| - k \times (\sigma_A + \sigma_B)$, where σ_A and σ_B are the standard deviation respectively of model A and model B, $\Delta_m = \bar{x}_A - \bar{x}_B$, and k which is a factor representing the number of standard deviations for the normal distribution which is a characteristic of the population proportion taken into consideration. Hence, the SDC is introduced as the relation between Δ_σ and Δ_m, i.e. $SDC_k = \Delta_\sigma / \Delta_m$.

The SDC is non-dimensional and can also easily be computed as a percentage of the mean difference as in (2) :

$$SDC_{k,\%} = 100 \left(1 - k \times \frac{\sigma_A + \sigma_B}{|\Delta_m|} \right) \tag{2}$$

The SDC enlightens about the toughness of the results, particularly when the samples include few measures (for instance because of highly variable engine speed). It can be used to state on the probability to obtain a minimum difference or that a mean difference is accurate (not a measurement error).

3 Results

3.1 Comparison with Pass-by noise measurement and simple time averaging

On the figure 1 below, results from the data processing method are compared with Pass-by measurement and time averaging method ones, for two different test bench measurement sessions (first with middling torque and speed control, second with improved control). Noise level difference is between CVT 'A' and 'B'. Numbers of measurements are equivalent for both sessions.

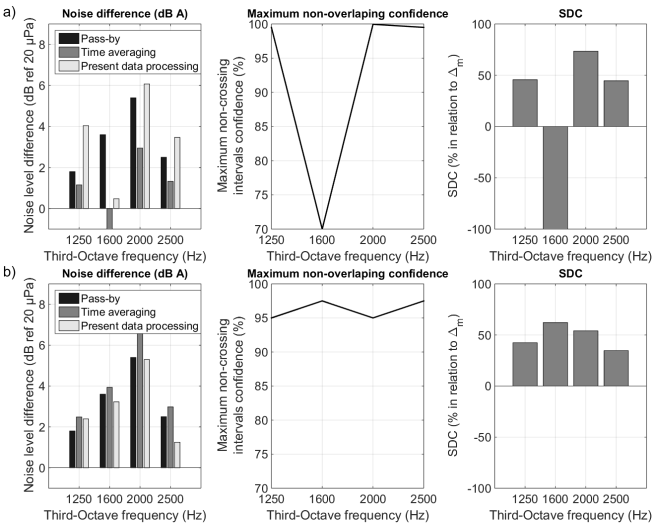


Figure 1: Comparison of noise level differences and performance of the method : a) with middling torque and speed control, b) with improved control.

3.2 Tests results analysis

Time averaging always returns relatively different results from Pass-by ones. The data processing method yields closer results to those from Pass-by, so long as the conditions on the test bench remain well controlled. Moreover, indicators facilitates the judgement on the quality of the measurements and help make predictions : in the case of the middling control, SDC values show that results are unreliable, unlike for the second case.

4 Discussion

During some of the tests, comparison of the 2 pulleys using time averaging on test bench returned completely false results compared with Pass-by comparison, as engine speed can vary a lot. Even if the proposed method takes more time to process data, gains are potentially large, thanks to statistical indicators improving the reliability of the results and predictions. Test conditions are also really important on the test bench so as to obtain comparable results with those from Pass-by. Moreover, this method provides results for an extended large range of speed.

5 Conclusions

A new data processing method for noise measurements on test bench has been described and used to compare noise levels between two snowmobile CVT. This method is a relatively simple algorithm associating noise level to the engine speed. Included statistical indicators make easier comparison and prediction, as long as test conditions on the bench are well controlled (reproduce real use conditions), and number of measures is large enough. Test bench measurement saves time and does not depend on environmental conditions compared with Pass-by measurement. The comparison fairly matches with Pass-by results and predictions seem to be made with reliability.

Acknowledgments

The authors wants to acknowledge the work of Walid Belgacem in the design of the test bench. This study was conducted as part of a project financially supported by NSERC and BRP.

References

[1] B.J. Dilworth. *Identification of ground and environmental effects to the pass-by noise testing of snowmobiles*. PhD thesis, 2009.

[2] Ralph K Hillquist. Motor vehicle noise spectra, their characteristics and dependence upon operating parameters. *J. Acoust. Soc. Am.*, 58(1) :2, 1975.

[3] M G Seshagiri Rao, P Ramalingeswara Rao, and K Srinivas Dev. Speed Dependence of Noise Emission Levels of Individual Motor Vehicles in Free Flow. *Acustica*, 67 :135–143, 1988.

[4] C. Dine and P. Fodiman. New experimental methods for an improved characterisation of the noise emission levels of railway systems. *J. Sound Vib.*, 231(3) :631–638, mar 2000.

OPTIMIZATION STUDY AND PANEL PARAMETER STUDY FOR NOISE RADIATION REDUCTION OF AN AIRCRAFT PANEL EXCITED BY TURBULENT FLOW

Steven A.J. Sonnenberg [†] **and Joana Rocha** [‡]
Department of Mechanical and Aerospace Engineering, Carleton University, Ottawa, Ontario.

