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Proceedings of the Annual Meeting of
The Canadian Acoustical Association

Acoustics: Bridge to the Future

Charlottetown Oct 2002

Acoustique: un Pont vers l'Avenir

L'Association Canadienne d'Acoustique
Les actes de la réunion annuelle

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PROCEEDINGS OF THE ANNUAL MEETING OF

The Canadian Acoustical Association

LES ACTES DE LA RÉUNION ANNUELLE DE

L'Association Canadienne d'Acoustique

Charlottetown, Prince Edward Island
8 – 11 October 2002

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Canadian Acoustical Association / l'Association Canadienne d'Acoustique

The Canadian Acoustical Association/l'Association Canadienne d'Acoustique (CAA/ACA) is a professional interdisciplinary organization that fosters communication among people working in all areas of acoustics in Canada, promotes the growth and practical application of knowledge in acoustics, encourages education, research, and employment in acoustics, and is potentially an umbrella organization through which general issues in education, employment and research can be addressed at a national and multidisciplinary level.

On behalf of the Government of Prince Edward Island, I would like to welcome all delegates of the Annual Meeting for the Canadian Acoustical Association to our fair Island.

As you join to discuss common issues and concerns, may you enjoy this opportunity to share your technical knowledge, experiences, and practices. I hope you will take advantage of your time in Prince Edward Island by visiting one of our many excellent restaurants, world-class golf courses, quality shops, entertainment venues, or perhaps a scenic drive along a country road to view the brilliantly colourful landscape of our province in October.

Please accept my best wishes for an effective meeting. I trust this visit to Prince Edward Island will provide you with the chance to enjoy our province and capture a small sample of the Island way of life.

Au nom du gouvernement de l'Île-du-Prince-Édouard, j'aimerais souhaiter la bienvenue à tous les délégués qui participeront à la réunion annuelle de l'Association canadienne d'acoustique qui se tiendra sur notre belle île.

À l'occasion de la rencontre qui vous permettra d'aborder les préoccupations et les problèmes communs, puissiez-vous en profiter pour partager vos connaissances techniques, vos expériences et vos pratiques. J'espère qu'au cours de votre séjour à l'Île-du-Prince-Édouard vous pourrez visiter l'un de nos nombreux et excellents restaurants, nos terrains de golf de classe mondiale, nos magasins de qualité, nos salles de spectacle ou encore faire une jolie balade le long d'une route de campagne afin d'observer le magnifique coloris du paysage de notre province en octobre.

C'est mon vœu le plus cher que votre rencontre soit des plus fructueuses et que votre visite à l'Île-du-Prince-Édouard vous permette d'apprécier notre province et de prendre un échantillon du mode de vie de l'Île.

Pat Binns
Premier of Prince Edward Island
Le premier ministre de l'Île-du-Prince-Édouard

UNIVERSITY *of*
PRINCE EDWARD ISLAND

550 University Avenue
Charlottetown
Prince Edward Island
Canada C1A 4P3

September 10, 2002

Welcome delegates,

On behalf of the University of Prince Edward Island (UPEI), I welcome delegates to the Annual Meeting of the Canadian Acoustical Association / l'Association Canadienne d'Acoustique, held for the first time in Charlottetown. I am pleased that through involvement of faculty, staff and students at UPEI your meeting is taking place in Prince Edward Island.

Your association is impressive in its fostering of education, research, and enterprise in the all-important field of the science of sound. The exposure to this extraordinary field that your meeting offers provides a once-in-a-lifetime opportunity for those associated with UPEI and the Province of PEI. In its quality and breadth, the scientific agenda is outstanding, and I am pleased to see faculty, students, and staff from our Arts, Science and Veterinary Medicine programs participating alongside experts in Government, Industry, and Universities from across Canada and beyond.

The University of Prince Edward Island is one of Canada's GREAT small universities. We are proud of its research achievements and contributions. This year, UPEI has taken the lead in two large federally funded multi-disciplinary, multi-institutional projects that involve acoustics. At UPEI, students are at the heart of everything we do. It is clear that your Association also places great value on students, offering the kind of individual attention, opportunity, and encouragement that we also prize.

It is therefore with utmost sincerity that I wish you the best for your meeting and thank you for joining us on PEI.

Sincerely,

WL MacLauchlan

H. Wade MacLauchlan
President and Vice-Chancellor

Office *of the* President

Tel: (902) 566-0400
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Acoustics: Bridge to the Future

Annual Meeting of the Canadian Acoustical Association, Charlottetown, October 9 -11, 2002

Annabel J. Cohen, Convener
Dept. of Psychology, University of Prince Edward Island, Charlottetown, PE, C1A 4P3, acohen@upei.ca

1. WELCOME

For the first time, Prince Edward Island is host to the Annual Meeting of the Canadian Acoustical Association/ l'Association Canadienne d'Acoustique (CAA-ACA). It has been more than 12 years since the meeting has taken place in Atlantic Canada. . What a special opportunity for people in the Maritime provinces to attend the meeting and for others "from away" to visit It has been an exciting challenge to serve as convener for this meeting. It is a privilege to welcome you all and to introduce these *2002 Conference Proceedings*.

2. CONFERENCE THEME

Progress in acoustics improves our lives in so many ways. Undeniably, acoustics provides a bridge to the future. Thus, *Acoustics: Bridge to the Future* supplies the conference theme, taking its inspiration from the Confederation Bridge – linking PEI to the mainland- and creating one of Canada's greatest engineering feats and architectural spectacles. The conference program shows that knowledge of acoustics forges new paths in domains as diverse as human communication, classroom design, noise control, audio engineering, audiology, underwater defence and exploration, and music perception. The 2002 Meeting reflects the diversity of acoustics disciplines, and the industrial, university, and government settings where activities in acoustics occur.

Regional Emphasis

Several themes close to the spirit of the province of PEI help to make the programme unique. Education is highly valued and a special session on classroom acoustics has been organized by CAA/ACA President John Bradley, of IRC, National Research Council, at the informal request of a member of the PEI Department of Education, Cheryl Perry. The University of Prince Edward Island (UPEI) is home to the only Veterinary College in Atlantic Canada, and animal welfare is also of great concern to Islanders. Representing this theme is a session on animal bioacoustics organized by Dr. Cathy Ryan of the Department of Psychology UPEI. Island life means proximity to the sea and two excellent sessions on underwater acoustics have been organized by Francine Desharnais and John Osler of Defence R& D Canada -Atlantic. Consistent with the importance of music to Island life and culture are sessions on Music Perception and Cognition, as well as musical performances presented throughout the meeting. Tranquility is one of the attractions of the Island, and several sessions on noise control and on environmental noise remind us of the importance of the commodity of quietness.

Keynote Addresses and Plenary Sessions

Highlighting the meeting are the keynote addresses presented by two eminent scientists with strong Atlantic ties. Dr. Harold Merklinger will open the meeting with the delivery of the first plenary lecture. He is the recently retired Director General of Defence Research and Development Canada - Atlantic (formerly DREA) and is an expert in the field of underwater acoustics. Dr. Floyd Toole will deliver the plenary lecture on the final day of the meeting. He is currently the Corporate Vice President Audio Engineering of Harman International Industries Inc. Dr. Toole, who was born in New Brunswick, was a scientist for many years in the Acoustics and Signal Processing Group at the National Research Council in Ottawa prior to his current position.

A third plenary session describes six multidisciplinary projects involving acoustics, recently made possible through the Canada Foundation for Innovation. Elsewhere on the program, another new collaborative program in Culture, Communication and Information Technology directed by William F. Thompson is announced. The conference program in general provides abundant evidence of acoustical bridges to the future and owes much to all those who contributed their work for presentation at the meeting.

Students

Students are the key bridge to the future through acoustics. A primary goal of the CAA/ACA is to foster education in acoustics. To this end, student participation in the meeting is encouraged and is subsidized. There are opportunities for students to win prizes for their presentations and to receive travel grants. Approximately 20% of the papers presented on the program are by graduate and undergraduate students from across Canada.

Exhibits

The conference provides a forum for exhibitors to demonstrate acoustic instrumentation and for attendees to speak directly to those who know most about this equipment. These products and innovations facilitate discoveries in and applications of acoustics. The participation of the exhibitors is greatly appreciated.

Canadian Acoustical Standards

A special session on acoustical standards has been organized by Cameron Sherry, and a meeting of the Canadian Standards Association Section Z107 on Noise will also take place. As well, a meeting of the Industrial Noise Subcommittee has been organized by Tim Kelsall.

Workshops/Seminars

The technical program exposes the latest developments in research in many fields of acoustics. Understanding this work, however, often demands more than a basic level of expertise. Education in acoustics is not widely available in schools and universities. To provide the fundamental acoustical knowledge underlying the new developments, several workshops/seminars before and after the meeting have been scheduled.

The NRC Institute for Research in Construction has mounted a workshop on noise and fire containment that is travelling to sites across Canada. The organizer, Dr. David Quirt, kindly accommodated their schedule to fit with that of the Canadian Acoustical Association. A second workshop preceding the meeting is being offered by Dr. Doug O'Shaughnessy of INRS Telecommunications, Université de Québec on Automatic Speech Recognition and Synthesis. Following the end of the meeting, a hands-on workshop on Humdrum, a method for statistical analysis and quantification of music is being presented by Bret Aarden, who has studied at Ohio State University with David Huron, the originator of this software.

The Conference Proceedings

This *Proceedings* volume of *Canadian Acoustics* devoted to the Annual Meeting opens with the papers of the two Keynote speakers and is followed by papers on the multidisciplinary CFI research projects involving acoustics. Papers are then grouped according to major subdisciplines of acoustics, beginning with Architectural Acoustics (sections on Building Acoustics, and Classroom Acoustics), Noise, Underwater Acoustics, Signal Processing, Acoustical Standards, Hearing Conservation, Animal Bioacoustics, Psychological Acoustics (including a cluster of papers emphasizing aging), Speech Sciences and Music Perception/ Cognition.. The *Proceedings* will provide a lasting memory and will enable those who cannot attend to share much of the content of the meeting.

3. ACKNOWLEDGEMENTS

Due to the contribution of many people, the PEI- 2002 CAA/ACA Annual Meeting promises to be intensive and professionally rewarding. May you find it so and enjoy it to the utmost. Special thanks is offered to Dr. David Stredulinsky, of DRDC-Atlantic, who served with me as co- chair of the Technical Program. Success of the program is in large part attributable to Dave's expertise in acoustics (e.g., noise, underwater, signal processing), his proficiency with computer technology, computer graphics, web-design and data-base management, his meticulous work, reliability, and tireless efforts. Many of the materials he developed for the web-site were designed with sustainability in mind and will benefit the society for years to come.

Dr. Ramani Ramakrishnan, Editor-in-Chief of *Canadian Acoustics* cooperated in publishing materials related to the meeting, including the publication of the advanced Abstracts Supplement. He was also most accommodating in regard to my guest editing these *Proceedings*. Assistance with translation was provided voluntarily by Reina Lamothe, Francine Desharnais and Stephane Dedieu and by David MacFarlane, (CIAF Université de Moncton, Shippegan). Several persons already mentioned organized extensive sessions, helping to ensure that Canada's smallest province could host one of its most exceptional meetings. Added to this list are the NRC Institute for Microstructural Sciences, GAUS at the University of Sherbrooke (particularly Nourredine Attala), and DRDC Atlantic (led by David Chapman). Assisting with communications was Robbie Arrabito of DRDC Ont.

The success of the meeting owes much to the University of Prince Edward Island. UPEI Department of Psychology Chair, Dr. Paul Boudreau set the tone of respect for the meeting and enabled Ms. Carol MacDonald to take responsibility for the registration, greatly assisted by Susan Doucette. The timely production of the advance Abstract Supplement and the *Proceedings* was attributable to the extraordinary expertise, versatility and care of Robert Drew of the UPEI Music Perception and Cognition Research Laboratory. Rob is acknowledged also for his guitar performance at the meeting and audiovisual assistance. Shawn King and Jay Macphail of the UPEI Audiovisual Dept., under the supervision of Dave Cairns, are thanked for audiovisual support and provision of equipment. UPEI also provided a generous donation through the Office of the President, Wade Maclauchlan. UPEI has also been home to meetings of the Island Acoustics Society over the years. Participants in the Society greatly helped in building the bridge from the CAA to Charlottetown.

Annabel Cohen, Ph. D.
Convener CAA/ACA 2002-PEI
Guest Editor, *Proceedings Issue of Canadian Acoustics*

Acoustique : Un Pont Vers L'Avenir

La Réunion Annuelle de l'Association Canadienne d'Acoustique Charlottetown, Octobre 9 -11, 2002

Annabel J. Cohen, Présidente de conférence
Dépt. de Psychologie, Université de l'Île-du-Prince Édouard, Charlottetown, ÎPÉ, C1A 4P3, acohen@upei.ca

1. BIENVENUE

Pour la première fois, l'Île de Prince Édouard sera l'hôte de la réunion annuelle de l'Association canadienne d'acoustique (ACA). Il y a plus de 12 ans depuis que la réunion a eu lieu au Canada atlantique. Quelle belle occasion pour les personnes des provinces d'assister à la réunion, et pour les autres, qui viennent de « loin », c'est une occasion opportune pour visiter. Cela a été un défi passionnant de servir en tant que présidente de la conférence. C'est un privilège de vous souhaiter la bienvenue et de présenter ces Cahiers des Actes *2002*.

2. THÈME DE LA CONFÉRENCE

L'acoustique améliore nos vies de beaucoup de façons. Bien sûr, l'acoustique fournit un pont vers l'avenir. Ensuite, *L'Acoustique : un Pont vers l'avenir* fournit le thème de la conférence, inspiré par le Pont de la Confédération qui lie l'ÎPÉ au continent et dont la création a été l'un des accomplissements d'ingéniérie le plus magnifique du monde et est une splendeur architecturale. Le Programme technique montre que la connaissance acoustique crée de chemins nouveaux dans les domaines divers, par exemple, la communication humaine, le design dans les salles de classe, le bruit, l'ingénierie auditif, l'audiologie, l'acoustique sous-marine, et la perception musicale. La réunion de l'année 2002 représente la diversité des disciplines acoustiques et les endroits de l'industrie, de l'université, et du gouvernement où les activités acoustique ont lieu.

L'Accent Régional

Quelques thèmes proche de l'esprit de la province de l'ÎPÉ aident à rendre le programme unique. L'éducation est d'une valeur énorme et il y a une session spéciale au sujet de l'acoustique de la salle de classe, organisée par le président de le l'ACA John Bradley, du CNRC, d'après la requête informelle d'une membre du ministère de L'Éducation de l'ÎPÉ, Cheryl Perry. C'est à l'Université de l'Île du Prince-Édouard (UÎPÉ) que se trouve le seul collège vétérinaire au Canada Atlantique et le bien-être des animaux touche plusieurs des habitants de l'ÎPÉ. En accord avec ce thème, il y a une session sur le sujet de bioacoustique animale, organisée par la Dr. Cathy Ryan du Département de la Psychologie de l'UÎPÉ. En ce qui concerne les environnements maritimes, il y a deux sessions qui traitent d'acoustique sous-marine et qui sont organisées par Francine Desharnais et John Osler de DRDC- Atlantique. En ce qui concerne l'importance de la musique à la vie et à la culture de l'ÎPÉ, il y a deux sessions touchant la perception et la cognition de la musique. Aussi, quelques spectacles musicaux ont lieu au cours de la réunion. La tranquillité est une des attractions de l'ÎPÉ et quelques sessions au sujet du contrôle du bruit et du bruit environnemental nous rappellent l'importance de la quiétude.

Les Séances Plénières

Deux points culminants du congrès sont fournis par deux scientifiques éminents, liés fortement aux provinces Atlantiques. Dr. Harold Merklinger ouvrira le congrès en prononçant la première conférence. Il a pris sa retraite tout récemment du poste de Directeur-général de DRDC-Atlantique (auparavant DREA). Il est un expert en acoustique sous-marine. Dr. Floyd Toole ouvrira la journée finale du congrès. Il est le Vice-président de l'Entreprise de l'Ingénierie Audio de Harman International Industries, Inc. Dr. Toole, qui est né au Nouveau-Brunswick était scientifique il y a plusieurs années auprès du Groupe de l'Acoustique et du traitement du Signal au CRN à Ottawa avant de commencer sa poste actuelle.

Une troisième séance plénière décrit six projets multi- disciplinaires qui touchent l'acoustique, appuyés par la Fondation canadienne de l'innovation (FCI). Par ailleurs, on annonce un autre programme collaboratif de Culture, communication et technologie informatique (CCIT) du Dr. William F. Thompson. Généralement, le programme technique offre une évidence abondante des ponts acoustiques vers l'avenir et le programme doit beaucoup à tous ceux qui ont contribué au congrès.

Les Étudiants

Les étudiants sont le pont primaire vers l'avenir par l'intermédiaire de l'acoustique. Un but fondamental de l'ACA est de promouvoir l'éducation dans les disciplines acoustiques. À cet effet, l'Association encourage et subventionne la participation des étudiants. Il y a des occasions de gagner des prix pour les présentations et de recevoir des bourses de voyage. Environ 20% des communications présentées au congrès sont signées des étudiants d'à travers du Canada.

L'Exposition

Le congrès fournit un forum pour des exposants qui peuvent montrer l'instrumentation acoustique. C'est une occasion pour les délégués de parler avec les personnes qui ont beaucoup d'information à propos de cet équipement. Ces produits et les innovations facilitent les découvertes acoustiques. La participation des exposants est appréciée beaucoup.

Normes Canadiennes

Une rencontre du comité sur les normes canadiennes en acoustique Z107 est organisée par Cameron Sherry.

Les Ateliers

Le programme technique fournit une occasion d'en apprendre sur des développements des plus récents de la recherche dans beaucoup de disciplines acoustiques. La compréhension de ce travail, cependant, demande souvent plus qu'une connaissance de base. L'accessibilité de l'éducation acoustique est souvent faible dans les écoles et les universités. Pour fournir la connaissance fondamentale qui est à la base des développements nouveaux, on offre quelques ateliers au début et à la fin du congrès principal. Un atelier est offert par IRC/CNRC sur la résistance au feu et l'insonorisation entre les unités de domiciles multifamiliaux. Un autre atelier est offert par le Dr. Doug O'Shaughnessy de INRS Télécom, Université du Québec à Montréal sur le sujet de la reconnaissance et la synthèse automatique de la parole. Bret Aarden de The Ohio State University offre un atelier pratique de Humdrum, une méthode d'analyse statistique et de quantification de la musique.

Les Actes du Congrès

Ces Actes du Congrès commencent avec les contributions des allocutions spéciales de Harold Merklinger et Floyd Toole. Suite à cela, sont présentés les projets multidisciplinaires FCI. Ensuite sont traités les sujets de l'acoustique architecturale, le bruit, l'acoustique sous-marine, le traitement des signaux, les normes en acoustique, la conservation de l'audition, la bio- acoustique animale, l'acoustique psychologique, les sciences de la parole, et la perception et la cognition de la musique. Les actes offriront un souvenir permanent et permettront à ceux qui ne peuvent pas y assister de prendre connaissance du contenu de la réunion.

3. REMERCIEMENTS

Bien sûr, à cause des contributions de beaucoup des personnes, la Réunion annuelle à l'ÎPÉ-2002 promet d'être une expérience à la fois très utile, intensive, et agréable. Des remerciements spéciaux sont offerts au Dr. David Stredulinsky (de DRDC-Atlantique) qui a partagé avec moi les responsibilités de co-ordonner le programme technique. Le succès du programme est en grande partie attribué à sa compétence acoustique, à sa compétence avec la technologie d'ordinateur, avec les graphiques d'ordinateur, avec le dessin du web, et avec l'administration de la base de données; son travail méticuleux et ses efforts infatigables. L'Association peut utiliser ses additions au site-web pour les congrès futurs.

Dr. Ramani Ramakrishnan, Rédacteur-en-chef d'Acoustique canadienne a été très coopératif. Il a publié beaucoup des renseignements touchant le congrès, y compris la brochure des Abstraits au mois de juin (une innovation de cette année). Il m'a donné généreusement l'occasion d'être la rédactrice invitée pour ces *Actes du Congrès*. On doit reconnaître l'assistance avec les traductions -pour des annonces du congrès (Reina Lamothe, Francine Desharnais, et Stephane Dedieu) et pour cette article de ces *Actes* (David MacFarlane, du CIAF, U de Moncton, Shippagan). Quelques personnes déjà mentionnées ont organisé des sessions spéciales, s'assurant que la province la plus petite du Canada a pu créer un congrès exceptionnel. À cette liste, on doit ajouter IMS/ CNRC, GAUS (l'Université de Sherbrooke, particulièrement Nourredine Attala), et DRDC-Atlantique (dirigé par David Chapman). Robbie Arrabito de DRDC- Ontario a assisté avec la publicité aux listes électronique.

Le succès du congrès doit beaucoup à L'Université de l'Île-du-Prince-Édouard (UÎPÉ). Le directeur du Département de Psychologie de l'UÎPÉ, le Dr. Paul Boudreau, a permis à certaines personnes d'œuvrer pour le congrès et il rendu possible l'aide exceptionnelle de Mme Carol MacDonald pour les inscriptions. L'aide de Susan Doucette avec cela et avec *les Actes* était aussi indispensable. La création du supplément des Abstraits et de ces *Cahiers des Actes* est attribuée à la compétence extraordinaire, la versatilité, et le soin de Robert Drew du Laboratoire de recherches dans la Perception auditive et de la Cognition de Musique de l'UÎPÉ. On doit remercier Rob pour avoir jouer de la guitare pendant le congrès et aussi pour son assistance audiovisuelle. Je remercie Shawn King et Jay Macphail du Dépt. de l'audiovisuel de l'UÎPÉ sous la direction de David Cairns, pour la provision d'appui audiovisuel et d'équipement. L'UÎPÉ aussi a fourni une donation généreuse de la part du bureau du Président de l'UÎPÉ, Wade Maclauchlan. Depuis des années, l'UÎPÉ accueille les rencontres de l'Association d'acoustique de l'Île. Les participants de cette Association ont beaucoup aidé dans la construction du pont de l'ACA au Congrès à Charlottetown.

Annabel J. Cohen, Ph. D.
Présidente du Congrès ACA 2002-ÎPÉ
Rédactrice Invitée, *les Cahiers des Actes*

PROCEEDINGS OF THE ANNUAL MEETING OF THE CANADIAN ACOUSTICAL ASSOCIATION

ACOUSTICS: BRIDGE TO THE FUTURE / ACOUSTIQUE: UN PONT VERS L'AVENIR

LES ACTES DE LA RÉUNION ANNUELLE DE L'ASSOCIATION CANADIENNE D'ACOUSTIQUE

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COLD WAR ACOUSTICS

Harold M. Merklinger
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INTRODUCTION

In the context of the theme of this meeting, an appropriate subtitle might be "A Bridge from the Past".

Circumstances around the end of World War II and the early post-war years conspired to precipitate a surprisingly large burst of research activity in acoustics that lasted about 50 years. I expect that many acousticians working in other branches of the field have wondered what all the underwater acoustics effort was about. I hope this paper will help to explain.

Certainly there has been military interest in underwater acoustics ever since Canadian-born Reginald Fessenden demonstrated an early underwater echo location system shortly following the sinking of the TITANIC. Both passive and active sonar were used during World War I to find submarines and this work was re-born with the onset of World War II. Yet even during WWII, research efforts in underwater acoustics fell far short of the level of effort yet to come. One of the truths of the WWII experience was that active sonar fell far short of expectations in dealing with submarines. The big breakthroughs in winning the Battle of the Atlantic were the breaking the German naval code, radio direction-finding on an ocean-basin scale, and the invention and application of RADAR.

So what happened during the Cold War? I'll try to answer that question, but please understand that this is just one man's opinion, and not the result of extensive historical investigation.

KEY FACTORS

I believe there were three main circumstances—perhaps four—that triggered the events that were to follow. I will outline each of these factors, indicate how they interacted, and then relate a bit about the historical consequences. While my prime point of view is that of a Canadian, I expect that this contest was quite symmetric: the view from the Warsaw Pact countries was probably remarkably similar.

1. <u>The State of German Technology</u>. The collapse of Hitler's Germany revealed to "The Allies"—including Britain, the USA, the USSR and Canada—the surprisingly advanced state of German military technology in a number of areas. And the Allies did what they could to absorb this knowledge while at the same time depriving Germany of it. The significant technologies relevant to this story are rocketry and a fundamental transformation in Submarines. Until the end of WWII, submarines were basically small surface warships that could make themselves hard to find by diving under the water for relatively short periods of time. The invention of the snorkel and early forms of air-independent propulsion—nuclear power is one, but there were and are others—transformed the submarine into an underwater warship that sometimes had to come to the surface. Furthermore, a craft designed primarily to operate fully submerged (rather than designed for stability on the ocean surface) could be much faster. And rocket technology would give this new generation of submarines long range weapons: ultimately weapons that could reach almost any city in the world from a launch point in the ocean.

2. <u>The State of Sonar</u>. As I have already stated, sonar had not really proven to be all that effective. It was really more of a deterrent: something that might scare off the submarines. Yet sound was just about the only hope for detecting these new more capable naval craft. WWII experience with passive sonar was promising, as was allied experience in using small explosives—and sonar—to find downed air crew. New oceanographic knowledge—spurred by the poor experience with sonar—helped to explain what was going wrong and how sonars might be fixed. There was cause to believe that sonar could be made to work.

3. <u>The Commencement of the Cold War.</u> At the end of WWII, the allies were on moderately friendly terms, though opinions certainly varied. The revelations of Igor Gouzenko (serving with the USSR Embassy in Ottawa) demonstrated, however, that the USSR regarded the capitalist allies very much as potential enemies. The revelations also strongly suggested a very aggressive posture for a Communist domination of post-WWII society. Furthermore both sides in this new contest had access to the German technology and there was a strong fear the development work would continue. And, of course, it did. And so defences also needed to be developed to counter these new threats.

4. <u>The Advent of things Nuclear</u>—explosives and propulsion—can certainly be regarded as another major influence. I tend to regard them as more of a "multiplier" factor. Nuclear propulsion speeded the transition to near full independence from the atmosphere. And nuclear warheads for rocket-propelled missiles made those rockets

much more menacing. But I think that even without the nuclear age, submarines and missiles would still have posed a threat that merited attention.

THE REACTION

The actions that followed went something like this.

Fast, missile-capable nuclear submarines were developed and deployed by both sides. They were deployed full-time off our coastlines. We in turn undertook to determine where every one was and to be in position to neutralize it if that should become necessary. The net result was that the "Cold War", while no shots were fired, was nevertheless a very active, every-day, and highly resource-intensive affair that lasted for about four decades.

While acoustic detection of submarines is not the only system that works, it certainly is one of the more effective ways, and probably the most cost-effective way, to do the job.

In addition to matters directly related to sonar, one needed to be able to move pursuing systems at a pace that could 'keep up'. Aircraft were a suitable solution—they had already proven effective in WWII—but for a time fast surface warships were also seriously considered for this job. The Canadian hydrofoil program leading to the FHE-400 was part of this effort.

Active sonars mounted on the hulls of surface ships did not work well in the North Atlantic. The cause was sound refraction due to typical temperature and salinity conditions in the ocean, coupled with the fact that sound velocity increases with pressure and hence depth, if all other factors remain unchanged. Solar warming of water near the surface tends to cause a downward bending of the sound rays, though wind-driven mixing of near-surface water can lead to upward bending of the rays. The net result is typically short ranges for typical active sonar systems. Yet the uniform deeper waters coupled with the pressure effect and warmer water near the surface also permit very effective long-range propagation at frequencies below one or two kilohertz (well below typical WWII active sonar frequencies).

The obvious solutions to the submarine detection problem then are:

1. <u>Place the active sonar system deeper</u> than the bottom of a surface ship by placing it in a towed submerged vehicle: The Variable Depth Sonar. (Even, perhaps, one deployed from a fixed-wing airplane!) Placing the sonar very deep—miles deep—in the ocean looked promising and was deemed feasible for large warships, but for destroyer-sized ships that length of cable proved to be a pretty effective sea anchor.

2. <u>Use passive listening at lower frequencies</u>. From late in WWII this was done by using freely-drifting "sonobuoys" that suspend a hydrophone at moderate depth and radio the sonar data to an aircraft, ship or ashore. Later we learned how to tow low frequency hydrophones behind a ship without suffering too much noise in so doing. (Actually the oil exploration companies did it first, but their streamers were too noisy to use at desired warship speeds. Then again, such streamers were first tried for military purposes during WWI.) Throughout the Cold War period, the race was on to build ever quieter submarines and ever more sensitive detection systems to find them.

3. <u>Place the hydrophones on the continental margins</u> and cable the signals ashore. And do this extensively all along our coastlines.

4. <u>Lower the frequencies used for active sonar</u>. This is not as simple as it might first seem. Consider briefly that lowering frequencies means making things bigger and heavier - for the same design, weight will increase as the cube of the factor by which frequency is lowered! So a reduction in sonar frequency from 10 KHz to 1 KHz might be expected to result in a 1000-fold increase in the weight of the sonar transducers.

5. And, of course, one can do most of these things better <u>from another submarine</u> than from a surface ship.

6. Another enduring concern throughout the 20th century was how to <u>locate and avoid sea mines</u>. As the mines are generally located underwater, acoustics again comes to the fore.

The acoustic anti-submarine efforts also led to intensive efforts to understand the ocean environment - an effort that has essentially taken on a life of its own, with many side benefits.

Another related issue of Canadian concern was the possibility that missile-launching submarines might use the Canadian Arctic as a launch area. This concern sparked sonar and oceanographic interest in our northern waters.

Submarines did not constitute the only Cold War threat, of course. Similar cold war battles also took place in the air, on the ground in Europe, and in many warmer skirmishes elsewhere.

CONCLUDING REMARKS

With the end of the Cold War, military submarines remain a concern, but they do not pose the enduring threat that they once did. Today there are other military concerns —such as terrorism—that occupy our defence planners' thoughts. There undoubtedly remain many potential military applications of acoustics, but acoustics does not occupy the place of prominence that it did during the Cold War.

THE AUDIO INDUSTRY : THE STATE OF OUR SCIENCE AND ART

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1. INTRODUCTION

Music and movies are art. Audio is a science. Using science in the service of art is the essence of the audio industry. There is a substantial, and still growing, foundation of scientific knowledge behind most of the audio products with which we are familiar, in spite of some glaring exceptions in the marketplace. Some of the old myths persist. In understanding the psychoacoustic portion of the science, loudspeakers have been especially troublesome. These electromechanical devices are required to operate over 10 octaves, with a dynamic range of 105 dB or more, delivering sound to our ears through rooms that are completely unpredictable in their size, shape, layout and acoustical characteristics. It seems like an almost hopeless task yet, through a combination of factors, including good engineering, psychoacoustic knowledge, and human adaptation, we manage to derive substantial satisfaction from our audio systems. And they continue to get better.

In my years at the National Research Council, I worked on providing some of the answers to the underlying questions[1,2]. Others have contributed more data, to the point where, now, we can say that truly good sound reproduction is no longer a matter of chance.

2. SEPARATING THE VARIABLES

At this stage, electronic devices, including the better storage media, are – or can be - essentially transparent. Assuming that a perfect voltage waveform is delivered to the terminals of the loudspeaker from a low impedance source, the next challenge is to minimize all audible linear and non-linear distortions in the transduction process. Then we optimize form of the radiated sound field, bearing in mind the physical nature of the listening environment, and find ways of taming the prominent and lively resonances of small rooms. Let us look at some of the major variables in this complicated picture.

2.1 Non-linear distortion. Simultaneous masking by the audio signal itself prevents much of the perception of non-linear distortions. Not all, of course, and not all of it is bad. At low frequencies significant numbers of listeners react positively to a little added timbral 'richness'. Good conventional engineering can reduce non-linear distortions to acceptable levels. Exceptions are usually the result of compromises driven by cost.

2.2 Linear distortions. Conventional engineering principles encourage maintaining the integrity of both amplitude and phase in the complex transfer function. This preserves waveform information. However, abundant psychoacoustic evidence tells us that we humans are substantially 'phase deaf', especially when listening in normally reverberant spaces. Even in circumstances where one may hear a difference, allocating a preference is difficult. We encounter these situations regularly in normal listening, whenever the direct sound from a source is modified by the addition of one or more strong reflections. We routinely can recognize differences but, since we know the source is unchanged, we do not assign preferences. If waveform information were critical to sound fidelity, where would one place a microphone to capture the definitive waveform of a grand piano? There is no such waveform.

In contrast, humans are remarkably sensitive to very small changes in frequency response or spectrum. In terms of overall loudness, changes of program level less than about 1 dB are normally inaudible. However, the threshold of detection for a spectral tilt is about 0.1 dB/oct., a Q=1 resonance can be detected when it adds about 0.3 dB to an otherwise flat frequency response. Narrower bandwidth, higher Q, spectral changes are less easily heard, with Q=50 spikes reaching 10 dB before arousing our conscious reactions with certain kinds of music. Ringing resonance decays tend to be audible only at very low frequencies. If this all seems countintuitive, consider also that our sensitivity to medium- and low-Q resonances is lowest in anechoic conditions, and increases when we listen in a reverberant space! A concert in the park is timbrally enriched when rain drives the orchestra into the community hall. Loudspeakers sound less colored in anechoic chambers than they do in normal rooms [3].

In terms of technical measurements, all of this argues against gross $\pm x$ dB tolerances on frequency response curves, unless the tolerance is very small : e.g. $\pm$ 0.5 dB. A perceptual criterion requires that the allowable tolerance be related to the bandwidth of the deviation. Oh yes, and 1/3-octave resolution is woefully inadequate when it comes to describing what might or might not be audible as a timbral difference. Critical bands apply to loudness summation, not the perception of timbre.

2.3 Frequency response and directivity. So, we can hear very small differences in amplitude response. What, then, is the target curve from which deviations are assessed? Is 'flat' the ideal? For electronic devices it clearly is. For loudspeakers, it depends on what is being measured. Loudspeakers radiate a three-dimensional sound field. In rooms all of this sound reaches listeners, most of it after one or more reflections from room boundaries and furnishings. To evaluate the performance of a loudspeaker it is necessary to collect enough data to be able to reconstruct the major features of the sounds arriving at a listener in a room. Figures 1 thru 3 illustrate the essence of the loudspeaker/room interface problem.

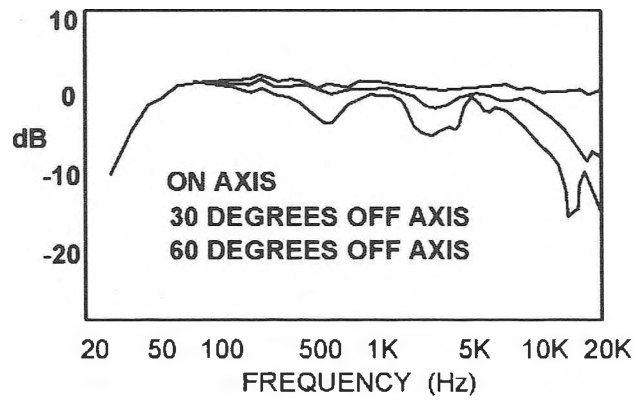

Fig.1 Anechoic frequency responses of a loudspeaker showing (top to bottom) a very smooth, flat, axial response and progressive deterioration off axis.

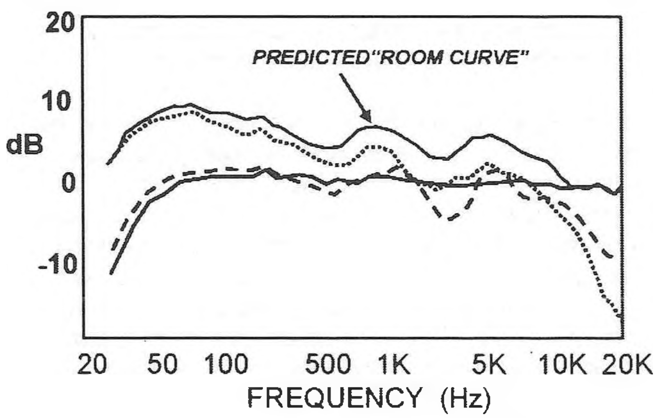

Fig. 2 The sequence of sounds arriving at a listener in a room. The first arrival is the direct sound (the solid line around 0 dB). Second is the sum of adjacent boundary reflections (dashed). Third is the reverberation, represented by total sound power (dotted). The solid curve plotted over them all is the energy sum of all three – a prediction of what might be measured in a real room.

It is evident from this that sound power is the dominant factor at low frequencies, and that the direct sound is dominant at the very highest frequencies. In between, over most of the frequency range, everything contributes. So, if the purpose of the measurements is to be able to anticipate loudspeaker performance in a room, it is necessary to measure everything. The on-axis response, by itself, is merely a start. The sound power is also incomplete evidence. All of it must be viewed as an ensemble.

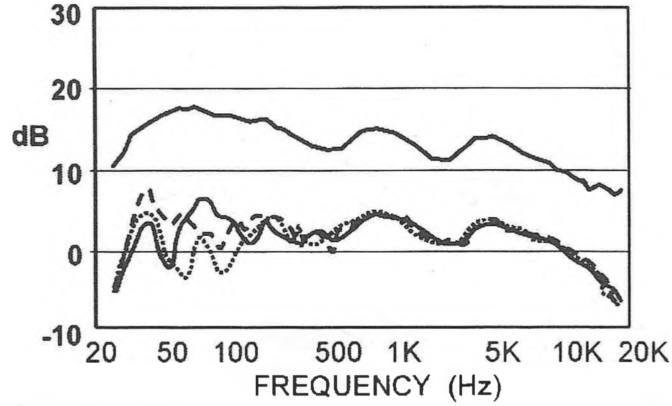

Fig. 3 The three bottom curves show steady-state measurements of the example loudspeaker in three typical locations in a normal room. The top curve is the predicted curve from Fig.2, raised 10 dB for clarity.

Fig.3 shows that, at frequencies below about 300-500 Hz, room resonances increasingly dominate what we hear. The differences are not subtle and the situation can be evaluated only by measurements in the room itself. However, above 300-500 Hz it is possible to predict with good accuracy what it delivered to the listening position from a collection of anechoic measurements that have been appropriately processed. Steady-state measurements in a room are reliable at low frequencies, but at middle and high frequencies they are useful only in conjunction with comprehensive anechoic data. In this example, the undulations at middle and high frequencies are in response to the frequency-dependent directivity of the loudspeaker, so it means that any attempts to change the shape of the room curve by equalization will, in fact, not correct the problem. The only solution to this kind of problem is a loudspeaker designed with directivity that is constant, or relatively so, over most of the frequency range. Only then will the direct, early reflected, and reverberant sounds convey similar timbral messages to the listeners [4].

It is a nice story, but do listeners agree? Yes. Hundreds of double-blind subjective evaluations, conducted over the past 20+ years, confirm that these are the loudspeakers that listeners award with the highest ratings. To get consistent opinions from listeners, however, it is necessary to deal with the huge variability at low-frequencies caused by standing waves and loudspeaker and listener locations. For the purposes of achieving consistency in listening tests, standardizing the room and locations is a practical solution. We currently use a pneumatic 'speaker shuffler' to achieve a consistent location for the active loudspeakers in listening tests [5]. Delivering consistently good sound to listeners in their homes is a much more formidable challenge.

3. AN AUDIO INDUSTRY PROBLEM

Audio enthusiasts tend to take for granted that everything upstream of the playback device is under control. The sad fact is that it isn't. Many factors in the sequence of events leading to a music recording or a film sound track contribute to systematic and random variations in the final product. Not the least of these are the humans involved in the process, but physical factors also have a say.