1 Introduction

The noise and vibration in an aircraft cabin, during cruise conditions, is primarily caused by the external turbulent boundary layer (TBL) [1]. The TBL causes the fuselage panels on the aircraft to vibrate, which radiate sound energy in the form of noise in the cabin. Therefore, it is of great interest to determine which aircraft panel parameter(s) is/are most effective in decreasing the amount of radiated sound power, and how to optimize these parameters to reduce the noise in the aircraft cabin.

One approach to calculate the radiated sound power (RSP) of vibrating structures is to use a modal analysis, as done by Roy and Lapi [2]. This approach is necessary when analyzing obscure shapes, but requires great computational power and time, making it difficult to iterate the calculations for optimization routines. Therefore, when looking at simple shapes, like that of a flat panel, analytical computational methods become a better choice. The analytical expressions for RSP can be derived for a given aircraft panel, in terms of the displacement power spectral density (PSD) [1, 3, 4]. The acceleration PSD is calculated from the displacement PSD, which is proportional to the RSP [4]. The analytical models developed were modified to account for other panel and enclosure combinations [5, 6]. Berry also showed that the same type of analytical analysis was possible for panels with arbitrary boundary conditions [7].

An analytical model that calculates the acceleration PSD was created in Matlab. A sensitivity study was performed on the panel parameters to determine the percent change in acceleration PSD, versus the percent change in seven different panel parameters. An analytical method to optimize an aircraft panel is presented in this paper for reducing the acceleration PSD of the panel caused by the TBL.

2 Methodology

To calculate the acceleration PSD of an aircraft panel, a few assumptions must be made; the panel is assumed to be flat, simply supported and not under tension. The panel is also assumed to not be fixed to an enclosure and with a TBL caused from a single constant freestream velocity. The displacement PSD at a single point (taken to be the centre of the panel in all calculations for this study) can be calculated for a given frequency, using the displacement PSD matrix as follows [1]:

$$S_{WW}(x_1, y_1, x_2, y_2, \omega) = \sum_{m_{x_1}, m_{x_2}=1}^{M_x^2} \sum_{m_{y_1}, m_{y_2}=1}^{M_y^2} \alpha_{m_{x_1}}(x_1) * \alpha_{m_{x_2}}(x_2) * \beta_{m_{y_1}}(y_1) * \beta_{m_{y_2}}(y_2) * S_{ww}(\omega)_{m_1, m_2} \quad (1)$$

Where $\alpha_{m_x} = \sqrt{\frac{2}{a}} \sin\left(\frac{m_x \pi x}{a}\right)$ and $\beta_{m_y} = \sqrt{\frac{2}{b}} \sin\left(\frac{m_y \pi y}{b}\right)$ are spatial functions that define the variation in vibration for a simply supported plate and Where (m_x, m_y) is the plate mode and $M = Mx = My$ is the total number of plate modes considered [1]. The velocity PSD (S_{VV}) and the acceleration PSD (S_{AA}) at a single point on the panel are as follows [4]:

$$S_{VV} = \omega^2 * S_{WW} \quad (2)$$

$$S_{AA} = \omega^4 * S_{WW} \quad (3)$$

A complete description/derivation of the equations used for this study can be found in Rocha's earlier work [1, 3].

3 Results

A sensitivity study was performed on seven panel parameters, to determine which parameter is most effective at reducing the acceleration PSD in select octave bands. The parameters were varied individually, and the changes in the acceleration PSD in each of the octave bands were analyzed. To run a full, in depth sensitivity study, all the frequencies in the human hearing range would need to be analyzed, but as the frequency increases, the analysis takes exponentially longer to run, therefore only four lower frequency octave bands have been analyzed. The octave bands are from 89.1-178 Hz, 178-355 Hz, 355-708 Hz and 708-1410 Hz. The sensitivity study was run for the following parameters: thickness, material density, panel width and length, elastic modulus, Poisson's ratio and damping ratio. Figure 1 contains a sensitivity study for a single octave band.

The analysis was then modified to determine the optimal panel parameters that resulted in the smallest average acceleration PSD over the octave band. The optimization analysis implemented in this study, is an add-on to Matlab. It uses an interior point algorithm for a nonlinear equation, which is a method of constrained minimization that solves a sequence of approximate minimization problems. The analysis is used to optimize each of the seven parameters individually and concurrently. Since the general trend of the sensitivity studies predicts that the minimum acceleration PSD is reached when both the thickness and the density are maximized to the upper constraint, optimizing these parameters individually simply results in the upper constraint. Therefore, it is of more interest to determine if there is a correlation between the

[†] Steven.Sonnenberg@carleton.ca
[‡] Joana.Rocha@carleton.ca

octave band and the panel length. Figure 2 shows the optimal panel length at the center frequency of different octave bands and compares these values to the calculated flexural wavelength, convective wavelength and acoustic wavelength at the same frequencies.