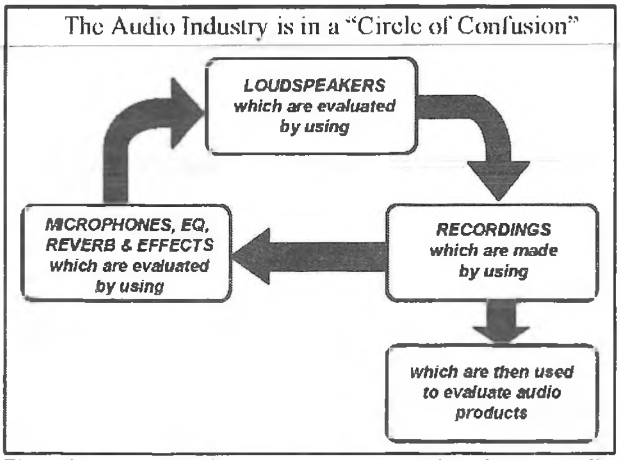

Fig. 4 Loudspeakers in rooms are the means by which recordings are judged while being made. They are the 'window' through which the art is viewed. If the window is colored or distorted, the art will be adjusted to compensate. The compensated art is then used for enjoyment (through different loudspeakers and rooms) or, worse, used as a basis for evaluations of other audio products.

Fig. 4 shows that our audio industry is trapped in a "circle of confusion" that can only be broken if there is a reliable similarity between the loudspeakers and rooms used in the production of the art, and those used during playback for our entertainment. Such consistency requires accurate technical measurements that show good correlation with listener preferences. These now exist.

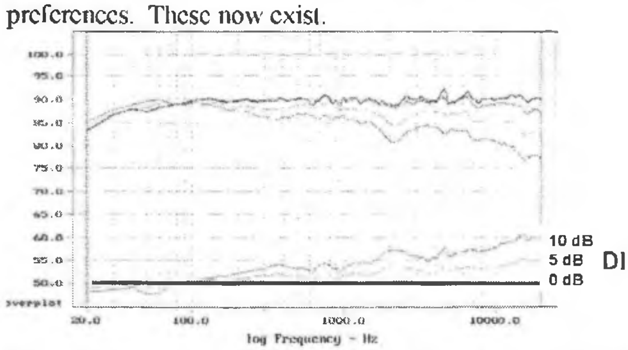

Fig. 5 Seventy-two anechoic measurements, made on horizontal and vertical orbits around a loudspeaker, processed to show (top to bottom): on-axis (direct sound), Average response within a ± 10° vertical, ± 30° horizontal listening window, estimated energy sum of the first six reflected sounds in an average room, and total sound power. At the bottom are the traditional directivity index (top) and an invented one for early reflections only. The measurements have 1/20-octave resolution.

Fig. 5 shows a form of measurements that has been found to correlate well with listener opinions as expressed in double-blind evaluations in normal rooms. The product described here is a good representation of the state of the art in loudspeakers today. It costs $10K/pr. (USD). In subjective evaluations of the best loudspeakers, it is common for the largest sources of judgment variation to be the recordings themselves, and individual differences among listeners.

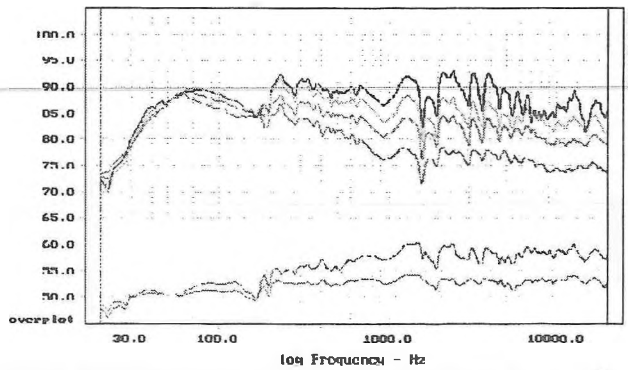

Fig. 6 A $11K/pr. product that was compared to the one in Fig. 5.

However, individual preferences are a factor only when the contests are very close. The products of Figs. 5 and 6 were evaluated by 124 unselected, untrained, listeners. All but one put this loudspeaker in a strong second place, and this person's preference was not statistically significant. There are two important lessons here: (1) most people DO agree on what is good if they are given an unbiased opportunity to judge, and (2) price is an unreliable indicator of sound quality (so, also, are the reviewers who raved about the product in Fig. 6). Our routine listening evaluations use persons selected for normal hearing and put through training to identify those with the necessary aptitudes (most people) and to increase their skills in detecting common faults and articulately describing what they hear. They become remarkably stable 'measuring instruments'[6].

The example shown in Fig. 6 is not uncommonly bad, but the good news is that more and more loudspeakers are emulating the performance of Fig. 5, even at affordable prices. Sacrifices at lower prices include the lowest bass frequencies, the ability to play cleanly at high sound levels, and visual aesthetics. The other good news is that there are a few professional studio monitor loudspeakers that bear comparison with Fig. 5. The standards within the audio industry, both consumer and professional, are rising.

Still, recordings remain more variable than we would like. In examining the loudspeakers used as professional monitors, one finds a range of sound quality only slightly less than that in the consumer domain. This is regrettable. However, some of the variable art comes from sources using the same good loudspeakers, so what is wrong? A recent investigation surveyed a large number of recording studios that used the same family of loudspeakers. Measurements were made at the head location of the recording engineers, and the data were compiled. The results were frightening.

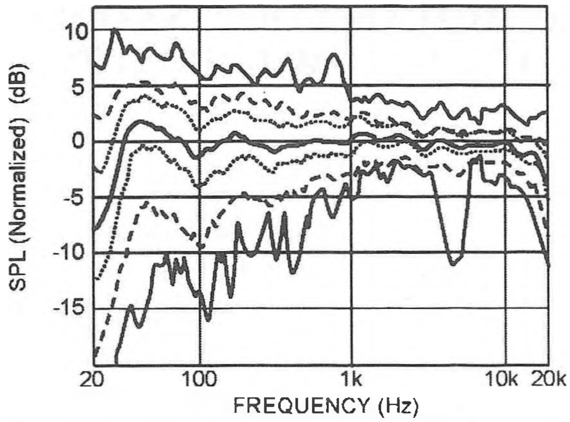

Fig. 8 The solid curve approximating the 0 dB line is the median of 250 measurements. This is the good news. The solid curves at the top and bottom represent the max/min limits of all of the measurements. The dashed curves show the upper and lower limits of 90 % of the measurements, and the dotted curves show the same for 50% of the measurements [7].

The picturesque character of the 250-curve median displays the ability of statistics to shield us from the truth. Only when we see the huge variations extant in individual studios do we see reasons why artists can be deceived about what is going into their master tapes and discs. Looking further, the largest variations are in the low frequency range where the room is in control. The majority of installations show relatively good performance in the middle and upper frequencies where the loudspeaker dominates. Correcting this situation, and the parallel situations in all of our homes, means coming to grips with room resonances, methods of measurement and equalization, and some new ways to employ multiple subwoofers in acoustic mode-canceling arrays.

A casual glance at a high-resolution curve measured in a small room suggests that perfection may be forever elusive. However, at low frequencies the dimensions of the standing waves are generous, the events are less numerous, the identification of individual resonant modes is possible, and solutions begin to present themselves. Damping with mechanically- or acoustically-tuned absorbers (or flexible walls) reduces the standing wave peak-to-trough ratio, producing more uniform bass over larger areas of the room. Equalization for specific listening locations is possible, so long as one addresses only the peaks of the resonant modes. Low-frequency room modes behave as minimum-phase systems and flattening the frequency response also eliminates the time-domain ringing. Doing this successfully requires high-resolution measurements to show the true center frequency and Q of the resonances, and parametric equalizers to match the shapes. Traditional 1/3-octave measurements and equalizers are not adequate.

Now that multichannel audio allows us to treat the frequency range below 80 Hz with dedicated subwoofers, they can be located to optimize bass performance, and it is possible to use multiple subwoofers to destructively drive modes or drive them at their pressure minima. For example, two woofers located at the 25% and 75% points across the 20-foot width of a room will seriously attenuate all width modes below 80 Hz. Many of us have probably done this accidentally in stereo setups, and it works because most low bass is monophonic.

This principle can be extended to a set of general solutions for rectangular rooms, where the objective is to minimize the variations in low-frequency performance over the central portion of a room, where several persons can enjoy multichannel performances of music or movies. It turns out that more than one subwoofer is needed, but more than four are not advantageous. The best of the practical arrangements are four subwoofers in the corners, in the mid-wall locations, or the 25% and 75% locations on front and back walls. Two in opposite mid wall locations are almost as good. Once relatively uniform performance is achieved over an area, intelligent equalization can be applied [8].

So, if we diligently apply the existing science, it can yield more consistently good recordings, and it can ensure that the finer qualities of the audio arts can be reliably delivered to our homes. Science, truly, can be used in the service of art.

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Relevant technical papers are also downloadable from www.harman.com, under "white papers".

CENTRE FOR RESEARCH ON BIOLOGICAL COMMUNICATION SYSTEMS (CRBCS): AN INTEGRATED SYSTEMS APPROACH TO COMMUNICATION

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Thanks to generous grants from the Canada Foundation for Innovation, the Ontario Innovation Trust Fund, and the City of Mississauga, I am pleased to announce the establishment of a Centre for Research on Biological Communication Systems (CRBCS), with research facilities located at the Mississauga campus of the University of Toronto (UTM), Queen's University, and Sheridan College. This Centre will provide researchers with state-of-the-art facilities to probe the nature of human and animal communication systems across the lifespan at levels ranging from gene expression to the development and organization of social behaviour.

The Centre's philosophy is to take an integrated systems approach to the study of human and animal communication. Biological communication systems are interactive and multimodal. Whereas most researchers tend to focus their efforts on one aspect of human or animal communication, and usually restrict themselves to a single information channel, researchers at CRBCS will be encouraged to study biological communication as an integrated system that is developing and changing across the lifespan. Hence, we are building research facilities and expanding our research complement so that we can 1) pursue a lifespan approach to human communication that takes into account its multimodal nature, 2) determine the effects of new technologies on human communication systems, and 3) trace the effects of genetic contributions to communication systems through to behaviour in animals and humans. Five research facilities will work together to obtain these goals.

1.0 HUMAN COMMUNICATION RESEARCH FACILITY (UTM)

Currently under construction (completion date, January, 2004) is a new 2000 m^2 research facility that will occupy the top floor of the new Communication, Culture, and Information Technology Building at UTM (partially funded by Ontario's SuperBuild Fund). This research facility will include: signal-processing equipment to measure, record, store, analyze, and process sounds recorded in a natural environment, equipment to provide virtual simulations of these environments; sound-attenuating chambers and associated equipment to be used for testing human-to-human and human-machine communication; a multimodal virtual-reality test station (identical to one at the Queen's Multimodal Research Facility); test stations to monitor attentional behaviour (e.g., eye movements) in studies of reading, reading development, and developmental dyslexia; test stations to assess age-related changes in visual processing of information; and video capturing and image processing equipment to be used in the development of instructional modules for use in second-language acquisition.

2.0 ANIMAL COMMUNICATION RESEARCH FACILITY (UTM)

The animal communication research facility will occupy newly renovated facilities (completion date, summer, 2004) which will consist of dedicated facilities for surgical, histological, molecular, genetic, and behavioral work, along with excellent outdoor "fields," and confocal and biotechnology facilities. As well, UTM has a first-class, well-serviced, small-mammal animal vivarium which is being expanded to accommodate small mammals for genetic research. These facilities will permit researchers to investigate communicative behaviour at a level that is often not possible in the study of human communicative interactions. By linking animal and human geneticists with sensory physiologists and behavioural scientists, we hope to identify the genes that contribute to the development and expression of communicative behaviours in both animals and humans, trace their influence on neural development, neurochemistry, and sensory functioning, and determine how their differential expression affects communication and social interaction.

3.0 QUEEN'S MULTIMODAL COMMUNICATION RESEARCH FACILITY

A new research laboratory (completion, summer, 2003) will give this facility a unique capacity for simultaneously monitoring several communicative behaviours, and for testing audiovisual integration during communication. Specifically, researchers in this lab will be able to measure 3D movements of the vocal tract in adult humans, measure and analyze facial movements, and monitor body, head, and eye position in both humans and animals. Moreover, Silicon Graphics computers and associated hardware and software will allow us to produce the high definition audio and visual stimuli required in studies of animal and human communication. Combined audio-visual tests stations and associated soundproof rooms will permit us to examine how the auditory and visual channels are integrated in face-to-face communications. Working in cooperation with the geneticists and neuroscientists at Queen's and UTM, this equipment will also be used to study communication in clinical populations where there are genetic bases to communication disorders.

4.0 VISUALIZATION DESIGN INSTITUTE (SHERIDAN)

The Visualization Design Institute (VDI) is dedicated to the pursuit of excellence of design in the field of computer visualization and simulation. VDI has the equipment and expertise that will allow CRBCS researchers to incorporate new animation, auralization, and visualization techniques into virtual-reality systems to be used in studies of communicative interactions between humans, and between humans and machines. Working in cooperation with researchers associated with the Human Communication Research Facility, and the Sheridan Elder Research Centre (see below), the VDI will also be involved in studies evaluating the impact and potential usefulness of new communication technologies in addressing the communication needs of special populations such as the elderly. By working together we will be able to *customize* systems to accommodate the psychological and physiological differences among individuals. This will increase system performance, reduce user problems, and increase user satisfaction. Because of its close ties with industry, the VDI will also serve as a conduit for translating research results into commercial products and services.

5.0 SHERIDAN ELDER RESEARCH CENTRE (SERC)

The Sheridan Elder Research Centre (SERC) was designed for research on psychosocial aspects of aging. When it is completed in the summer of 2003, it will enable CRBCS researchers to study older adults in a more naturalistic setting than that of the artificial and sterile environments characteristic of research laboratories. Among other things it will permit the creation and manipulation of an environment that could serve as a model for adult day programs. This existence of such an environment will allow us to study communication in the elderly in a naturalistic setting, and to determine how communication deficits affect lifestyle choices, and test ways to improve communication in such an environment. In this way we hope to be able to translate our research findings into practical solutions, and test these solutions in a realistic environment. In addition, SERC's contact with 120 community-based field sites will provide not only input at the research development phase, but also a vehicle for translating research results into practical solutions to real world problems. Finally, our history of cooperation with the City of Mississauga and its Board of Trade will also help in identifying community-based health problems related to communication, as well as provide a venue for translating research on these problems into practical solutions.

6.0 OPPORTUNITIES FOR TRAINING

CRBCS offers a number of opportunities for graduate and postgraduate training. Among these is a newly funded Canadian Institutes of Health Research (CIHR) training grant in Communication and Social Interaction in Healthy Aging. This is a multi-disciplinary and multi-institutional training program open to graduate students who are working toward an advanced degree in one of the following departments: Audiology and Speech-Language Pathology (Université de Montréal), Biomedical Engineering (University of Toronto), Psychology (Concordia University, McMaster University, University of Calgary, University of Toronto), and Optometry (Université de Montréal). The goal of the program is to provide an environment that will permit students to develop transdisciplinary approaches to the study of communication systems and their effects on social interaction in older adults. The specific objectives of this program are to provide trainees with (1) excellent training within one's own discipline, (2) an appreciation of the potential contributions and advantages of a transdisciplinary approach, (3) a working knowledge of the techniques used in the other disciplines, including their strengths and weaknesses, (4) experience as part of a transdisciplinary research team, and (5) training in effective research translation. Students in the program will receive generous support to attend workshops to receive training in audiology, speech-language pathology, optometry, cognitive assessment, and the development and use of assistive technologies. In addition they will spend one semester at a department or institution other than their own working on a transdisciplinary research project. All trainees will be fully supported and receive generous support to attend conferences, workshops, and seminars.

CRBCS also will have a number of openings for postdoctoral training. These positions and other opportunities for collaborative research will be posted on our website, which is currently under construction (when completed the address for this website, as well as the one for the CIHR training program may be found at the website of the CIHR Research Group on Sensory and Cognitive Aging, http://www.erin.utoronto.ca/~w3cihrsc/Cihr/index.htm). We are excited about the new research and training opportunities afforded to us because of the generous support we have received from the Canada Foundation for Innovation, the Ontario Innovation Trust Fund, the City of Mississauga, the Natural Sciences and Engineering Research Council, and CIHR. We look forward to good relations and a number of cooperative ventures with scientists, humanists, health care professionals, industrial partners, the general public, and all those who share an interest in biological communication systems.

HEARING ACCESSIBILITY, ASSISTIVE TECHNOLOGY, and ACOUSTIC DESIGN RESEARCH FACILITY

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1. INTRODUCTION

We aspire to establish the first research facility in the world dedicated to applying recent advances in computer virtual reality simulation and digital signal processing to the health issues of people with hearing problems using an ecological approach. Our plan is guided by the recent re-conceptualization of impairment, disability, and handicap by the World Health Organization (WHO, 1999), whereby a person with a health problem must be understood in terms of how they function in their physical and social context. In the new labs, our interdisciplinary team will study how listeners function in realistic activities in realistic contexts and we will build on this knowledge to derive new solutions to alleviate disabling and handicapping conditions for listeners. We will co-develop technologies, rehabilitative practices, and built-environment and social standards such that solutions are achieved in context. Highly qualified personnel will be trained in research related to careers in education, engineering, and health.

2. OBJECTIVE

The overall objective of our basic research is to understand how the interactions between listeners and the listening environment affect the performance of complex behaviours such as communication. The associated applied research concerns how behavioral, technological, and environmental solutions can be combined to optimize performance and improve health and quality of life. Performance and design standards will be developed for communications technology and the built environment. Rehabilitative, educational, and health promotion programs will be planned and evaluated.

3. INTERDISCIPLINARY TEAM

The proposed research can only be accomplished if a new interdisciplinary team of researchers can work together in a new infrastructure. We will need biological, behavioural and social scientists (Cynader, Gick, Jamieson, Kingstone, Pichora-Fuller, Schneider) to model the complexities of listeners and their relationships to their social and physical environment; we will need engineers and computer scientists (Fels, Hodgson, Laszlo, Pai) to prepare complex stimuli using digital signal processing methods and design new technologies and environments; and we will need applied scientists in health and education (Jamieson, Pichora-Fuller) to develop interventions that will enhance auditory performance and quality of life. The proposed infrastructure will draw on and integrate the disciplinary expertise of each core investigator. Links to other CFI-funded initiatives will enable useful exchanges between a broader network of basic and applied researchers.

The ten core investigators are based in six faculties (Applied Science, Arts, Education, Graduate Studies, Medicine, Science) and two universities. They are Kathy Pichora-Fuller (Project Leader, Audiologist and Psychologist, Institute for Hearing Accessibility Research[1]), Murray Hodgson (Physicist, School of Occupational & Environmental Hygiene; Dept of Mechanical Engineering), Janet Jamieson (Psychologist, Dept of Special Education and Educational & Counselling Psychology), Alan Kingstone (Psychologist, Dept of Psychology), Charles Laszlo (Biomedical Engineer, Dept of Electrical & Computer Engineering), and Dinesh Pai (Computer Scientist, Dept of Computer Sciences); liaison to other CFI-funded projects are Max Cynader (Neuroscientist, Brain Research Centre), Sidney Fels, (Electrical Engineer, Institute for Computer Information and Cognitive Systems), Bryan Gick (Linguist, Interdisciplinary Speech Research Lab), and Bruce Schneider (Psychologist, Centre for Research on Biological Communication Systems, UTM).

4. LABS

Six laboratories constitute the infrastructure.

4.1 Auditory Virtual Reality Lab

AVR simulations will involve computing three components to construct realistic auditory scenes: 1. filter properties for the listener's two ears (i.e., customized head-related transfer functions - HRTFs); 2. properties of the space (e.g. size of room and absorptive properties of the walls, ceiling, and floor); 3. acoustical signals associated with various sound sources (e.g. speech, music, background sounds such as air-conditioning noise, chairs scraping on a floor). When these components are computationally combined, and the resulting left- and right-ear soundfiles are played out over earphones, the listener sitting in our lab will experience sound as if listening to real sources in a real room. Thus, we will create the experience of listening to a lecture in a classroom, an announcement in the subway, or multiple talkers at a cocktail party, etc. We will design such stimuli for use in perceptual and cognitive experiments and new engineering and clinical evaluation procedures, with our outcomes guiding the development of further AVR research, technology design, and behavioral interventions. The lab extends and applies more basic technologies to be developed at ICICS.

4.2 Anechoic Environment Lab

An anechoic chamber and control roomwill be used to measure sound without the contamination of reverberation. We will need this pure environment to obtain accurate measures of customized HRTFs, properties of auditory objects (e.g., contact sounds), performance characteristics of assistive listening technology, and performance of listeners. The lab provides an important control condition for both the virtual and physical simulation conditions.

4.3 Variable Acoustical Space Lab

Typical situations will be physically simulated to study of the effect of environmental factors (e.g., acoustics, lighting) on task-related behaviours, validate virtual simulations of real environments, and assess changes in behaviour resulting from specific environmental design modifications. A large space is required so that variable walls, floors, and ceilings can be added to alter room dimensions and the absorptive properties of the surfaces.

4.4 Perception and Cognition Lab

Two sound-attenuating booths for soundfield testing and control rooms are needed. Experiments will be conducted using computers to present multi-modal stimuli and record and score listener responses. Two additional work stations are needed for stimulus and experimental protocol preparation and programming, data analysis, manuscript preparation, on-line library access, etc.. Selected behavioural experiments may be adapted for brain imaging experiments to be conducted in facilities at the BRC.

4.5 Recording Studio

A sound-treated room with video- and audio-recording equipment and variable lighting will be used to obtain naturalistic samples of speech as either data or stimuli, with editing, sound analysis, and transcription facilities for subsequent manipulation of the raw recordings. Sophisticated analyses of these samples can be conducted in facilities at the ISRL.

4.6 Design Lab

This lab will be essential for us to design, construct, and test custom-built electronic and mechanical experimental equipment (e.g. response boxes, robotic interfaces) and prototype assistive technologies and listening devices. Electronics and computer equipment is required, including circuit modelling, design and layout software, tools for construction of circuits and devices.

5. LOCATION

The proposed facility will be located in the Rotary Hearing Centre (RHC) to be built as a self-contained wing of a new ambulatory care building planned for Vancouver Hospital in the heart of the city. The RHC will bring together a unique combination of resources and expertise that are now geographically dispersed: city-wide hospital and community hearing services (audiology and otolaryngology annual caseload of 25,000), UBC research and teaching, public education and consumer advocacy facilities. This city-centre location will greatly expand the number of research participants we can recruit from the city and provincial population. Co-location of our team with others engaged in clinical health research, teaching, and service delivery confers huge advantages in terms of networking with academic and professional colleagues and engaging the public, and especially hard-of-hearing consumers, in community-based research.

REFERENCES

WHO (World Health Organization). International Classification of Impairments, Disabilities, and Handicaps (ICIDH-2), Geneva : WHO, 1999.

FOOTNOTE

1. Effective September 1, 2002, Kathy Pichora-Fuller began an appointment in the Dept of Psychology at UTM; she retains an adjunct appointment at UBC.

INSTITUTE FOR INTERDISCIPLINARY RESEARCH IN CULTURE, MULTIMEDIA, TECHNOLOGY, AND COGNITION.

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1. INTRODUCTION

The term multimedia refers to simultaneously presented audio and visual media types, such as visual images and text, and auditory speech, music, and sound effects. People are adept at encoding such complex audiovisual materials as occur naturally in conversation or dramatic performance for example. The challenge arises from electronic multimedia because audio and visual information can be *arbitrarily* combined in ways that may not be optimal for the human receiver. The design of electronic multimedia is for the most part "one-size-fits-all". It overlooks individual differences, and the way in which culture impacts on learning. The technology is typically developed with the implicit requirement of "user-adapt" rather than the other way around: technology "adapt-to-user". The Institute for Interdisciplinary Research in Culture, Multimedia, Technology, and Cognition (CMTC) aims to develop theoretically-based guidelines for application of multimedia technology in education within a cultural context.

The proposed research to be conducted by the CMTC Institute will examine how culture impacts on the use of multimedia technology, particularly in education where adoption of multimedia technology is widespread. This research program which focuses on the interaction between culture and multimedia within an educational context can be well addressed through the test bed provided by the multicultural populations of New Brunswick and Prince Edward Island. Approximately one-third of the population of New Brunswick is Francophone, and is served by the completely Francophone Université de Moncton. The three universities (University of Prince Edward Island, Université of Moncton, and University of New Brunswick) are uniquely poised to study these complex issues of multimedia through cross-cultural research within the new CMTC Institute. Focusing on language, music, thinking, and social communication, the Institute will work toward a new model of the learner within the context of culture and electronic multimedia. This model can guide educators, technology designers, communities, families, and individuals who need to make decisions involving learning and technology.

Researchers from the fields of Education, Modern Languages, Linguistics, Music, Computer Science, Artificial Intelligence, and Psychology will cooperate in the study of audiovisual representations of various teaching materials, including cultural archives, as well as teaching interactions. Different disciplinary perspectives of the same audiovisual information imply the overwhelming richness of audiovisual data and highlight the hypothesis that culture may unknowingly predispose a learner to attend only to a particular aspect of audiovisual information available.

In the test of this hypothesis, four inter-related objectives guide the CMTC-Institute researchers

(1) to define mental processes underlying multimedia perception and cognition including the role of cultural background

(2) to create electronic multimedia courseware that exploits culturally relevant archival resources

(3) to evaluate the effectiveness of this electronic multimedia in educational and laboratory settings

(4) to create and test the effectiveness of artificial intelligence software that generates culturally-adaptive multimedia te teac teaching tools.

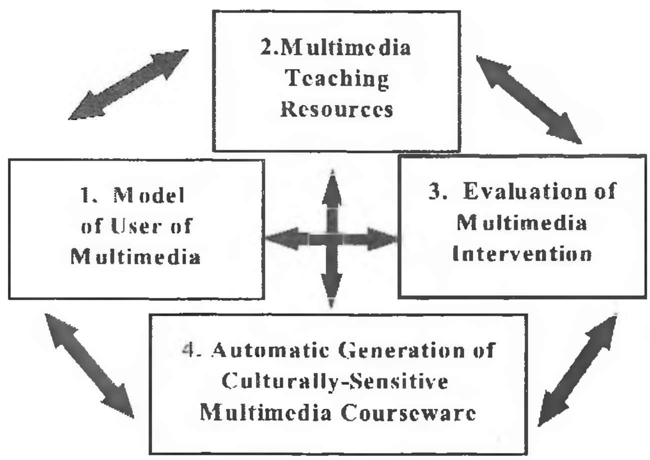

Fig. 1. Goals of the CMTC-Institute

2. THE INFRASTRUCTURE

The accomplishment of this work requires
(a) laboratories dedicated to behaviourial/cognitive/linguistic response measurement, multimedia development and usability
(b) large-scale digital storage/retrieval
(c) interactive multimedia classrooms and
(d) high-speed communications and teleconferencing.

These facilities will enable
1. creation and manipulation of audiovisual materials, and development of culturally relevant archival and teaching resources
2. storing AV material, text, archives and associated metadata.
3. Human response monitoring
(a) brain-wave recording (b) eye-movements (c) heart-rate and other physiological effects of multimedia, and correlation of such measures with other responses such as interest, memory, and cultural relevance
4. psychoacoustics and visual psychophysics and their interaction
5. cross-cultural research on use of multimedia software in live and distance-education teaching settings, including the recording and analysis of good teaching and learning environments
6. artificial tutoring programs
7. storage of discipline-specific metadata of audiovisual records
8. teleconferencing hardware/software, for data collection in classroom settings, research on interactive synchronous distance learning,and sharing information across CMTC nodes.

The multimedia, networking, and teleconferencing infrastructure made available through the Canada Foundation for Innovation will support a unique world-class Institute aiming to
(a) define cultural variables that filter multimedia perception/cognition
(b) develop and test a universal model of the learner
(c) develop unique cultural archival data bases (e.g., Oral history and music of Francophone Acadian, Prince Edward Island, Aboriginal region)
(d) link cultural databases to the design of culturally sensitive courseware. This new knowledge will give Canada a competitive edge in international markets for distance education, and will expedite lifelong learning across disciplines, intelligence levels, and cultures. Space does not permit develop of all facets of this project. Only the major theme is highlighted.

3. ACOUSTICS AND CMTC INSTITUTE

Because multimedia involves audiovisual integration, there is clearly a role for acoustics in this work. Although much research has been conducted on the perception of individual media types (e.g., text, visual images, speech, and music), less is known about the perceptual cognitive integration of the different media. In particular how can a balance be achieved between increased interest and information overload created when multiple types of media are electronically presented.

With respect to acoustics specifically, while much research has been conducted on the elements of sound, less attention has been directed to meaningful sounds of music and sound effects. In addition, while much research has been conducted on psycholinguistics and speech acoustics, there are many remaining unknowns that are relevant to theoretical and practical issues of best use of acoustically presented speech in multimedia applications in a cultural context. For example: are there universal characteristics of speech intonation that convey emotional meaning? how many audio locations is optimal for presentation of sound effects, music, and voice in an interactive learning situation? under what circumstances does music distract from a lesson or training presented via computer? does culturally significant music facilitate learning? does automatic speech recognition/translation assist the learning process of learners whose native language is other than that of the lecturer? can a second language be better taught through a "talking head" as opposed to acoustic information only? These and other questions entailing knowledge of acoustics will be addressed within the scope of the CMTC Institute. Thus, the Institute will exploit knowledge of acoustics and at the same time contribute to it while at the same time providing opportunities for research training and the acquisition of sophisticated technological skills.

Wladyslav Cichocki, Department of French, is the Chief Liaison for UNB to the CMTC Institute and Friedemann Sallis, Department of Music, is the Chief Liaison for U de M. A primary focus of UNB is French as a second language, of UNB artificial tutoring, and of UPEI English as a second language. Appreciation is expressed to the Canada Foundation for Innovation and industrial and government partners, for the infrastructure enabling the collaboration of three campuses and their researchers across many disciplines in order to model the mind of the user of multimedia, in a cultural context.

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Prospects For A Test Method For Rating Floor Toppings On Joist Floors

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1. INTRODUCTION

Recently ASTM issued a new test method, E2179[1], for rating floor coverings or toppings. A standard tapping machine is operated on a bare concrete slab and on a topping to be evaluated. The reduction in impact noise in the room below is used to calculate the change in impact insulation class (IIC) relative to an imaginary reference concrete slab. The foundation on which this test method rests is that the improvement is independent of the concrete slab thickness. E2179 is almost identical to ISO 140-8[2].

Users are warned in E2179 not to use the improvement spectrum to estimate improvements that the topping might produce on a joist floor with a wood subfloor. The improvements are not the same as those for a concrete slab[3].

This paper describes some of the current efforts to develop a similar test procedure for toppings on joist floors and some of the problems that need to be resolved.

2. TOPPINGS ON DIFFERENT JOIST FLOORS

For impact noise improvements from a test method for joist floors to be useful, they need to be applicable to any joist floor. To find whether this is true in practice, eight resilient materials were placed in turn below a 1.2 m square piece of OSB on six different joist floor systems and the reductions in impact sound pressure level were measured[4]. The basic floor system incorporated 200 mm deep steel joists, resilient metal channels, glass fibre batts, a plywood subfloor, and a 13 mm gypsum board ceiling. Either one or two layers were used in the subfloor and the ceiling and in one case, no ceiling was installed. The latter case was included with the hope that this simple floor might be acceptable as a standard floor in a new test method. Figure 1 is typical of the results obtained and shows that the improvement obtained depends quite strongly on the structure of the floor system. The results obtained suggest that any improvement spectrum from a test method will not be applicable to all kinds of joist floors.

3. USE OF A SMALL WOOD ASSEMBLY ON A CONCRETE SLAB

A major practical obstacle for testing toppings on joist floors is that the standard joist floor must be constructed or available each time a topping system is to be evaluated. For many laboratories, this means complete construction of the standard floor each time tests are to be run — a costly procedure. To eliminate the need for construction of a complete joist floor, Jonasson[5,6] suggested that a small assembly comprising only some studs and a subfloor could be used on top of a concrete slab. Toppings would then be placed on top of this for evaluation. His experiments revealed problems that needed further work. A short test of his method in another laboratory[7] was also not encouraging.

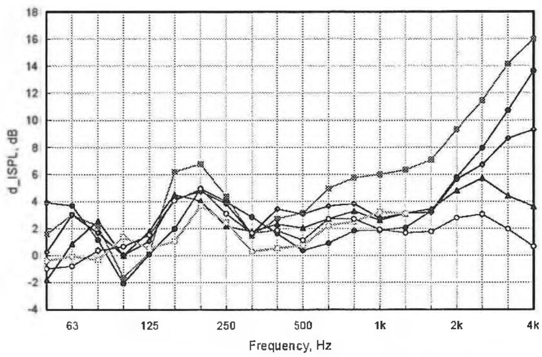

Figure 1: Reduction in impact sound pressure level (d_ISPL) for 15 mm OSB on 6 mm cork placed on six floor systems.

4. RESULTS FROM JAPAN, ISO TC43 WG22

ISO working group 22 is developing a test method for toppings on joist floors. In the current draft, any one of three standard floors can be used in a laboratory. As part of the work of the group, five different toppings (Table 1) were placed on the three standard floors and the improvements measured in a single laboratory. Analysis[4] of the test data circulated within the task group reveals some of the difficulties to be resolved before a new test method can be prepared.

For four of the five toppings, the improvements obtained on each standard floor agreed well up to about 500 Hz; above that differences of around 10 dB were common. In one case, the improvements for each standard floor agreed well at all frequencies. Figure 2 shows one set of improvements for a vinyl floor covering.

The differences in the improvement spectra above 500 Hz become unimportant when the spectra are subtracted from the levels for a bare joist floor and the improved IIC or $L_{n,w}$ ratings are calculated. Once this is done, the single number rating is determined entirely by levels at frequencies below about 250 Hz. This suggests that a test method that required measurements in frequency range from 50 to 500 Hz could give reproducible and useful results. This might be true when the topping surface is not too hard and a resilient layer is present. Toppings that include ceramic tiles on the upper surface and that create more high frequency noise may lead to quite different conclusions; this needs to be investigated.

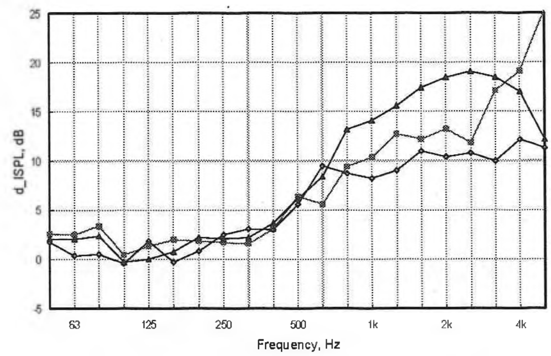

Figure 2: Reduction in impact sound pressure level (d_ISPL) for vinyl floor covering on three proposed standard floors.

5. ISO AND ASTM RATINGS

Differences between the ISO 717-2[7] and the ASTM E989[8] rating procedures also need to be addressed. E989 limits adverse deviations to 8 dB; the ISO $L_{n,w}$ procedure does not. ISO 717-2 also suggests that a better rating, at least for toppings on concrete floors, is the energy sum of the impact levels over the frequency range of interest minus 15 dB. The frequency range to be used is defined to be the same as for IIC or $L_{n,w}$, but extending the range down to 50 Hz is suggested.

Table 1 shows single number ratings calculated for the five toppings tested on the three proposed standard floors, denoted J, C, and G. In addition to IIC and $L_{n,w}$, the table shows reductions in the unweighted energy sums for the two frequency ranges indicated. Examination of this table reveals several problems.

Considering the first three toppings and any of the ratings one sees that range for tests on the three standard floors can be as much as 2 dB. Also, these toppings are not ranked consistently when tested on the three different floors. The uncertainty for measurements in different laboratories is almost certainly higher. Limiting the test method to using only a single standard floor would be preferable but not useful in all countries since typical constructions vary from place to place.

The conflict among the rating systems in the last two rows of Table 1 for the toppings incorporating the floating floor raises serious doubts about the value of current rating systems and how they relate to subjective reactions. The floating wood floor with the rubber/cork mat on top has ISO ratings that range from -1 to +17 and none are close to the IIC rating.

6. SUMMARY

Preparing a new test method for joist floors will be a difficult task. Some of the difficulties might be reduced by specifying only a single standard floor for Canadian or US use. Many of the thin toppings sold today would get improvement ratings close to zero, which is perhaps worth knowing about since it seems to represent reality for these toppings on joist floors. It is known that low frequency noise is a common problem with joist floors and the IIC rating does not address low frequencies directly. So, an essential component of a new test procedure is a new rating that deals with low frequency noise transmission.

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2 ISO 140-8 Acoustics - Laboratory measurements of the reduction of transmitted impact noise by floor coverings on a heavyweight floor.

3 Warnock, A.C.C. *Impact Sound Measurements on Floors Covered with Small Patches of Resilient Materials or Floating Assemblies*, IRC-IR-802, March, 2001

4 See papers http://www.nrc.ca/~warnock/ASTME33/e3303.html

5 Jonasson, H. *A simplified method to determine impact sound improvement on lightweight floors*, INCE 2001.

6 Kartous, N.M.S. and Jonasson, H. *Impact noise improvement in lightweight floors*. ICA 2001.

7 ISO 717-2 Rating of sound insulation in buildings and of building elements — Part 2: Impact sound insulation.

8 ASTM E989 *Standard Classification for Determination of Impact Insulation Class (IIC).*

Table 1: Improvement ratings for five toppings tested on three different floor systems (J, C, G) in a single laboratory.

	$\Delta L_{n,w}$			ΔIIC			ΔL_{sum} (50-3150 Hz)			ΔL_{sum} (100-3150 Hz)		
Topping	J	C	G	J	C	G	J	C	G	J	C	G
Vinyl	3	2	2	2	2	1	2	1	2	2	1	1
Rubber/cork	1	2	1	-1	-3	-3	-2	-4	-2	-1	-1	-2
Rubber/cork on vinyl	2	3	4	0	-1	-1	-2	-3	0	0	0	0
Floating wood	12	12	12	2	3	3	5	2	1	6	7	6
Rubber/cork on Floating wood	17	15	14	6	6	5	3	-1	1	10	10	8

Flanking Transmission In Wood Framed Multifamily Dwellings

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1. INTRODUCTION

A three-year project at the National Research Council Canada to study flanking transmission in wood-framed construction under controlled conditions was recently completed [1,2]. The focus was horizontal and vertical flanking involving the wall/floor junction in multifamily buildings built to resist wind or seismic loads. This paper reports on the effect of joist orientation (relative to the wall/floor junction), junction blocking details, joist type (solid lumber vs. wood-I joists), and wall framing (double stud, single stud or single stud shear walls) for airborne excitation. The wall and floor specimens divide the test facility into four rooms (labeled A, B, C, and D in the figures). Flanking paths involving room surfaces other than those of the test specimens are negligible.

2. RESULTS

The influence of joist orientation was tested both with wood-I joists and with solid lumber joists, but effects of joist continuity and junction details complicate the comparison. The OSB subfloor was continuous under the AB partition wall in both cases. With joists perpendicular to the partition wall, the wood-I joists were also continuous under the wall.

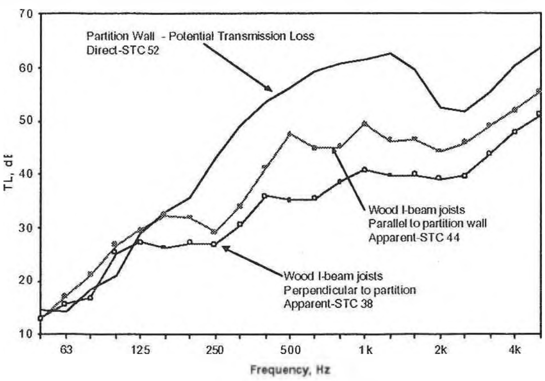

Figure 1: Apparent TL between rooms A and B for the two wood-I joist orientations shown in Figure 2. The direct transmission loss through the wall construction is also shown for comparison.

It is clear from Figure 1 that the apparent TL between rooms A and B is well below the direct TL of the wall. A series of measurements with different surfaces shielded showed that the floor-floor path between rooms A and B limits sound transmission. Clearly, flanking transmission is strongest with joists perpendicular to (and continuous under) the party wall. In both cases, improving the party wall would not appreciably affect the apparent transmission loss.