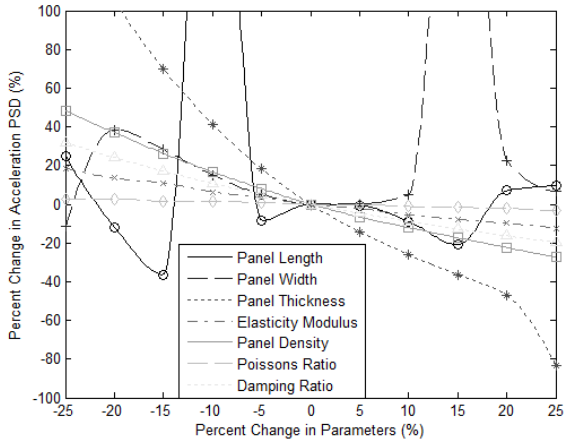


Figure 1: Percent change in acceleration PSD versus percent change in panel parameter for octave 355-708 Hz with limited Y-axis extents.

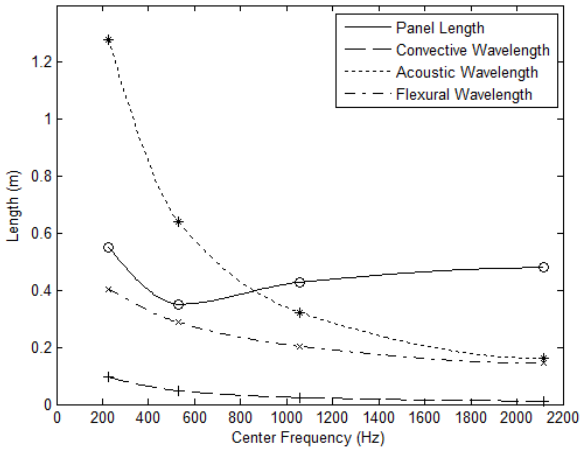


Figure 2: Optimal panel length at the center frequency of different octave bands compared to the calculated flexural wavelength, convective wavelength and acoustic wavelength.

4 Discussion

As shown in Figure 1, both the panel width and length have fluctuating values. These fluctuations are believed to occur because the panel width and length are main components of calculating $S_{tbl}(\omega)$. The variables are located within sinusoidal functions, with the change in these parameters being non-linear. For this reason, these parameters cannot be defined by a simple trend, and therefore, are not the most sensitive at reducing the overall acceleration PSD.

It was found that the two parameters that are the most effective for reducing the average acceleration PSD within the different octave bands are panel thickness and panel density as these two parameters have the steepest slopes. As the thickness is increased, the higher frequency noise is reduced which is what can be expected. However it has less effect on the lower frequency (longer wavelength) signals. Even though the panel density has more gradual slopes in comparison to the panel thickness, the trend is more consistent across all of the analyzed octave bands. Therefore, it is likely that panel density is the most sensitive at reducing the overall noise across the human hearing range, whereas thickness may be the most sensitive at reducing the noise at certain octave bands.

It was assumed that the optimal panel length would follow the same exponential decay as the flexural wavelength, convective wavelength and acoustic wavelength, however, Figure 2 does not support this hypothesis. The optimization routine currently finds local minimums in the constrained space rather than the global minimum. The next step in this research will to be to modify the optimization routine to find the global minimums and determine the true trend of the panel length.

5 Conclusion

An optimization study is presented, with the objective to reduce the acceleration PSD of a panel excited by a TBL. It is shown that panel thickness and panel density are the most consistent, and effective parameters at reducing the acceleration PSD at different octave bands in the human hearing range.

References

[1] J. Rocha, A. Suleman and F. Lau, "An accurate Coupled Structural- Acoustic Analytical Framework for the Prediction of Random and Flow-Induced Noise in Transport Vehicles: Its Validation," *Canadian Acoustics*, vol. 37, no. 3, 2009.

[2] N. Roy and M. Lapi, "Efficient Computation of the Radiated Sound Power of Vibrating Structures using a Modal Approach," in *Acoustics*, Paris, 2008.

[3] J. Rocha and D. Palumbo, "On the Sensitivity of Sound Power Radiated by Aircraft Panels to Turbulent Boundary Layer Parameters," *Journal of Sound and Vibrations*, vol. 331, pp. 4785-4806, 2012.

[4] J. Rocha, "Sound Radiation and Vibration of Composite Panels Excited by Turbulent Flow: Analytical Prediction and Analysis," *Shock and Vibration*, vol. 2014, no. 316481, pp. 1-18, 2014.

[5] J. Rocha, A. Suleman and F. Lau, "Turbulent Boundary Layer Induced Noise and Vibration of a Multi-Panel Walled Acoustic Enclosure," *Canadian Acoustics*, vol. 38, no. 4, pp. 9-22, 2010.

[6] J. Rocha, A. Suleman and F. Lau, "Prediction of Turbulent Boundary Layer Induced Noise in the Cabin of a BWB Aircraft," *Shock and Vibration*, vol. 19, pp. 693-705, 2012.

[7] A. Berry, J.-L. Guyader and J. Nicolas, "A general formulation for the sound radiation from rectangular, baffled plates with arbitrary boundary conditions," *Journal of the Acoustical Society of America*, vol. 88, no. 6, pp. 2792-2802, 1990.

Quantification Of Footstep Noise Transmission In The Interior Of An Apartment Building

Max De Castro Magalhães

In this paper an experimental approach was used to investigate the structure-borne noise transmission in an apartment building where the main noise source was heeled shoes. Several transmission paths were studied and their effect on the overall noise inside the rooms was determined by measurements of the sound pressures at two receiver locations. The principal transfer paths were highlighted and classified in terms of their importance. The total structure-borne was predicted from the incoherent addition of two types of heeled shoes by subtracting the residual noise from other paths. A comparison was also made between direct measurements and prediction.

Scantek, Inc.
Calibration Laboratory

Scantek offers traceable, high quality and prompt periodic calibration of any brand of sound and vibration instrumentation

Calibration and Service Capabilities:

- Microphones
- Preamplifiers
- Acoustical Calibrators
- Sound Level Meters & Analyzers
- Accelerometers & Vibration Meters
- Vibration Calibrators and more

ISO 17025:2005 and ANSI/NCSL Z-540-1: 1994 Accredited Calibration Laboratory

Scantek, Inc.
Sales, Rental, Calibration
www.ScantekInc.com/calibration 800-224-3813

EDITORIAL BOARD - COMITÉ ÉDITORIAL

Hearing Conservation - Préservation de l'ouïe
Alberto Behar (416) 265-1816 albehar31@gmail.com
Ryerson University

Technical Notes - Exposés techniques
Umberto Berardi 416 979 5000 (3263) uberardi@ryerson.ca
Ryerson University

Special Issue: Canadian Acoustics Cities - Édition spéciale : Acoustique canadienne des villes
Umberto Berardi 416 979 5000 (3263) uberardi@ryerson.ca
Ryerson University

Musical Acoustics / Electroacoustics - Acoustique musicale / Électroacoustique
Annabel J Cohen acohen@upei.ca
University of P.E.I.

Signal Processing / Numerical Methods - Traitement des signaux / Méthodes numériques
Tiago H. Falk (514) 228-7022 falk@emt.inrs.ca
Institut national de la recherche scientifique (INRS-EMT)

Aeroacoustics - Aéroacoustique
Anant Grewal (613) 991-5465 anant.grewal@nrc-cnrc.gc.ca
National Research Council

Physiological Acoustics - Physio-acoustique
Robert Harrison (416) 813-6535 rvh@sickkids.ca
Hospital for Sick Children, Toronto

Underwater Acoustics - Acoustique sous-marine
Garry J. Heard 902-426-3100 x310 garry.heard@gmail.com
Defence R-D Canada Atlantic Research Centre

Psychological Acoustics - Psycho-acoustique
Jeffery A. Jones jjones@wlu.ca
Wilfrid Laurier University

Consulting - Consultation
Tim Kelsall 905-403-3932 tkelsall@hatch.ca
Hatch

Architectural Acoustics - Acoustique architecturale
Jean-François Latour (514) 393-8000 jean-francois.latour@snclavalin.com
SNC-Lavalin

Shocks / Vibrations - Chocs / Vibrations
Pierre Marcotte marcotte.pierre@irsst.qc.ca
IRSST

Hearing Sciences - Sciences de l'audition
Kathleen Pichora Fuller (647) 794-7560 k.pichora.fuller@utoronto.ca
University of Toronto

Speech Sciences - Sciences de la parole
Linda Polka 514-398-7235 linda.polka@mcgill.ca
McGill University

Physical Acoustics / Ultrasounds - Acoustique physique / Ultrasons
Werner Richarz wricharz@echologics.com

Engineering Acoustics / Noise Control - Génie acoustique / Contrôle du bruit
Joana Rocha joana.rocha@carleton.ca
Carleton University

Bio-Acoustics - Bio-acoustique
Jahan Tavakkoli (416) 979-5000 jtavakkoli@ryerson.ca
Ryerson University

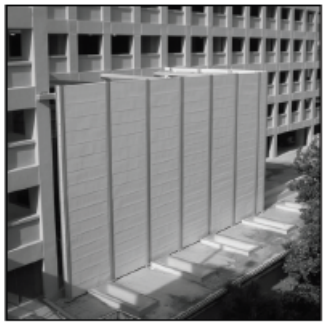


Environmental Noise Control

Community friendly solutions for chillers and cooling towers

- Over 50 Years of Proven Design and Performance
- Independently Tested Products
- On Grade or Roof Top Applications
- Maximum Noise Reduction
- Low System Pressure Loss

Central Energy Plant Louvers



Cooling Tower Barrier Wall System

KINETICS Noise Control

The Canadian Acoustical Association - L'Association canadienne d'acoustique

CANADIAN ACOUSTICS TELEGRAM ANNOUNCEMENTS - ANNONCES TÉLÉGRAPHIQUES DE L'ACOUSTIQUE CANADIENNE

Tri-Agency Open Access Policy on Publications

The good news? Publication and copyright policies of Canadian Acoustics journal are fully compliant with these new rules! That's another good reason for reseachers to publish in Canadian Acoustics!