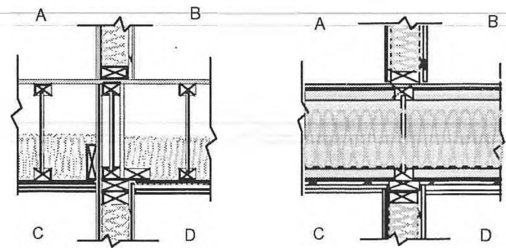

Figure 2: Floor-wall junction details with wood-I joists.

Figure 3 shows that the solid wood joists gave similar apparent TL results, for both joist orientations. Construction details are shown in Figure 4. Only the OSB was continuous across the junction, and this junction was more complex.

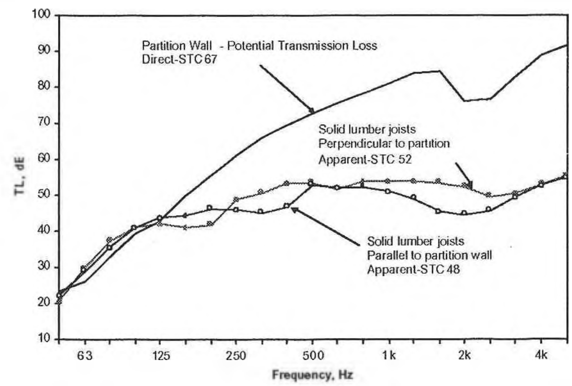

Figure 3: Apparent TL between rooms A and B, for the solid wood joist constructions shown in Figure 4.

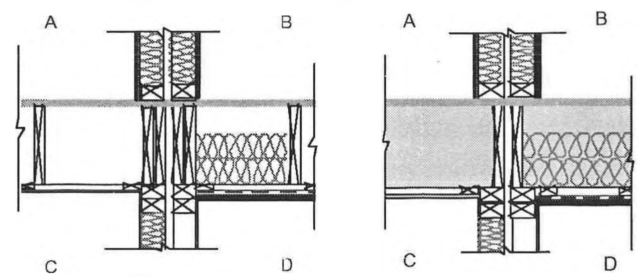

Figure 4: Floor-wall junction details with solid lumber joists.

Clearly any attempt to improve the sound isolation between rooms A and B must focus on the paths involving the floor.

This can be done either by reducing the energy getting into the floor structure, or by increasing the attenuation at the floor/wall junction. Figure 5 shows five junctions tested with the same floor (wood-I joists parallel to the wall) to assess the influence of the floor/wall junction on the flanking paths.

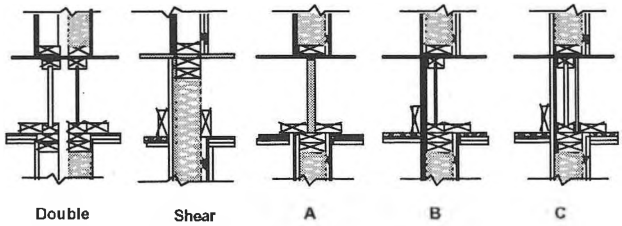

Figure 5: Sketches showing the floor-wall junction details for five variants of the double and single stud walls.

Figure 6 shows the apparent TL measured between rooms A and B, together with direct TL for these walls. In all cases the transmission is dominated by paths involving the floor, but with the more complex joint (double wall) flanking was suppressed noticeably. Single stud walls A, B, and C had essentially identical apparent TL; only case C is shown.

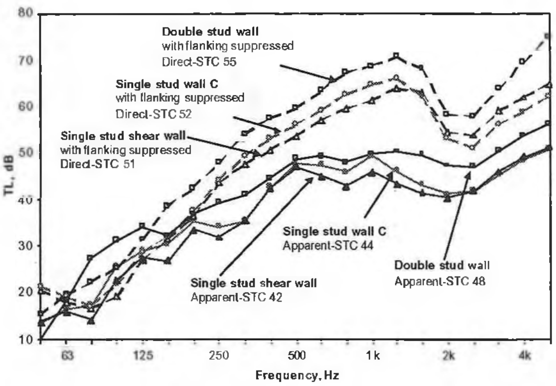

Figure 6: Apparent TL between rooms A and B for the double stud wall, single stud shear wall, and single stud wall C. Also shown is the TL expected for each wall with flanking paths suppressed.

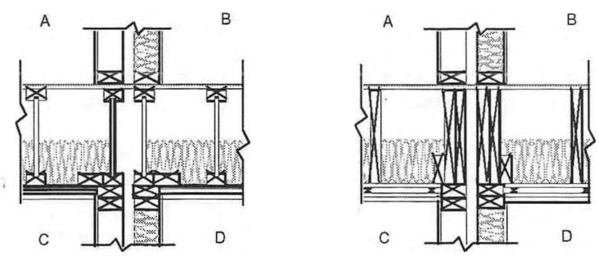

Figure 7: Sketches showing the floor-wall junctions details for comparison of solid lumber joists and wood-I joists.

The wood-I joists commonly used in current construction are lighter than traditional solid lumber so it was of interest to consider what effect this would have on the performance of the floor/wall junction. The construction details are shown in Figure 7. The apparent TL between rooms A and B, shown in Figure 8, is similar despite changing from wood-I to solid lumber joists. In both cases the TL is dominated by flanking paths involving the floor, in particular the floor-floor path. When viewing the changes in Figure 8 it is important to realize that changing the joist type affects three components in the transmission path - power incident on the junction, junction transmission, and radiation in the receiver room - which cannot be fully separated.

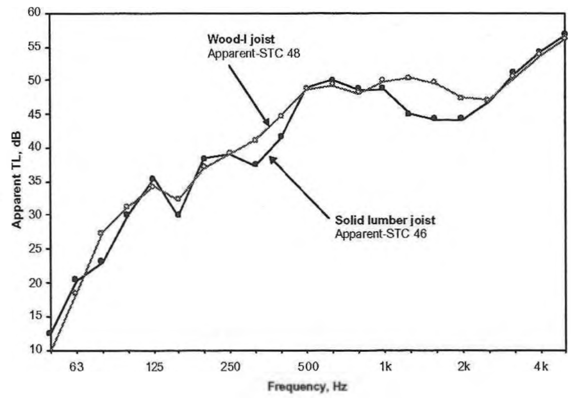

Figure 8: Apparent transmission loss between rooms A and B constructions using solid lumber and wood-I joists.

In the vertical direction between rooms A and C, the apparent TL showed little difference between the double and single stud walls. This is consistent with the direct path through the floor being the dominant path for vertical transmission. Joist type is not overly important for direct TL, especially when expressed as a single number rating. This is consistent with earlier findings [3]. Changes observable in the low and high frequencies for direct transmission do not correlate with those for flanking transmission (Figure 8). This suggests that joist type affects direct and flanking transmission differently.

3. CONCLUSION AND REFERENCES

A study of flanking transmission in wood frame construction has shown that for airborne excitation the floor/wall junction in multifamily buildings provides serious structural flanking when a continuous subfloor is used to provide resistance to wind or seismic loading.

1. R.E. Halliwell, J.D. Quirt, T.R.T. Nightingale, "Construction details affecting flanking transmission in wood framed multifamily dwellings," Proceedings INTERNOISE 2002
2. T.R.T. Nightingale, R.E. Halliwell, J.D. Quirt, "Vibration response of floors and the effectiveness of toppings to control flanking transmission", INTERNOISE 2002
3. A.C.C. Warnock, J.A Birta, Internal Report IRC-IR-811, National Research Council Canada, July 2000.

This work was supported by a consortium that included Canada Mortgage and Housing, Forintek Canada, Marriott International, Owens Corning, Trus Joist, and USG.

USING FLOOR TOPPINGS TO CONTROL FLANKING TRANSMISSION

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1. INTRODUCTION

The change in flanking sound insulation due to adding a floor topping is shown to be different for paths where energy propagates perpendicular to the joists, compared to those where energy propagates parallel to the joists. This paper summarises results presented elsewhere [1,2]. Floor vibration mappings reveal that a topping will change not only the power injected but also the propagation losses across the floor. The most effective toppings reduce input power and increase propagation losses relative to the bare floor. One type of topping exhibits significant improvement in the flanking sound insulation in one direction and a significant worsening in the other.

2. EVALUATION METHOD

The magnitude of changes to the impact sound insulation for room pairs AB and AC are compared to show that in general the improvement for direct transmission will not match that for flanking transmission involving the floor surface. The change in impact sound insulation between room pair AB is examined for each of the toppings with the joists parallel to the junction (Figure 2), and perpendicular (Figure 1). The improvement for floor flanking paths can be strongly dependent on joist orientation.

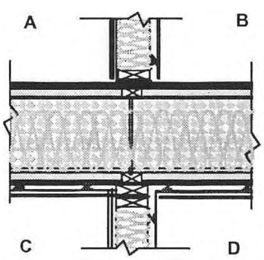

Figure 1: Sketch of the test specimen with joists perpendicular to the junction.

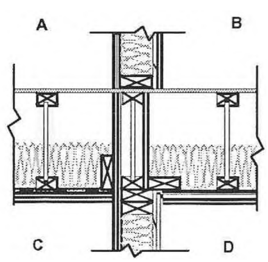

Figure 2: Sketch of the test specimen with joists parallel to the junction.

Material	Nominal Thickness, mm	Surface Density, kg/m^2	Application
OSB Overlay	18	11.7	Stapled 305 mm o.c.
Bonded-Gypsum-concrete	25	47.1	Bonded with agent

Table 1: Properties of the topping layers.

The oriented strand board (OSB) overlay consisted of adding a second layer of the subfloor material; it did not extend under the partition wall. The gypsum-concrete was applied in place, allowing bonding to all surfaces contacted including the gypsum board of walls in rooms A and B. Procedures of ASTM E1007 were followed, although measurements between rooms A and B are nominally outside the scope. Normalised impact sound pressure levels (NISPL) were measured with the ISO hammer box located at the same four positions near the center of the floor in room A. The hammers impart power only into the floor surface, which is involved in all transmission paths to rooms B and C. Hence, the change in NISPL measured in rooms B and C with and without a topping indicates how well the topping controls flanking and direct transmission of impact sound.

3. DIRECT VS FLANKING TRANSMISSION

Since the NISPL for room pair AC is controlled by direct transmission, the change in AC sound insulation can be compared to the change for AB, to assess the effectiveness of a topping to control direct and flanking transmission, respectively. Each topping was applied to the constructions of Figure 1 and Figure 2 to determine if orientation of the joists is an important factor.

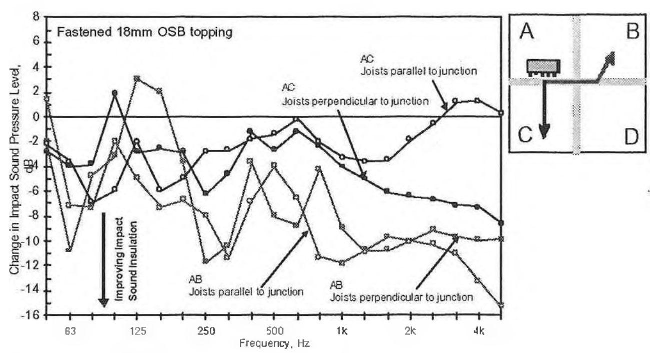

Figure 3: Change in NISPL due to 18 mm OSB overlay, as a function of the orientation of the joists shown in Figure 1 and 2.

Figure 3 shows the change in the receiver room NISPL due adding an 18 mm OSB overlay. It shows that the overlay reduced the NISPL (improved sound insulation) for both direct and flanking transmission. For frequencies above 200 Hz, regardless of the orientation of the joists, there is a greater reduction in NISPL for room pair AB than for AC. The overlay controls direct transmission less effectively than flanking transmission involving the floor surface. With the OSB overlay, the improvement for flanking transmission is not very sensitive to the orientation of the joists.

Figure 4 shows that for both direct and flanking transmission, adding the gypsum-concrete topping without an interlayer will increase the NISPL relative to the bare floor in the high frequencies. The most important feature however is the relation of the curves. In the frequency range 160–2000 Hz with the joists perpendicular to the junction, Figure 4 shows the topping attenuates floor flanking paths more than direct transmission though the floor. The opposite is true when the joists are parallel to the junction, since in the frequency range 400–2000 Hz the topping is better at controlling direct transmission than flanking transmission.

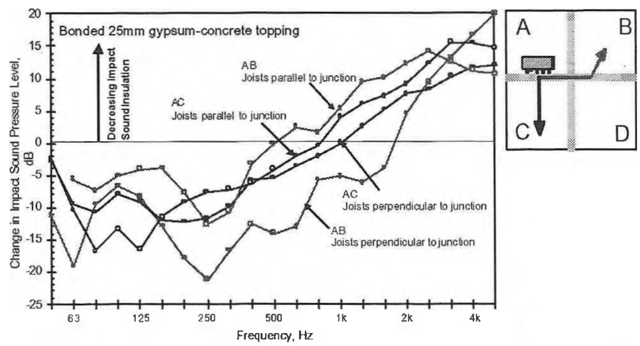

Figure 4: Change in NISPL due to applying bonded 25 mm gypsum-concrete topping as a function of joist orientation.

Changes in NISPL will differ because adding a topping changes not only power injected by the ISO hammer box, but also the propagation losses in the floor surface. Changing injected power should affect direct and flanking transmission similarly. However, for the highly-damped floors considered here, changes to propagation losses will affect flanking transmission more than direct transmission, because propagation determines the incident structural power at the flanking junction.

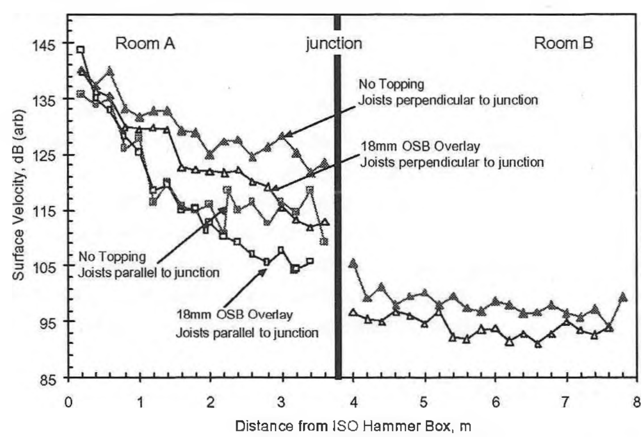

Figure 5: Surface vibration levels at 1 kHz measured along two orthogonal lines from the source with/without the OSB overlay.

Estimates of the change in propagation losses can be obtained from velocity levels measured along two orthogonal lines from the source. One line is parallel to the joist orientation, while the other is perpendicular. Results for the bare floor and the 18 mm OSB overlay, in Figure 5, indicate that for both joist orientations, the topping increased propagation losses in the exposed surface (i.e., there is a greater level difference between source and junction with the topping than without).

Figure 6 shows that adding the bonded gypsum-concrete topping applied to the same bare floor increases propagation losses when the joists are perpendicular to the junction. Here the vibration energy propagates parallel to the joists to reach the junction. The opposite is true when the joists are parallel to the junction, since to reach the junction, vibration energy must propagate perpendicular to the joists. Thus, the bonded topping is less effective when applied to flanking paths where the joists are parallel to the junction. This is consistent with the trends in the impact sound pressure levels shown in Figure 4.

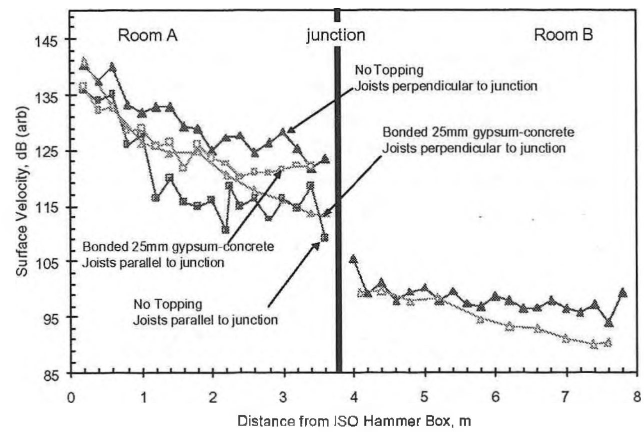

Figure 6: Surface vibration levels at 1 kHz measured along two orthogonal lines from the source with/without gypsum-concrete.

4. CONCLUSIONS AND REFERENCES

Vibration maps indicate that adding a topping may increase or decrease propagation losses across the floor relative to the bare floor. The change in propagation loss was shown to be a function of the type of topping and the orientation of the joists in the floor to which it was applied. A topping that increases propagation losses relative to the bare floor will be more effective in controlling flanking than direct transmission. The important implication is impact sound insulation improvements due to adding a topping when there is no appreciable flanking (i.e., using ASTM E492) should not be generalised to situations where flanking transmission across the floor surface is important.

1. R.E. Halliwell, J.D. Quirt, T.R.T. Nightingale, "Construction details affecting flanking transmission in wood framed multifamily dwellings," Proceedings INTERNOISE 2002
2. T.R.T. Nightingale, R.E. Halliwell, J.D. Quirt, "Vibration response of floors and the effectiveness of toppings to control flanking transmission", INTERNOISE 2002

This work was supported by a consortium that included Canada Mortgage and Housing, Forintek Canada, Marriott International, Owens Corning, Trus Joist, and USG.

DIRECTIVITY OF HUMAN TALKERS

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1. INTRODUCTION

Human talkers radiate more speech energy forward than to the side or to the rear. This directivity can be utilized when planning open office layouts to increase speech privacy. While talker directivity had been studied[1,2,3] not enough information had been published to allow detailed computer calculations. So, as part of a project studying new kinds of open-offices, the sound fields around 20 male and 20 female talkers were more completely examined.

2. DATA COLLECTION

The sound field surrounding the talkers was surveyed using two arrays of microphones arranged on two fixed arcs on orthogonal meridian planes of a sphere of 1-meter radius centered at the talker's mouth in the anechoic chamber at IRC. Figure 1 shows a schematic drawing of the arrangement of the 16 microphones used in the measurement.

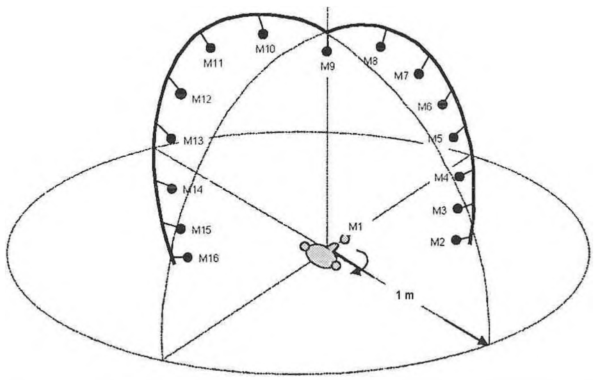

Figure 1: Schematic drawing of the microphone arrangement used for the survey of the sound field surrounding human talkers.

The subject sat on a rotatable chair. The height was adjusted so the subject's mouth was located at the center of the spherical surface of the microphone arrays with the subject looking straight ahead. For a complete directivity measurement, the subject was rotated through 90° at 15° intervals. Since there were two microphone arcs, this covered an azimuthal range of 180° on one side; symmetry about the head was assumed. Microphone M9, directly above the talker's head, was used as the reference.

The talker was asked to remain seated with the same posture and to talk normally at each orientation on some subject for 40 seconds without being concerned about exact duplication of the words. Signals from the microphones were recorded simultaneously on two synchronized DAT recorders (Tascam DA-38). The recorded signals were analyzed in 1/3-octave bands using a B&K 2144 real time analyzer programmed to collect 30-second L_{eq} values. When analysis was complete, relative levels were obtained for 141 positions distributed on a sphere around the talker's head.

As well as obtaining directivity matrices, some other factors were investigated. The main conclusions are listed here.