The Canadian Institutes of Health Research (CIHR), the Natural Sciences and Engineering Research Council of Canada (NSERC) and the Social Sciences and Humanities Research Council of Canada (SSHRC) are federal granting agencies that promote and support research, research training and innovation within Canada. As publicly funded organizations, the Agencies have a fundamental interest in promoting the availability of findings that result from the research they fund, including research publications and data, to the widest possible audience, and at the earliest possible opportunity. Societal advancement is made possible through widespread and barrier-free access to cutting-edge research and knowledge, enabling researchers, scholars, clinicians, policymakers, private sector and not-for-profit organizations and the public to use and build on this knowledge.According to a new policy, all grant recipients that were funded in whole or in part by NSERC or SSHRC for grants awarded May 1, 2015 and onwar (Januray 1, 2008 for CIHR) are required to ensure that any peer-reviewed journal publications arising from Agency-supported research are freely accessible within 12 months of publication.

May 28th 2015

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/

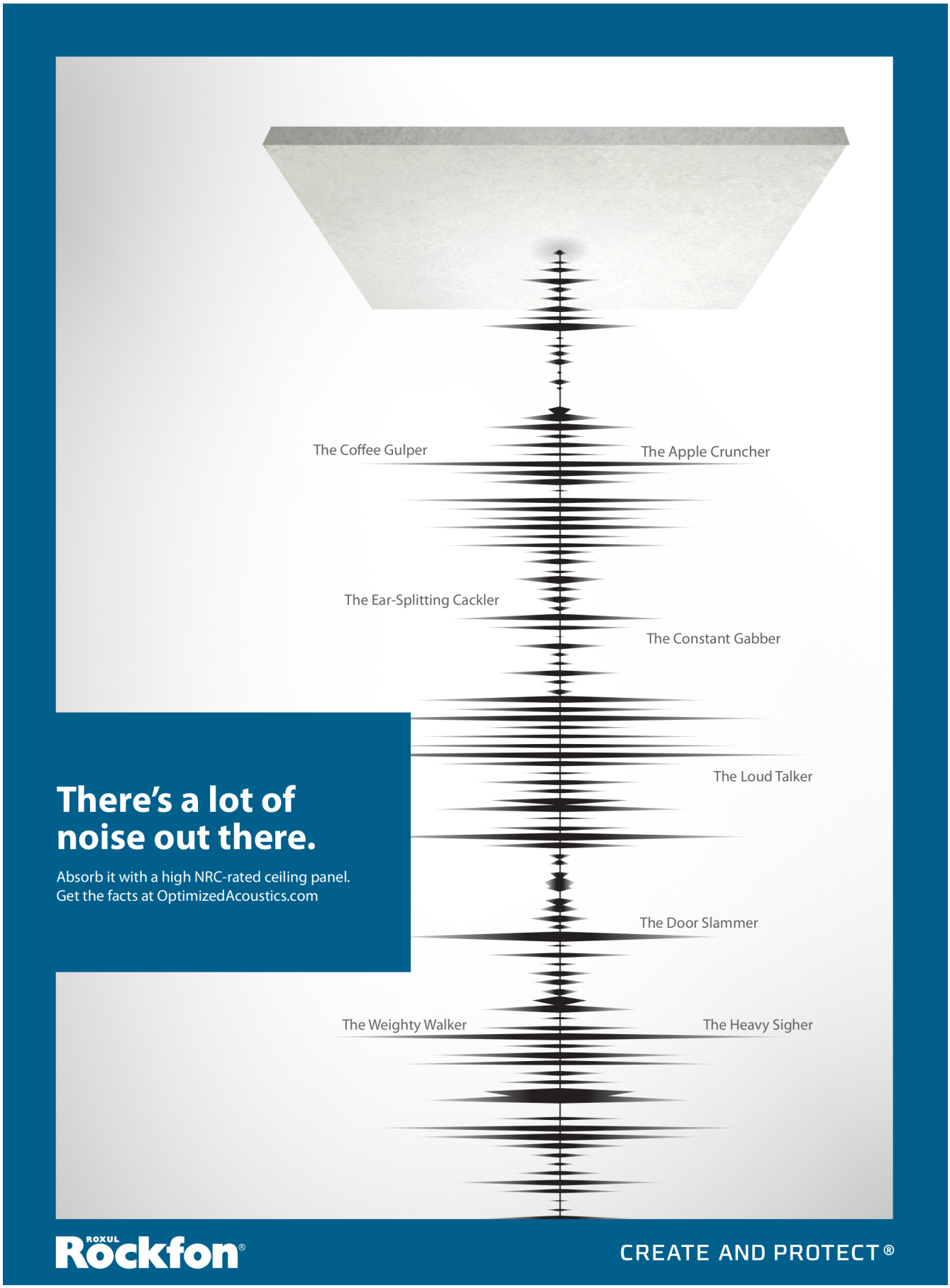
August 5th 2015

CAA is now social!