- The average male voice spectrum shape obtained in these measurements agreed well with that from field measurements[4]. Some differences were observed for the female voice spectra.
- No significant differences in directivity were detected between male and female talkers although average spectra differed.
- No significant differences in directivity were detected between English and French talkers. The frequency content of the two languages was similar.
- Similar directivities were obtained for both normal and loud voice levels but significant changes behind the talker were observed for low voice level.
- One-third-octave band directivities showed fairly good agreement with those published by Moreno & Pfretzschner[2] and by Dunn & Farnsworth[1].
- Similar directivities were obtained for the average human talker and the B&K Head and Torso Simulator.

The directivity patterns for the males, females and all subjects are shown in Figure 2 for the vertical and the horizontal planes. The figure shows that there is a reduction of around 7 dBA in voice level directly behind a talker. By positioning work stations to take advantage of this directionality, intrusive speech can be reduced in open-office workstations provided the furnishings are sufficiently absorptive. The detailed directivity information can be used in computer models of proposed office arrangements.

3. VOCAL EFFORT

Only one male speaker was asked to vary the loudness of speech so general conclusions can not be made. However, the plots in Figure 3 show that in this case, the speech energy directed behind the subject was about 4 dBA less than when a normal or raised voice effort was used. Thus inducing office occupants to lower their voices may have even more benefits when occupants face away from each other.

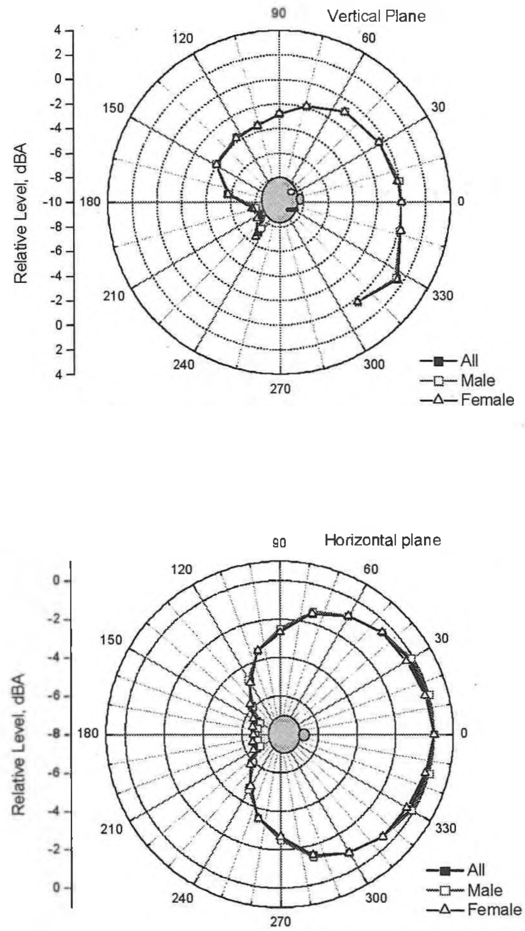

Figure 2: Comparison of the relative A-weighted levels in the frontal vertical and horizontal planes of the male and female talkers.

4. SUMMARY

The details of the work are available in an IRC internal report[5].

ACKNOWLEDGEMENTS

This work was supported by Public Works and Government Services Canada.

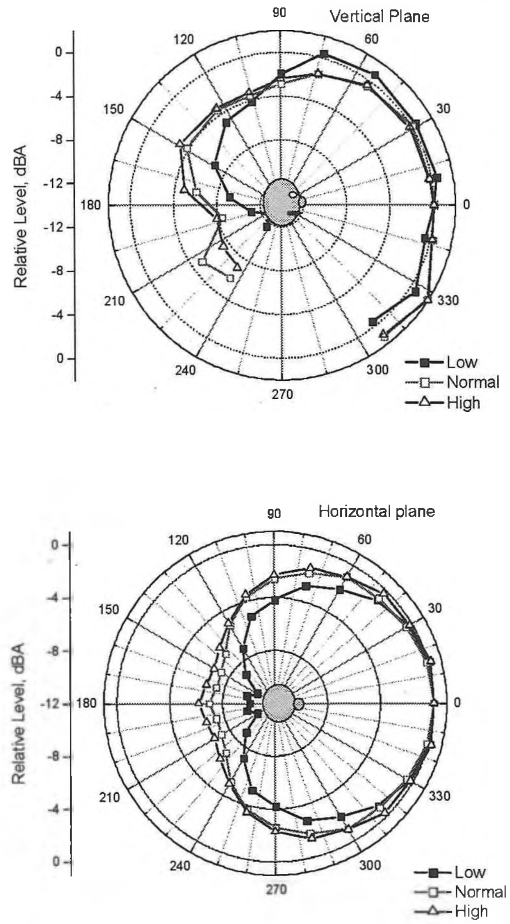

Figure 3: Directivity patterns of the relative A-weighted levels of a male talker speaking at three different voice levels.

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[1] Dunn, H.K. and Farnsworth, D.W. "*Exploration of Pressure Field Around the Human Head During Speech*", J.A.S.A. Vol. 10, pp184-199, January 1939.

[2] Moreno, A. and Pfretzschner, J. "*Human Head Directivity in Speech Emission: A New Approach*", Acoustics Letters Vol. 1, pp78-84, 1978.

[3] McKendree, F.S. "*Directivity Indices of Human Talkers in English Speech*", Proceedings Inter-Noise 86, pp911-916, July, 1986.

[4] Warnock, A.C.C. and Chu, W.T. "*Voice and Background Noise Levels Measured in Open Offices,*" Internal Report IR-837, January 2002.

[5] Chu, W.T and Warnock, A.C.C. "Detailed Directivity of Sound Fields around Human Talkers." IRC Research Report 104, 2002.

SPEECH LEVELS IN OPEN PLAN OFFICES

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1. INTRODUCTION

A key variable for predicting speech privacy in open offices is the loudness of speech in such spaces. Since the environment influences how loudly we speak, speech levels need to be measured in typical open office situations so they can be used to calculate speech privacy expected there more accurately. The most extensive set of data on speech levels is that in the report by Pearsons et al[1]. The mean spectrum for "normal conversational speech" from that study was incorporated into ASTM 1130[2] for estimating speech privacy in open plan offices. A more recent study[3] agreed fairly well with that of Pearsons. However, it is not obvious that "normal" voice levels are appropriate for open offices.

2. DATA COLLECTION

Measurements of voice levels used in face-to-face conversation were made in nine open offices in the Ottawa/Hull Area on behalf of Public Works and Government Services Canada (PWGSC).

Subjects were asked to wear a headset microphone and to speak as naturally and freely as they normally would in their workstation to a person sitting next to them (the interviewer). The speech was to last for 1 minute without interruption. Any topic could be chosen. To help with making a choice, four suggestions were provided: their job, their last holiday, the town they grew up in, or the route they take to work in the morning. The monologue could be in French or English. The same interviewer was used at each site. The voices recorded were recorded on a calibrated digital tape recorder and later analyzed in the laboratory using a 1/3 octave-band real time analyzer. Using a 30 second segment of the recording and an integration time of 1/16 second, a total of 480 samples were obtained for each 1/3-octave band from 160 Hz to 8 kHz. From this information, mean, L_1, L_{10}, L_{50}, L_{90}, and L_{eq} were calculated in each band. A calibration procedure was used to convert to levels 0.9 m in front of the talker.

3. RESULTS

The distribution of average A-weighted sound pressure level for each subject measured in the nine office sites is shown as a histogram in the Figure 1. There is a peak at 51dBA for the male group, whereas the dominant levels for the female group spread from 47 to 53 dBA. For all subjects, the average male and female voice levels were 51.3 dBA and 50.6 dBA respectively.

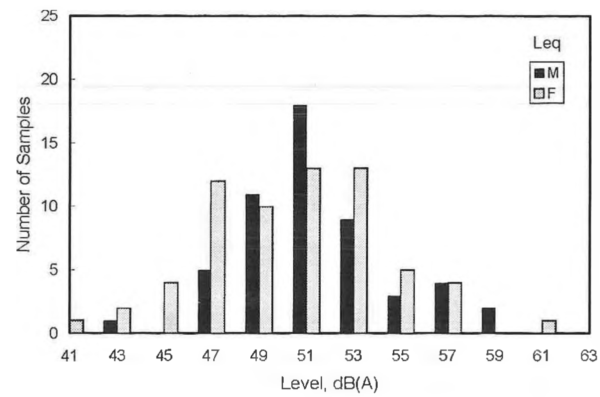

Figure 1: Distribution of A-weighted speech levels, dBA, for the 118 subjects in the nine open plan offices.

Figure 2 compares the mean long-term average 1/3 octave-band speech levels from all 118 subjects with the spectra given by the ANSI[4] and ASTM standards. Results from the current study are significantly lower than those given by the standards. A probable explanation is that the current study used conversational speech with two people sitting close together whereas the ASTM standard and possibly the ANSI standard used levels based on subjects reciting fixed texts and being asked to speak in a "normal" voice. Also shown in Figure 2 are Pearsons' results for "casual conversation". They agree well with the current results.

The A-weighted levels for each of the curves in Figure 2 are given in the following table.

This study	50 dBA
ANSI S3.5	59 dBA
ASTM E1130	57 dBA
Pearsons Casual	50 dBA

These differences in voice level lead to quite large differences in Speech Intelligibility Index and therefore speech privacy.

In the same offices measurements of propagation were made between workstations. Assuming a background noise level of 45 dBA and the ASTM E1130 voice spectrum, values of

Speech Intelligibility Index (SII) and Articulation Index (AI) were calculated for each workstation pair. The relationship found between the two ratings was

$$SII = 1.03\ AI + 0.06,\ \ r^2 = 0.997.$$

Calculations in E1130 are made using fewer frequency bands than in ANSI S3.5. Since the correlation between SII and AI was found to be so high, ASTM E33 decided not to change E1130.

AI=0.15 (SII=0.2) is commonly taken as the upper limit for normal privacy in open offices. With an assumed 45 dBA background spectrum and the voice levels in E1130, only about 10% of the occupants would experience AI $\leq$ 0.15. If instead the measured average voice level is used, about 50% of the occupants would experience AI $\leq$ 0.15.

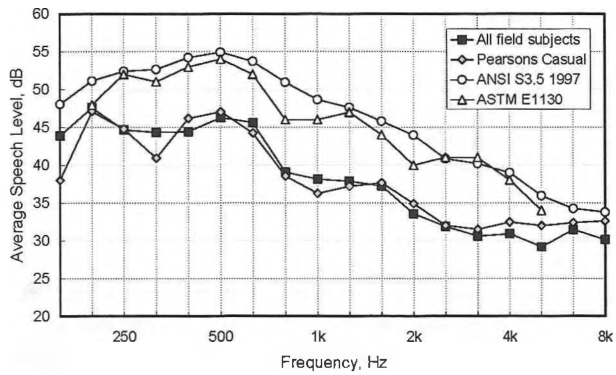

Figure 2: Comparison of the mean average 1/3 octave-band speech levels at 0.9 m in front of the talker.

ASTM E1130 and ANSI S3.5 give the difference between the peak and the L_{eq} speech levels in each 1/3-octave band as 12 dB. Results from the current study are shown in Figure 3. With the exception of a few low-frequency bands, all the differences are significantly greater than 12 dB although the A-weighted differences are close to 12.

4. COMMENTS

The speech levels were measured when the subjects were speaking to an interviewer sitting less than 1 metre away. They are not appropriate for conversations with co-workers at distances of 3 or 4 metres. It is thought that average voice levels used during telephone conversations are not likely to be very different from those measured in this project but further measurements would be needed to verify this assumption.
Observations in the offices during the recordings supported common experience that the voice level used depends strongly on factors such as the distance between the talker and the listener and the subject of the conversation. Office etiquette can also be a factor. If there is continual social pressure from co-workers to speak quietly, just as in a library setting, then one can expect less annoyance from intrusive speech. A study of behaviour in open offices might lead to procedures to control intrusive speech that are at least as effective as extensive use of barriers and sound absorbing materials.

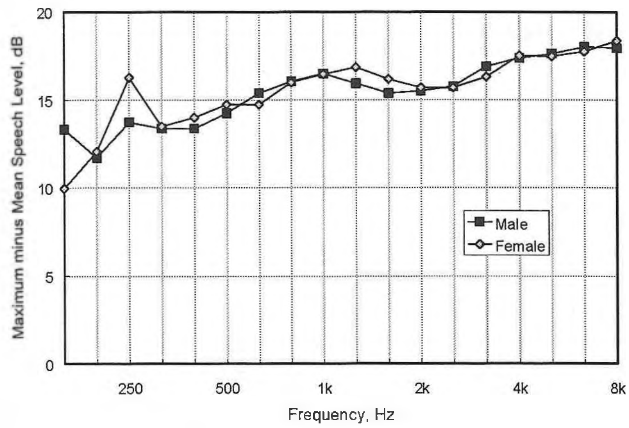

Figure 3: Average difference between the peak levels and the L_{eq} levels for the male and female groups. The difference between the average A-weighted L_{eq} and the average A-weighted peak values was 12.3 dBA for males and 12.2 dBA for females.

The details of the work are available in an IRC internal report[5].

ACKNOWLEDGEMENTS

This work was supported by Public Works and Government Services Canada.

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[2] ASTM E1130 *Standard test method for objective measurement of speech privacy in open offices using articulation index.*

[3] McKendree, F.S., "*Directivity indices of human talkers in English speech*," Proceedings of Inter-Noise 86, pp. 911-916, July 21-23, 1986.

[4] ANSI S3.5 *American National Standard Methods for the Calculation of the Speech Intelligibility Index.*

[5] Warnock, A.C.C. and Chu, W.T. "*Voice and Background Noise Levels Measured in Open Offices*," IRC Internal Report IR-837, January 2002.

Quantifying Public Address System Performance in a Public Transit Environment

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ABSTRACT

The Toronto Transit Commission (TTC) new Sheppard subway line is nearly complete. It adds 5 new stations and 6.4 km of tunnel to their system. From the beginning of the overall design of the stations, efforts were made to ensure the public would be able to understand public address(PA)announcements during revenue service and in emergency situations. The PA system design goal was a speech transmission index (STI) of 0.45 or 'Fair' intelligibility. As each station is completed, it is being measured using Maximum Length Sequences in a PC based measurement system in order to validate the performance of the PA and the station architectural treatment. The paper will discuss the PA system used, summarize the measurements, provide some assessment as to the performance of acoustic treatments used in the station finishes as well as comment upon the practical aspects of using the Dirac measurement system.

1. INTRODUCTION

The Toronto Transit Commission (TTC) is completing the new Sheppard Subway line,. The PA system is considered an essential part of the emergency response and alarm systems. As part of setting up the system design standards, the project concluded that an STI of 0.45 from the PA system was appropriate.[1, 2] This paper summarises measurements of the STI in the stations.

Table.1 Generally Accepted STI Numerical Ranges and Ratings

STI Range	Rating
Less than 0.3	Unsatisfactory
0.3 – 0.45	Poor
0.45 – 0.6	Fair
0.6 – 0.75	Good
Greater than 0.75	Excellent

2. STI TESTING

Testing is done using Dirac software from the Netherlands and currently distributed by Bruel and Kjaer, running on a Dell notebook fed by a Rion NA29E Type I sound level meter. The system uses Maximum Length Sequences to determine the impulse response of the station and PA. From the impulse response, the Reverberation Time, the STI and other acoustical parameters are then calculated. The system is calibrated to adjust for any limitations in the sound card and has a 60 dB Total Harmonic Distortion plus Noise ratio. STI is implemented in accordance with IEC 60268-16.[3, 4]

Two types of measurement were made. In the first, a high quality loudspeaker was used to measure the acoustics of the station alone. The microphone was typically placed approximately 6m from the PA speaker and STI and the Early Decay Time (EDT) were measured. Then the computer signal was fed into the station PA system and the STI and EDT of the full system and station were measured.

Three stations out of five have so far been completed to the point where they can be tested: Leslie, Don Mills and Bayview. They were substantially complete insofar as interior finishes were concerned. Platform dimensions were nominally 100 x 20 x 4m (including the rail area) and concourse dimensions 100 x 20 x 5m. The areas tested were the subway platform, the ticket concourse located above the platform area and the upper bus terminal. In each space four tests were made: with a single high quality loudspeaker approximately 6m from the microphone, to determine the station performance independent of the PA system, with the microphone 1m from a PA speaker to determine the PA system performance independent of the station acoustics and at locations 1.2m above the floor either under a PA speaker or halfway between sets of four PA speakers to determine typical performance at a patron's ear.

3. RESULTS OF TESTING

Table 2, gives the results of testing at Bayview Station. Tests were carried out through the station PA system and using the high quality loudspeaker (speaker, in italics). STI is given for male and female voices as described in IEC 60268-16[5]. Although the original work specified only a single STI, the new standard (1998) specifies both and it was decided that the most recent standard should be used.

Table.2 Bayview Station Results

Preliminary Results	Source	STI female		STI male		EDT 500
S upper concourse untreated between speakers	PA	Fair	0.45	Poor	0.42	3.28
S upper concourse untreated 1.5 m under speaker	PA	Fair	0.56	Fair	0.52	3.05
N ticket concourse between speakers some treatment	PA	Fair	*0.52*	Fair	*0.52*	1.76
N ticket concourse PSB at 8m pointing to microphone	*Speaker*	*Good*	*0.62*	*Good*	*0.61*	*1.31*
N ticket concourse PSB at 8m pointing up	*Speaker*	*Fair*	*0.57*	*Fair*	*0.56*	*1.36*
S ticket concourse treated - under speaker	PA	Good	0.63	Good	0.62	1.37
S ticket concourse treated between speakers	PA	Fair	0.57	Fair	0.55	1.8
platform between 2 speakers	PA	Fair	*0.45*	Fair	*0.45*	2.62
platform under speaker	PA	Fair	0.55	Fair	0.53	2.28

Several comments are in order:

The treated upper areas met or in many cases exceeded the requirement of STI 0.45, or Fair performance. The platforms met the criterion under the PA speakers and were marginally below between the PA speakers. Untreated areas, gave generally poor results. The PA speakers at 1m were tested at Leslie station and generally gave Good performance on the upper levels (with higher ceilings) and Fair performance at the Platform level.

4. CONCLUSIONS

It is concluded that main parameters affecting transmission of speech from the PA system, in approximate order of importance, were: reverberation of the space, dimensions, PA speaker placement, and PA system performance. In general the reverberation was adequately controlled to 1.5 seconds, but artificial reverberation from the row of PA speakers along the platform increased this. Clearly the acoustical treatment significantly improved speech intelligibility. The ceiling height was the most important dimension and interacted with the PA speaker placement to reduce speech intelligibility on the platform. This will require further attention. The PA system performed adequately, but the PA speakers, which were provided and installed separately, did not give usable output in the 125 Hz octave band, decreasing STI somewhat. They also did not have sufficient dispersion, allowing the intelligibility to decrease significantly between speakers. However it is clear that the acoustical treatment and specifying a good quality PA system have contributed significantly to providing better intelligibility of announcements for passengers.

ACKNOWLEDGEMENTS

This work was done as part of the commissioning of the TTC stations and the support of the TTC and Hatch Mott MacDonald is gratefully acknowledged. However, the conclusions are those of the authors alone.

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CLASSROOM ACOUSTICS FOR ACCEPTABLE SPEECH RECOGNITION: A REVIEW

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Introduction

One might think that the basic acoustical requirements for classrooms are well established. The fact that communication problems are frequently encountered and the current flourish of new work on this topic indicates that problems remain to be solved. As Fig. 1 illustrates, speech intelligibility scores increase with increasing speech signal-to-noise ratio (S/N) until near 100% intelligibility is reached. Thus good conditions for speech are a question of obtaining adequate S/N ratios. However, one must appreciate what contributes to 'effective' speech and noise levels.

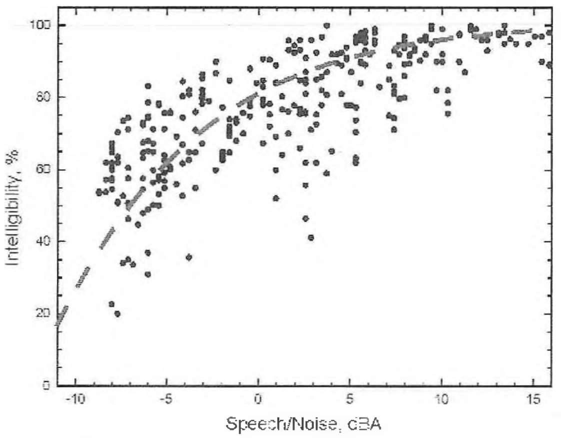

Fig. 1. Intelligibility vs. speech signal-to-noise ratio[1].

Components of Effective Speech Levels

The 'effective' speech level is that due to the combination of the direct sound and early arriving reflections of the speech sound that together contribute to increasing intelligibility. The noise is the combination of ambient noise in the room plus reverberant speech sounds that together decrease intelligibility. This effective speech/noise ratio was termed a useful/detrimental (U/D) sound ratio by Lochner and Burger [2] and intelligibility scores are well related to U/D values as seen in Fig. 2.

In spite of extensive early work, many research studies have shown no appreciation of the different effects of early and late-arriving reflections of speech sounds. Some have even questioned the importance of early reflections. New results, shown in Fig. 3 confirm that early reflection energy has approximately the same benefit as increased direct sound energy for improving speech intelligibility scores and that this holds true for listeners with hearing impairment too. Further analyses have demonstrated that early reflection energy can increase effective speech levels in rooms by as much as 9 dB. In some situations, such as when the talker turns their head or for listeners at the rear of a room, early reflection energy is critical to understanding speech.

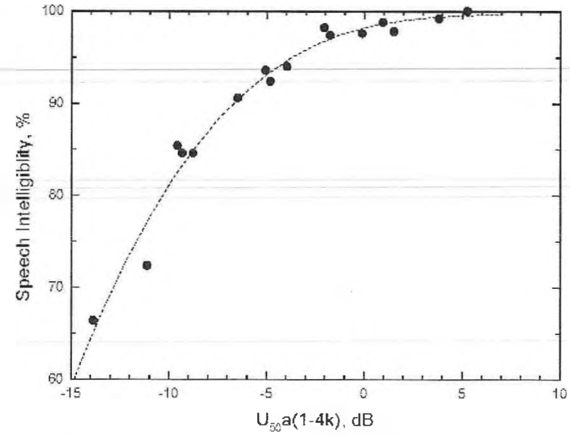

Fig. 2. Intelligibility vs. useful/detrimental ratio for 1 to 4 kHz results. [3].

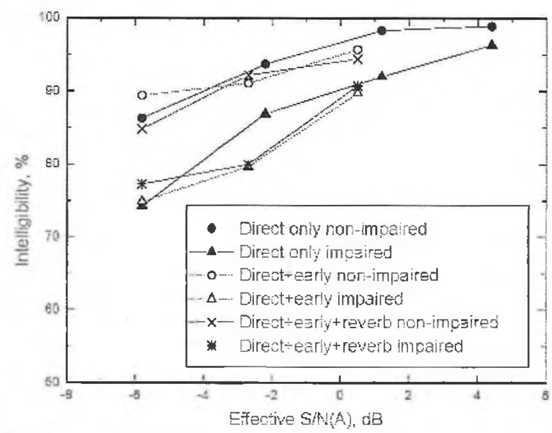

Fig. 3. Demonstration of the benefit of early reflections [4].

Measuring Speech and Noise Levels

Almost all reported measurements of noise levels in classrooms suggest that noise levels are too high for optimum speech communication. Fig. 4 summarizes measured noise levels from various studies. These vary from 40 to over 80 dBA and show a trend to be greater for classes of younger children. It is not always clear how such measurements were made and in some cases these ambient levels may also include some student activity noise. Similarly, there is some uncertainty as to the level of speech sounds produced by typical teachers. To resolve the need for more representative measurements of speech and noise levels, Hodgson et al. [6] proposed deriving them from statistical distributions of recorded speech and noise levels in classrooms. Fig. 5 shows an example of this technique for a male high school teacher using an overhead projector. The peaks due to the teacher's voice level and

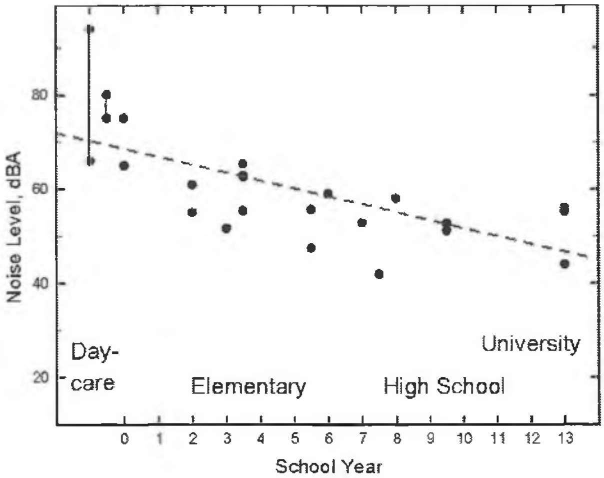

Fig 4. Reported classroom noise levels[5].

the projector are clearly identified. Preliminary measurements of this type suggest that teacher's voice levels are louder than often assumed.

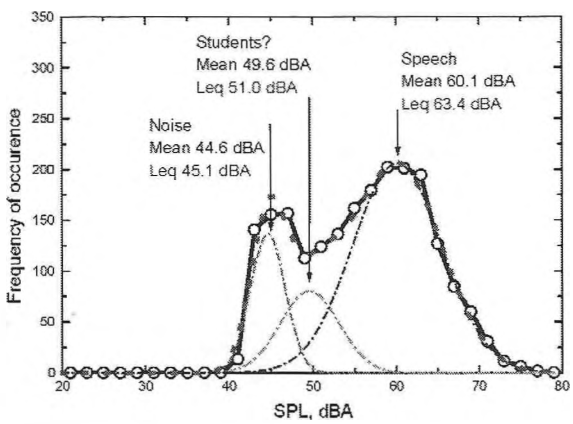

Fig. 5. Statistical distribution of speech and noise levels.

Room Acoustics Criteria for Classrooms

Near-optimum conditions for speech communication in school classrooms are often said to require noise levels to be no more than 35 dBA and an optimum reverberation time (RT) of about 0.5 s [7,8]. The derivation of such criteria are based on assumed speech source levels and typical ambient noise levels. There is also a trade-off between optimum reverberation time and the maximum acceptable ambient noise level as illustrated by the equal U/D contours in Fig. 6. These indicate that in noisier conditions a slightly higher reverberation time would be optimum. This is due to the relation between increased early reflections and RT. (Of course, too much reverberation adds unwanted later arriving speech sounds). It is also known that younger children and other special groups need better conditions than young adults to achieve the same intelligibility scores. Estimates of the required maximum noise levels as a function of the age of younger listeners can lead to very low noise level requirements [5] that seem contrary to common experience.

However, if the assumed teacher voice level is incorrect and if teachers tend to use a stronger voice level, the background noise level requirements could be 5 or 10 dBA higher. Recent measurements indicate that teacher's voice levels correspond to about 65 dBA (at 1 m) rather than the 55 dBA assumed in deriving the 35 dBA maximum noise level criteria. There is also the question of the maximum voice level that teachers should use to avoid any voice impairment that is a common problem among many teachers [5].

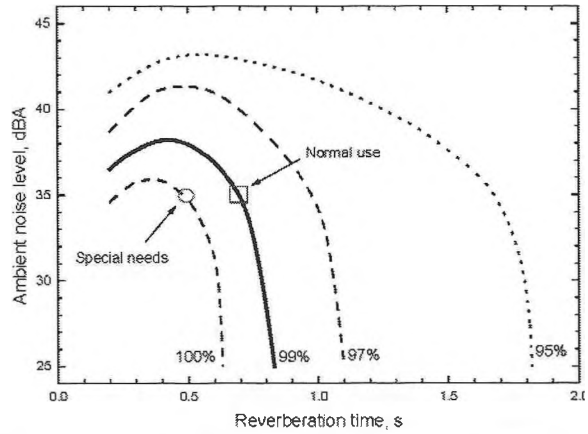

Fig. 6. Equal U/D contours by noise level and RT[7].

Conclusions

Although the basic principles seem well established, there is still considerable uncertainty in current derivations of acoustical criteria for classrooms. We need to better define safe teacher voice levels and the required speech/noise ratios for all ages of children. We also need to develop procedures for designing rooms that maximize the benefits of early-reflected speech sounds. A new project, part of the Canadian Language and Literacy Research network, will attempt to answer some of these questions.

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AN EASY TO USE MODEL TO AID IN THE ACOUSTIC DESIGN OF CLASSROOMS.

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1 Introduction

Evolving classroom codes and standards require the design professional to limit the background noise level, and target the reverberation time to a specified time. In order to do this, estimates of design parameters need to be calculated. To facilitate this process, a model based on classical acoustics has been developed which allows the designer to select component performance characteristics from menus of measured data.
These data are entered into an energy balance, which predicts the overall classroom diffuse field sound pressure level. The direct fields from sound sources inside the room are added to predict overall sound pressure level as a function of position in the classroom. Reverberation time is also calculated. In this paper, the formulation of the model is described. Validation experiments and comparisons with another analysis tool are shown, and a design example is included.

2 Formulation

Sound travels to and from a classroom space by structural and airborne paths. This model addresses airborne paths only. Care should be taken isolate structure borne sources such as roof mounted central HVAC components, but these effects are not included in this model. Sound energy enters the space by dispersion from the adjacent spaces through walls, windows, doors, ceilings, and floors. Sound energy is generated in the space by noise sources such as HVAC units and terminals, lighting fixtures, plumbing, computer equipment, occupants, and etc. (background noise does not include occupant generated noise). Sound energy is dissipated within the space by absorption of the rooms' surfaces, furnishings, and occupants. Figure 1 shows the classroom energy balance. Note that the dispersive energy can travel either way depending on the relative strength of the driving potential between the classroom and the adjacent space.

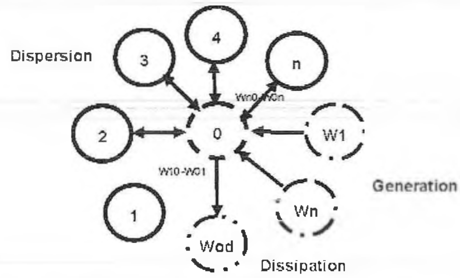

Figure 1. Classroom energy balance.

The relationship between energy and sound pressure, Eq. 1, is

$$\overline{E} = \frac{\overline{p}_{sp}^2}{\rho_0 c_0^2} V$$

where

$\overline{E}$ – Space Averaged Energy
$\overline{p}_{sp}$ = Space Averaged Sound Pressure
V = Room Volume
ρ_0 = Density of Air
c_0 = Air Speed of Sound

The relationship between power dispersed and transmission loss factor's (TL's), Eq. 2, is

$$\Delta P_{n\to 0} = \tau_{n\to 0}\frac{A_{wn}}{4\rho_0 c_0}\left(p_n^2 - p_0^2\right) = C_n\left(p_n^2 - p_0^2\right)$$

where

$$TL_{n\to 0} = 10\mathrm{Log}\left(\frac{1}{\tau_{n\to 0}}\right)$$

The relationship between power dissipated and absorption coefficients, Eq. 3, is

$$P_{od} = \frac{A}{4\rho_0 c_0}\overline{\alpha} p_0^2 = D p_0^2$$

where

$$\overline{\alpha} = \frac{\sum \alpha_i A_i}{\sum A_i}$$

Summing these terms results in the following simplified power balance equation. Eq. 4,

$$p_0^2 = \frac{\sum C_i p_i^2 + \sum W_i}{\sum C_i + D}$$

where

$$C_n = \tau_n \frac{A_{wn}}{4\rho_0 c_0}$$

Wi = Internal power

$$D = \frac{A}{4\rho_0 c_0}\bar{\alpha}$$

The direct field is added to the diffuse field to get the total local pressure. The direct field model, Eq. 5, is

General Case : n sources of W_i & Q_i

where

W_i = power, watts

$$Q_i = \text{Directivity} = \frac{4\pi r^2 c p_r^2}{\rho_0 c^2 W_i},$$

$$L_p = 10Log\left(\frac{W_r \rho_o c}{4\pi p_r^2}\sum \frac{Q_i \, {W_i}/{W_r}}{r_i^2}\right)$$

Ref ASHRAE Handbook of Fund. Chap. 6

3 Validation

The diffuse field portion of this simplified model was compared to predictions made by a commercially available statistical energy analysis program. Predictions were also compared to measurements made in a suit of reverberation rooms. The case shown in Figure 2 includes the combined effects of generation and a dispersive path .

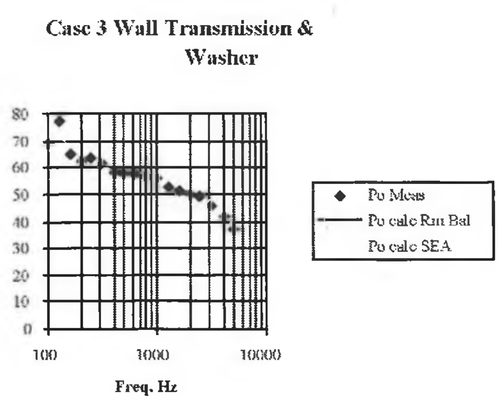

Figure 2. Combination of generation and dispersion.

Figure 3 shows the combined effects of a window and a wall.

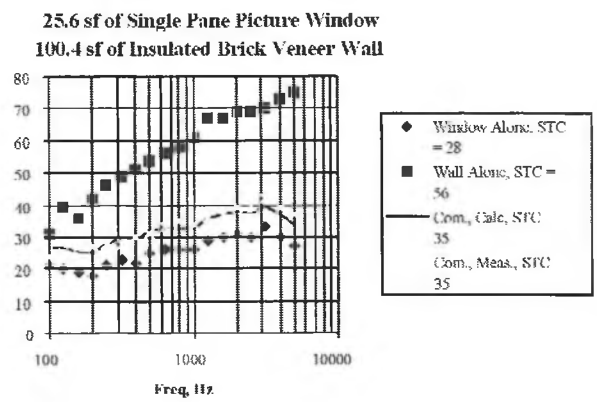

Figure 3. Combination of window and wall.

4 Design Example

The maximum sound power level emitted by two alternate HVAC designs were calculated which meet a 35 dBA background noise design criteria. One is a unit ventilator under a window along a side wall. The other is a central system with energy exchange in the hall way, and air ducted to a diffuser in the center of the ceiling.

Table 1. Maximum HVAC sound power level.

Case	Unit Ventilator Under Window	Ceiling Outlet
Child Seated	31	34
Adult Seated	32	33
Seated Middle of Room	34	
Adult Standing		33

5 Observations

- There is no significant difference between the SEA program and simplified energy balance model for this simple case.
- Direct field effects can be significant especially in the HVAC under window case.
- Ceiling HVAC outlets result in lower levels at the student's ear than under window HVAC unit of equal strength.

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EFFECT OF EARLY REFLECTIONS ON DIFFICULTY OF LISTENING TO SPEECH IN NOISE AND REVERBERATION

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1. INTRODUCTION

When word recognition tests are used as a subjective index for speech intelligibility, it is difficult to describe the differences among sound fields with signal-to-noise ratios above 0dBA because the scores are all very high and usually above 90% in these sound fields[1]. It is necessary to consider another subjective rating to evaluate these sound fields for speech.

Two of listening tests assessed word intelligibility and perceived difficulty of listening to speech in simulated sound fields. The first part was in sound fields including either only direct sound or direct sound with early reflections and under two constant levels of ambient noise. The second part used three types of sound fields: only direct sound, direct sound with reverberation and direct sound with early reflections and reverberation, all with a constant level of ambient noise. Additionally, paired comparison tests were used for some of sound fields in the second part to confirm the significance of some differences.

The purposes of this study are 1) to compare listening difficulty with word intelligibility scores, 2) to show clear evidence that early reflections improve subjective ratings of speech transmission.

2. METHOD

2.1 Sound field simulation procedures

All simulated sound fields used a 7-channel electro acoustic system with loudspeakers arranged around the listener in anechoic room. The loudspeaker located directly in front of the listener produced the simulated direct sound and in some experiments also produced reverberant sound. The other six loudspeakers each produced one early reflection and in some experiments reverberant sound. The early reflections arrived at the listener within the first 50 ms after the direct sound. Each loudspeaker also reproduced simulated ambient noise with a spectrum shape corresponding to that of an NC 40 contour and with measured overall level at the listener of 48dBA. The noise signals to each loudspeaker were not coherent. A noise level of 45dBA was used in the first experiment with the same frequency characteristics as the 48 dBA noise.

2.2 Subjects, speech intelligibility tests and listening difficulty rating

Subjects varied from 22 to 58 years of age and they didn't report any hearing disabilities. More than 11 subjects were used in each experiment.

1st and 2nd experiment: Speech intelligibility scores were obtained using a modified rhyme test[2]. The test words were embedded in the sentence "Word number__ is , write that down" and were spoken by a male talker. Subjects were also asked to rate listening difficulty of each test sentence using the four categories: (1) Not difficult, (2) Slightly difficult, (3) Moderately difficult, (4) Very difficult. Sound fields were presented in random order for each sentence and ten responses were collected per person per condition.

3rd experiment: Psychological scales of listening difficulty were obtained to assess the significance of differences among conditions with Sheffe's paired comparison test [3]. After listening to a pair of sentences presented in different conditions, subjects rated the differences in 1 of 5 categories.

3. RESULTS & DISCUSSION

1st experiment : The first comparisons were based on the results of tests in which subjects performed speech intelligibility tests and listening difficulty ratings for sound fields with varied speech signal-to-noise ratio (S/N) and for two types of reflection conditions. In one series of tests the sound fields consisted of only a direct sound and varied S/N was obtained by varying the amplitude of the direct speech sound with 45dBA and 48dBA of constant noise. In other series of tests three levels of direct speech sound were used and S/N was varied by adding 3dBA and 6dBA of increased levels of early reflections in combination with the same constant noise levels. Figure 1 shows the relations between S/N and listening difficulty rating and word recognition score. Both subjective ratings show good relations with S/N. Although word recognition score reaches 90% above a S/N of 0dBA, listening difficulty is 90% and just starts to decrease its value. Listening difficulty linearly decreases for S/N from −2.5dBA to 15dBA. Listening difficulty better evaluates these conditions than word recognition scores. Comparing the cases with early reflections and those without early reflection, early reflection energy has the same effect on speech intelligibility and listening difficulty ratings as increased direct sound level.

2nd experiment : The second series of conditions was created to confirm that the effect of early reflections also exist in cases including later arriving speech sounds (reverberation). Both word recognition tests and listening difficulty measurements were done as in the 1st experiment. There were 3 series of conditions. In one series the sound fields consisted of only a direct sound and varied S/N was obtained by varying the amplitude of the direct speech sound with 48dBA of constant noise. In the second series the sound fields consisted of a direct sound and two levels of reverberation. Reverberation time was 1.2 second and the reverberant speech level was 53.0dBA for the more reverberant case called "A" and 51.6dBA for the less reverberant case called "B". There were four levels of direct sound (3dBA steps from 49dBA) for each reverberant case. In the third series, two levels of early reflections, which increased the effective signal level by 3dBA and 6dBA, were added to the 49dBA of direct sound condition of each of "A" and "B" and were compared with cases which have the same effective signal level. Word recognition scores reached above 90% in all cases except the lowest S/N case and one cannot differentiate between conditions. On the other hand, listening difficulty ratings vary from 100% to 1.5% as shown in Figure 2. Reverberation effects on listening difficulty as determinant factor for both "A" and "B" case without early reflections. The lowest direct sound case in "B" deviates from the main trend and showed lower intelligibility for its S/N value than expected. In all other cases adding early reflections increased the effective S/N and decreased the resulting difficulty rating. This was thought to be due to the large scatter in listening difficulty scores and this was verified in 3rd experiment using a paired comparison test.