Canadian Acoustical Association is moving to the social media!

Find us on social media:Twitter: CanAcousticalFacebook: facebook.com/canadianacousticalassociation

December 14th 2015


The Coffee Gulper
The Apple Cruncher
The Ear-Splitting Cackler
The Constant Gabber
The Loud Talker
The Door Slammer
The Weighty Walker
The Heavy Sigher
There's a lot of noise out there.
Absorb it with a high NRC-rated ceiling panel.
Get the facts at OptimizedAcoustics.com
ROXUL
Rockfon®
CREATE AND PROTECT®

Soft dB

MEZZO

Innovative and cost-effective acoustic solution

- Class 1 Sound Level Meter,
- Building Acoustics,
- Noise Monitoring,
- Sound Intensity,
- Sound Intensity Imagery.



Visit www.SoftdB.com/mezzo for more info

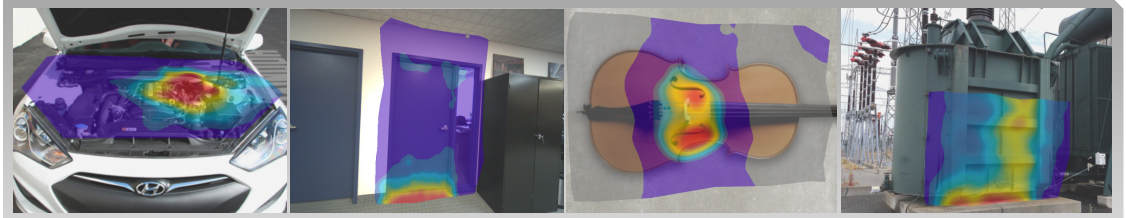
Range of Analyzers



Stand-Alone Noise Monitoring



High Quality Sound Intensity Imagery



BOEING Carrier OTIS inrs BRP ALCOA Atlas Copco

Soft dB 1040, Belvedere Avenue, Suite 215, Quebec (Qc) G1S 3G3 Canada

Phone 1 (418) 686-0993
Toll Free 1 (866) 686-0993

Web Site www.softdb.com
E-mail info@softdb.com

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

Canadian Acoustical Association
Association canadienne d'acoustique

Application for Membership

CAA membership is open to all individuals who have an interest in acoustics. Annual dues total $100.00 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 $100.00 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 $ 100.00
[ ] CAA Student Membership $ 50.00

Corporate Subscriptions (4 issues/yr)
[ ] $100 including mailing in Canada
[ ] $108 including mailing to USA,
[ ] $115 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, The Canadian Acoustical Association, PO Box 74068, Ottawa, Ontario, K1M 2H9, Canada

Association Canadienne d'Acoustique
Canadian Acoustical Association

Formulaire d'adhésion

L'adhésion à l'ACA est ouverte à tous ceux qui s'intéressent à l'acoustique. La cotisation annuelle est de 100.00$ pour les membres individuels, et de 50.00$ pour les étudiants. Tous les membres reçoivent ***l'Acoustique Canadienne***, la revue de l'association. Les nouveaux abonnements reçus avant le 31 août s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de ***l'Acoustique Canadienne*** de cette année. Les nouveaux abonnements reçus après le 31 août s'appliquent à l'année suivante.

Abonnement pour la revue *Acoustique Canadienne* et abonnement de soutien

Les abonnements pour la revue ***Acoustique Canadienne*** sont disponibles pour les compagnies et autres établissements au coût annuel de 100.00$. Des compagnies et établissements préfèrent souvent la cotisation de membre bienfaiteur, de 475.00$ par année, pour assister financièrement l'ACA. La liste des membres bienfaiteurs est publiée dans chaque issue de la revue ***Acoustique Canadienne***. Les nouveaux abonnements reçus avant le 31 août s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de ***l'Acoustique Canadienne*** de cette année. Les nouveaux abonnements reçus après le 31 août s'appliquent à l'année suivante.

Notez que tous les paiements électroniques sont acceptés en ligne http://jcaa.caa-aca.ca

Pour correspondance administrative et financière:

Nom / Organisation _____

Adresse _____

Ville/Province _____ Code postal _____Pays _____

Téléphone _____ Téléc. _____ Courriel _____

Adresse postale pour la revue Acoustique Canadienne

Nom / Organisation _____

Adresse _____

Ville/Province _____ Code postal _____Pays _____

Cocher vos champs d'intérêt: (maximum 3)

1. Acoustique architecturale
2. Génie acoustique / Contrôle du bruit
3. Acoustique physique / Ultrasons
4. Acoustique musicale / Électro-acoustique
5. Physio / Psycho-acoustique
6. Chocs et vibrations
7. Audition
8. Parole
9. Acoustique sous-marine
10. Traitement des signaux /Méthodes numériques
11. Autre

Prière de remplir pour les étudiants et étudiantes:

_____ _____ _____ _____

(Université) (Nom d'un membre du corps professoral) (Signature du membre du corps professoral) (Date)

Cocher la case appropriée:

[ ] Membre individuel 100.00 $
[ ] Membre étudiant(e) 50.00 $

Abonnement institutionnel

[ ] 100 $ à l'intérieur du Canada
[ ] 108 $ vers les États-Unis
[ ] 115 $ tout autre envoi international

[ ] Abonnement de soutien 475.00 $ (comprend l'abonnement à *L'acoustique Canadienne*)

Merci de noter que le moyen de paiement privilégie est le paiement par carte crédit en ligne à *http://jcaa.caa-aca.ca*

Pour les individus ou les organisations qui préféreraient payer par chèque, l'inscription se fait en ligne à *http://jcaa.caa-aca.ca* puis le chèque peut être envoyé à :

Secrétaire exécutif, Association canadienne d'acoustique, BP 74068, Ottawa, Ontario, K1M 2H9, Canada

AWC 2017

Acoustics Week in Canada 2017

Semaine canadienne de l'acoustique 2017



Save the Date!

October 11-13, 2017

Guelph, Ontario

You are cordially invited to the 2017 Acoustics Week in Canada conference to be hosted at the *Delta Guelph Hotel and Conference Centre* in Guelph Ontario.

Technical sessions are being planned to cover the range of acoustic interests, with a few special surprises in the works. If you would like to propose or organize a special session on a specific topic, please contact Christian Giguère, technical chair for the conference (cgiguere@uottawa.ca).

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

An exhibition of acoustical products and services will be held during the conference. If you are interested in reserving your spot at the exhibition or in sponsorship, Bernard Feder (bfeder@hgcengineering.com) would be pleased to hear from you.

Guelph has the services of a larger city but is close to the countryside. Take a few days before or after the conference to enjoy the area! Visit the St. Jacobs Market, Mennonite country or Octoberfest in nearby Kitchener-Waterloo or enjoy the fall colours.

Plan to attend!

Peter VanDelden, conference chair

(Peter.vandelden@rwdi.com)

Réservez cette date !

11 au 13 octobre, 2017

Guelph , Ontario

Vous êtes cordialement invités à la Semaine canadienne de l'acoustique 2017 qui se tiendra au *Delta Guelph Hôtel and Conference Centre* à Guelph en Ontario.

Des sessions techniques sont prévues pour couvrir l'éventail des intérêts acoustiques, avec quelques surprises spéciales dans les œuvres . Si vous souhaitez proposer ou organiser une session spéciale sur un sujet précis, SVP contactez Christian Giguère , président du comité technique pour la conférence (cgiguere@uottawa.ca) .

Les étudiants sont encouragés à participer. Les étudiants qui présentent des communications seront admissibles à l'un des trois prix de 500 $ qui sera attribué. Des bourses de conférence seront également disponibles pour les étudiants dont les articles sont acceptés pour présentation.

Une exposition de produits et services acoustiques aura lieu lors de la conférence. Si vous êtes intéressé à réserver votre place à l'exposition ou dans le parrainage, Bernard Feder (bfeder@hgcengineering.com) sera heureux de vous entendre.

Guelph a les atouts d' une grande ville, mais est restée proche de la nature. Prenez quelques jours avant ou après la conférence pour profiter de la région ! Visitez le marché *St Jacobs Market*., le pays mennonite ou l'Octoberfest dans les environs de Kitchener - Waterloo ou profiter des couleurs de l'automne.

Prévoyez d'y assister!

Peter VanDelden , président de la conférence

(Peter.vandelden@rwdi.com)

Canadian Acoustical Association
Association canadienne d'acoustique

CANADIAN ACOUSTICAL ASSOCIATION
ASSOCIATION CANADIENNE D'ACOUSTIQUE

PRESIDENT
PRÉSIDENT

Frank A. Russo
Ryerson University
president@caa-aca.ca

PAST PRESIDENT
PRÉSIDENT SORTANT

Christian Giguère
University of Ottawa
past-president@caa-aca.ca

EXECUTIVE SECRETARY
SECRÉTAIRE

Roberto Racca
JASCO Applied Sciences
secretary@caa-aca.ca

UPCOMING CONFERENCE CONVENOR
ORGANISATEUR DE CONFÉRENCE (FUTURE)

Kathleen Pichora Fuller
University of Toronto
conference@caa-aca.ca

PAST CONFERENCE CONVENOR
ORGANISATEUR DE CONFÉRENCE (PASSÉE)

Sean Pecknold
DRDC
sean.pecknold@drdc-rddc.gc.ca

TREASURER
TRÉSORIER

Dalila Giusti
Jade Acoustics Inc.
treasurer@caa-aca.ca

AWARDS COORDINATOR
COORDINATEUR DES PRIX

Hugues Nelisse
IRSST
awards-coordinator@caa-aca.ca

EDITOR-IN-CHIEF
RÉDACTEUR EN CHEF

Jérémie Voix
ÉTS, Université du Québec
editor@caa-aca.ca

WEBMASTER
WEBMESTRE

Sean Pecknold
DRDC
web@caa-aca.ca

BOARD OF DIRECTORS - CONSEIL D'ADMINISTRATION

Alberto Behar
Ryerson University
albehar31@gmail.com

Bill Gastmeier
HGC Engineering
gastmeier@rogers.com

Bryan Gick
University of British Columbia
gick@mail.ubc.ca

Hugues Nelisse
IRSST
nelisse.hugues@irsst.qc.ca

Sean Pecknold
DRDC
sean.pecknold@drdc-rddc.gc.ca

Kathleen Pichora Fuller
University of Toronto
k.pichora.fuller@utoronto.ca

Joana Rocha
Carleton University
joana.rocha@carleton.ca

Mehrzad Salkhordeh
dB Noise Reduction Inc.
mehrzad@dbnoisereduction.com

SUSTAINING SUBSCRIBERS - ABONNÉS DE SOUTIEN

The Canadian Acoustical Association gratefully acknowledges the financial assistance of the Sustaining Subscribers listed below. Their annual donations (of $475.00 or more) enable the journal to be distributed to all at a reasonable cost.
L'Association Canadienne d'Acoustique tient à témoigner sa reconnaissance à l'égard de ses Abonnés de Soutien en publiant ci-dessous leur nom et leur adresse. En amortissants les coûts de publication et de distribution, les dons annuels (de 475.00$ et plus) rendent le journal accessible à tous les membres.

Acoustec Inc.
J.G. Migneron - 418-496-6600
courrier@acoustec.qc.ca
acoustec.qc.ca

Aercoustics Engineering Ltd.
Nicholas Sylvestre-Williams - (416) 249-3361
NicholasS@aercoustics.com
aercoustics.com

AMEC
Mr. Frank Babic - 905.568.2929
frank.babic@amec.com
amec.com

Bruel & Kjaer North America Inc.
Andrew Khoury - 514-695-8225
andrew.khoury@bksv.com
bksv.com

Dalimar Instruments Inc
Monsieur Daniel Larose - 450-424-0033
daniel@dalimar.ca
www.dalimar.ca

dB Noise Reduction
Mehrzad Salkhordeh - 519-651-3330 x 220
mehrzad@dbnoisereduction.com
dbnoisereduction.com

FFA Consultants in Acoustics and Noise Control
Clifford Faszer - 403.508.4996
info@ffaacoustics.com
ffaacoustics.com

GHD
Tim C Wiens - 519-884-0510 x2352
tim.wiens@ghd.com
ghd.com

H.L. Blachford Ltd.
Duncan Spence - 905-823-3200
amsales@blachford.ca
blachford.ca

HGC Engineering Ltd.
Bill Gastmeier -
gastmeier@rogers.com
hgcengineering.com

Integral DX Engineering Ltd.
Greg Clunis - 613-761-1565
greg@integraldxengineering.ca
integraldxengineering.ca

Jade Acoustics Inc.
Ms. Dalila Giusti - 905-660-2444
dalila@jadeacoustics.com
jadeacoustics.com

JASCO Applied Sciences (Canada) Ltd.
Roberto Racca -
roberto.racca@jasco.com
jasco.com

Peutz & Associés
Marc Asselineau - 33-1-45230500
m.asselineau@peutz.fr
peutz.fr

Pliteq Inc.
Wil Byrick - 416-449-0049
wbyrick@pliteq.com
pliteq.com

Pyrok Inc.
Howard Podolsky - 914-777-7770
mrpyrok@aol.com
pyrok.com

ROCKFON
Dr. Gary S. Madaras - 708.563.4548
gary.madaras@rockfon.com
rockfon.com

ROXUL
Meghan Billingsley - 9053387600
amanda@birchallpr.com
roxul.com

RWDI AIR Inc.
Mr. Peter VanDelden - 519-823-1311
peter.vandelden@rwdi.com
rwdi.com

Scantek Inc.
Steve Marshall - 1-410-290-7726
S.Marshall@scantekinc.com
scantekinc.com

True Grit Consulting Ltd.
Ms Ina Chomyshyn - 807-626-5640
ina@tgcl.ca
tgcl.ca

Vibra-Sonic Control
Mr. Chris N. Wolfe - 604-294-9495
chris@vibra-sonic.com
vibra-sonic.com

Wakefield Acoustics Ltd.
Mr. Clair W. Wakefield - 250-370-9302
clair@wakefieldacoustics.com
wakefieldacoustics.com

West Caldwell Calibration Labs
Mary Richard - 905-554-8005
wcclca@gmail.com
wccl.com