3rd experiment : Two only direct sound with noise cases, three cases with "B" reverberant condition and two cases with "B" reverberant conditions and early reflections were selected from the experimental conditions in 2nd experiment to confirm the effect of early reflections more precisely. 42 pairs of speech were presented twice to 11 subjects. The results in Figure 2 describe that the effect of early reflections are more than the effect of increasing direct sound energy and the difference between the condition with early reflections and without them is significant at the $p<0.05$ level. The results confirmed that the lowest S/N condition with late arrival energy is the most difficult in conditions with late reflections and not the same as the results in the 2nd experiment which included larger scatter.

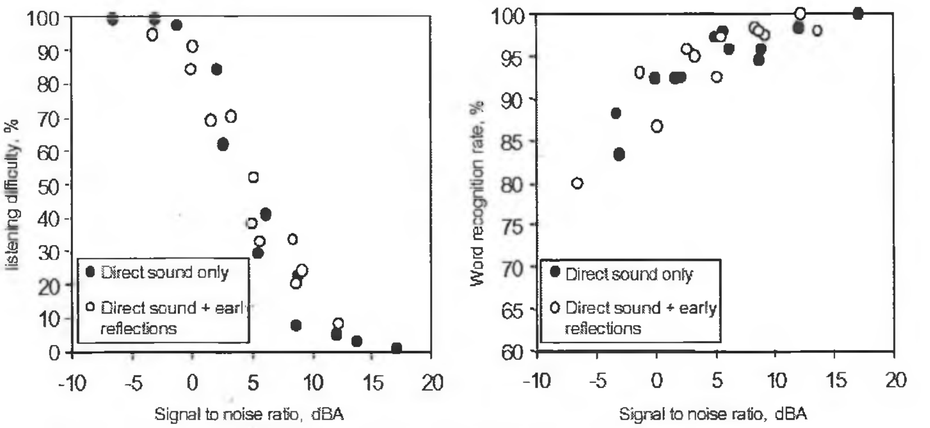

Figure1. Relation between S/N(A) and listening difficulty (left panel) and word recognition rate (right panel).

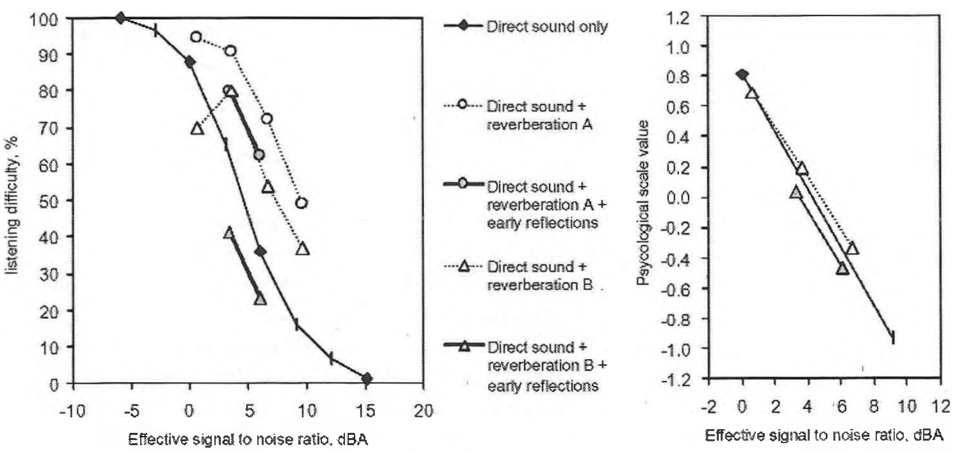

Figure2. Relation between Effective S/N(A) and listening difficulty (left panel) and psychological scale of listening difficulty obtained by paired comparison method (right panel). In calculating Effective S/N(A) values, the reverberant speech sound was excluded from the signal level.

4. CONCLUSION

The results demonstrate that: 1) difficulty starts to decrease for conditions in which word intelligibility scores are above 90%, and difficulty scores decrease to 5% around a 15dB signal-to-noise ratio as in the first experiment; 2) added early reflections increase the effective signal-to-noise ratio much more than in the conditions with reverberation. The second result suggests that the effective benefit of early reflections on listening difficulty ratings is greater than expected from the simple increase in early arriving speech energy.

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Noise Reduction in Early Childhood Centers

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Early Childhood Centers

The ability of a child to achieve to his/her maximum potential in a typical early childhood setting is diminished by the presence of a hearing loss. Even with early identification, intervention, and appropriate amplification, learning will be impeded in noisy centers. On Prince Edward Island, poor acoustic environments create significant barriers to the developmental progress of preschool children. Initially, early intervention with parents of infants and toddlers with hearing loss occurs in their homes. An auditory verbal therapist guides parents in using strategies that maximize their child's use of hearing in learning spoken language through *listening* rather than *watching*. Generally, the acoustic conditions of the homes are controllable and favorable to listening. Parents sit next to the child and speak close to the microphone of the child's hearing aid or cochlear implant at a regular volume. Background noise is minimized. The parent's speech is melodic, rhythmic, expressive and repetitive. Parents use a variety of techniques called *acoustic highlighting* to enhance the audibility of a spoken message.

Transitions to many island day care centers and kindergarten classes provide serious challenges for clear speech perception for all children. However, a poor acoustic environment will be devastating to the speech, language, and auditory skills development of children with fluctuating or permanent hearing loss. A study of classroom ambient noise levels in 33 Florida elementary classrooms found that occupied kindergarten classes were the noisiest (Rosenberg, 1999). These kindergarten classrooms were located within the elementary school buildings. Prince Edward Island early childhood centers are privately owned and operate in community centers, church basements, and modified residential buildings. Often several small activity groups are being led concurrently in an open concept facility. For easy cleaning and to minimize allergies, hard surface flooring and walls are standard. Draperies are avoided as well. However, the abundance of hard surface areas results in increased reverberation.

Ceiling height and angle, the room size, shape, and design, and continuous noise sources need to be evaluated in determining the listening conditions of early childhood centers. If room acoustics are not addressed and appropriate modifications are not implemented, research studies of school age children have identified deficits in student achievement, behavior, attention, persistence, cardiovascular health, and reading achievement (Anderson, 2001). Poor acoustical conditions diminish potential learning, language, and social development for everyone, but will especially inhibit the auditory brain development of children with any degree of hearing loss----and on any given day 1/4 to 1/3 of the children in preschool centers may be experiencing an educationally significant hearing loss in at least one ear (Flexer, 1995).

Because school boards own and maintain their buildings, they have the ability to make modifications and improvements for listening. Most early childhood center owners are renting space within a community facility and operate on severely limited budgets. These environments require creative strategies to improve the acoustics for young children. Additionally, Prince Edward Island's government supports *inclusionary practice*, which holds that all students are entitled to equitable access to learning, achievement and the pursuit of excellence in their education. The practice of inclusion incorporates basic values that promote participation, friendship and belonging. This policy increases the responsibility to evaluate early childhood centers and to make all learning environments for children acoustically accessible for everyone. Legislation of acoustical standards for classrooms is currently under consideration by the United States Access Board. Standards would apply to new school construction and renovations, but " real change in classroom acoustics is more likely if the affects of acoustic interference on behavior and learning were more widely recognized by architects, educators, educational administration, school board members, and legislators" (Anderson, 1999). Educators are often unaware of the link between noise and children's performance.

Prince Edward Island's government also has a strong commitment to literacy development---and the building blocks of reading and writing, the phonemes, need to be richly available for the normal development of phonological awareness. Background noise levels can mask speech sounds. 90% of the speech sounds that carry the meaning of language are consonants. The low intensity, high frequency consonant sounds such as s, t, and f are particularly important for intelligibility. These often mark tense, plurality, and possessives in spoken language and are obliterated by higher intensity background noise. It is not sufficient for spoken messages to be merely *audible*. They must be *intelligible* for auditory discrimination and phonological awareness, the basis for literacy success, to develop correctly (Lundberg, 1988). Excessive reverberation changes the quality of the speech signal and causes an acoustic smearing or distortion of speech sounds. Adult listeners have the advantage of decades of listening experience and language usage knowledge. Their auditory cortex is completely myelenized and developed. Adults are able to make sense of incomplete or distorted auditory input using auditory closure. For children, whose auditory cortex is not fully myelenized until age fifteen, the outcome is decreased auditory discrimination and an inability to accurately perceive when speech sounds begin and end. This has huge implications for accurate speech, vocabulary and language development, the foundation of reading and spelling success (Robertson, 2000).

Children with hearing loss may have difficulty with speech recognition in ideal conditions. If they are at a distance from the teacher, speech intensity may be reduced until sounds are inaudible. The greater the level of background noise and reverberation in the room, the closer these students must be to the speaker. Seating in classrooms is more readily managed than in early childhood centers. Young listeners are required to expend greater *listening effort* in settings with poor acoustics and fatigue more quickly. The poorer the listening conditions become, the greater the listening effort is required. When younger children experience auditory fatigue, they "tune-out" and miss much of the instructional and social opportunities being offered.

Hearing loss itself creates an additional acoustic filter effect on top of the degradation and distortion of the speech signal caused by the physical conditions of the acoustic environment. Hearing aids amplify all sound from both within and outside of the listening area. Hearing aids often increase the hearing and listening difficulties instead of improving them in adverse acoustic conditions. Use of personal and sound field FM systems to improve the signal-to-noise ratio in both early childhood centers and school classrooms gives children who are deaf or hard of hearing a chance to access clear, precise speech input from educators. FM system use will always give the advantage of reducing the negative impact of distance on spoken language. However, it is even more effective in quiet, controlled acoustic environments. Background noise needs to be managed and reduced.

Noise causes listeners to pay attention to the most critical or attention-grabbing aspects of a situation and to ignore more subtle, less immediately relevant cues. Preschoolers are very susceptible to visual and auditory distraction and will lose concentration frequently in the presence of background noise. They completely miss, oversimplify, or make erroneous assumptions about complex social relationships and the rules for interpersonal conduct. Learning and behavior outcomes are diminished. Learning to follow directions and increasing sequential auditory memory depends on concentration, the ability to maintain attention, and complex verbal processing.

Adverse conditions also create irritating, annoying noise that children with an auditory sensitivity will actively seek to avoid. Visual and kinesthetic learners are going to be at a serious disadvantage under noisy conditions. Overcrowding in classrooms creates additional noise as well. Today's teaching styles encourage cooperative, interactive, small group activities. An undesirable outcome is that several small groups conversing simultaneously just create more background speech babble that reverberates, smearing and distorting the verbal messages.

Prince Edward Island has just begun its second year of publicly funded kindergarten. It is a wonderful benefit to Island children to have programs accessible in their communities taught by early childhood educators. The Department of Education is one partner in the process of developing a standard curriculum and defining the standards for teacher training and qualifications. Health codes and building codes exist to make sure our children are in healthy, safe environments. Acoustical standards will additionally assist in evaluations of preschool settings so young children all experience listening comfort and an enhanced accessibility to spoken language, critical conditions necessary for auditory brain development.

Noise level data collected from a sample of kindergartens will be reported during this session.

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SUBJECTIVE MEASURES TO EVALUATE SPEECH INTELLIGIBILITY, QUALITY AND DIFFICULTY IN ROOMS FOR YOUNG AND ELDERLY LISTENERS

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1. INTRODUCTION

Although many studies have evaluated speech transmission in rooms, they have used many different techniques making it difficult to compare their results. This paper presents an overview of several studies carried out by the author in Japan. They include relations between subjective measures to evaluate speech intelligibility (syllabic articulation, word recognition in sentences, sentence recognition) and objective acoustical measures. They also include consideration of different ideas for evaluating speech transmission quality in terms of easiness of speech perception (using paired comparison tests) and perceived difficulty of listening to speech. All experiments were carried out in simulated sound fields. These studies also compared the characteristics of elderly listeners and younger listeners for each measure.

2. SPEECH INTELLIGIBILITY TEST AND EASINESS OF SPEECH PERCEPTION

2.1 Speech intelligibility test

Figure 1 shows the results of speech intelligibility tests of young subjects in reverberant fields [1][2]. The experimental fields were reproduced in an anechoic chamber using an electro-acoustics system. Subjects were asked to write down what they heard for word intelligibility and syllabic articulation and were asked to write the response to questions and commands for sentence indelibility. Word intelligibility and syllabic articulation may not identify significant differences among sound fields even if subjects noticed the difference in conditions. Sentence intelligibility did decrease with reverberation time up to 3 s. Although the results suggest that sentence intelligibility seems to be good measure, it has the following problems : (1) to avoid learning effects, each sentence could be presented only once to each subject, (2) it is difficult to control the degree of difficulty of questions and commands, (3) subject's knowledge could affect the results. These problems cause large scatter in the results.

2.2 Easiness of Speech Perception

Figure 2 shows Easiness of speech perception with the paired comparison method[2][3]. After listening to a pair of sentences presented in different conditions, subjects rated the differences in 1 of 5 categories. The results can describe the significant differences of each pair of conditions and were highly correlated with Speech Transmission Index and some other physical indices. Easiness can identify differences among of each conditions for speech perception more precisely than speech intelligibility measures.

3. EASINESS IN AUDITORIUM AND MULTI DIMMENSIONAL SCALING ANALYSIS

To confirm Easiness can be applied to actual sound fields, Easiness was measured in simulated sound fields reproduced in an anechoic chamber with binaural impulse responses measured in 20 auditoriums with and without a sound reinforcement system [3][4]. Young subjects were used for this experiment. Figure 3 shows the relation between physical values of C50 obtained from impulse responses measured with an omni-directional microphone. Separate relationships are found for without-sound system and with system conditions. To find factors related to the systematic differences of each condition, Multi Dimensional Scaling (MDS) analyses were carried out and obtained a four-dimensional space for Easiness.. Figure 4 presents the plane of the first dimension (D1) and second dimension (D2) with the results of regression analyses of several

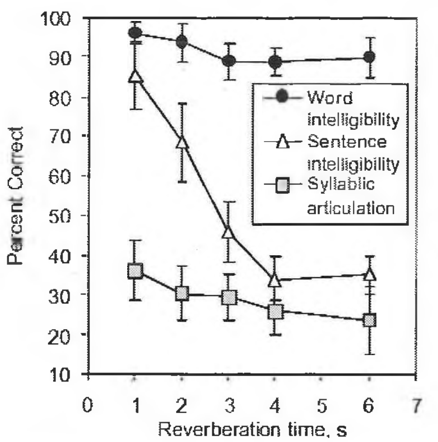

Figure1. Relation between reverberation time and speech intelligibility scores in reverberant fields. Error bar shows standard deviation.

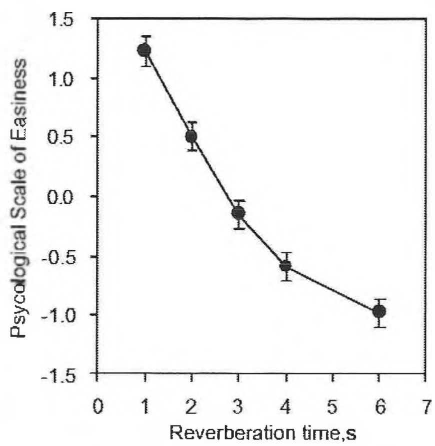

Figure2. Relation between Easiness of speech perception obtained by paired comparison method. Error bar presents yardstick (p<0.05).

acoustical indices with psycho-acoustical measure. It was found that STI and D1 were well correlated (R=0.90) but there aren't any indices describing other dimensions clearly. IACC and wD50 (frequency-weighed D50) show correlation but both of them are not enough to describe other factors. Binaural factors and frequency characteristics are expected to have some effects on Easiness. Another problem with Easiness is that Easiness can evaluate only relatively and cannot indicate how easy is good enough for speech communication in rooms. This is the reason why listening difficulty is presented with absolute numbers for rating the conditions.

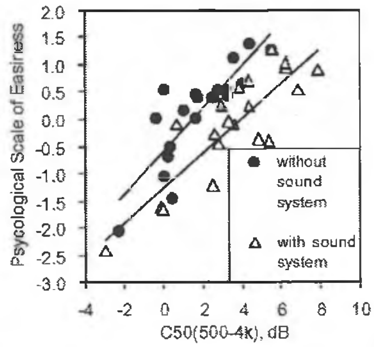

Figure 3. Relation between C50 and Easiness of speech perception in simulated sound fields of auditoriums.

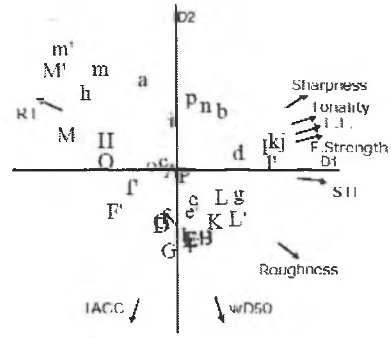

Figure 4. D1-D2 plane as a result of MDS analysis of Easiness. Results of regression analysis with psycho-acoustical measures and acoustical indices are also presented.

4. RELATION BETWEEN SCORES OF YOUNG LISTENERS AND THOSE OF ELDERLY LISTERS

Several speech intelligibility tests were performed in noise and/or reverberation in experimental sound fields with young and elderly listeners[1][5]. Figure 5 shows the relation between speech intelligibility scores of young listeners and those of elderly listeners. It can be said that scores of elderly listeners are 25% lower than those of young listeners for all speech intelligibility measures.

Figure 6 shows the relation of listening difficulty between young and elderly listeners in reverberant fields. As the regression line shows, the variation of listening difficulty for elderly listeners is half that of the young listeners and even when no young listener feels difficulty, 50% of elderly listeners felt difficulty when they heard the speech.

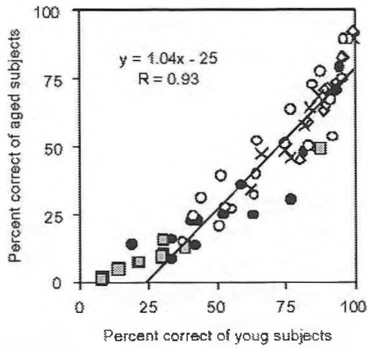

Figure 5. Relation between speech intelligibility scores of young and aged subjects in noise and/or reverberation. Different symbols present different measuring condition and measuring method.

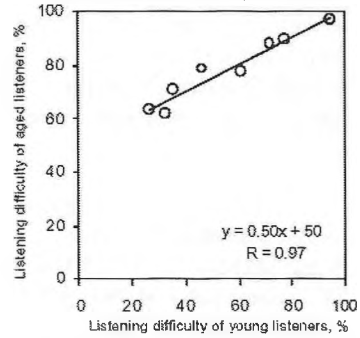

Figure 6. Relation between speech listening difficulty of young subjects and aged subjects in reverberation.

5. SUMMERY

Figure 7 illustrates how an ideal physical index would evaluate acoustical conditions for speech for the young and the aged. Speech intelligibility tests could be applied for zone (a) and listening difficulty used in zone (b) for young listeners. Signal-to-noise ratio works for conditions with ambient noise and without reflections. In terms of S/N, a 25% difference of scores between the young and the aged corresponds to 5dBA signal-to noise ratio [1] in zone (a).

Although zone (c) is good enough for young listeners, conditions in this range will still be inadequate for aged listeners and listening difficulty of aged listeners should be investigated. Easiness is a good measure to study factors that have an effect on speech transmission even if it is relative rating. Each speech rating measure is useful over different specific limited ranges of physical measures. Thus each best describes different aspects of speech transmission conditions.

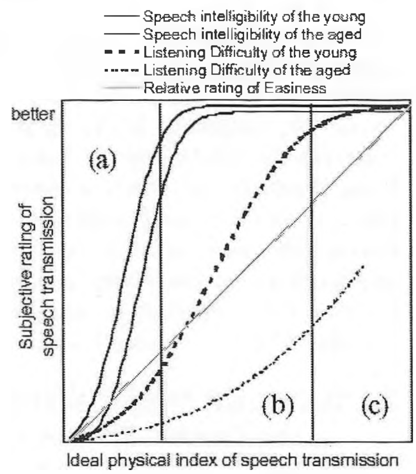

Figure 7. Illustration of relation between ideal physical index of speech transmission and subjective ratings of speech transmission.

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Crawler Transporter Noise Control Study and Noise Control Design

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1. INTRODUCTION

The Crawler Transporter is the world's largest tracked vehicle known weighing 2,721 metric tonnes with a length of approximately 40 metres and a width of approximately 35 metres. The Kennedy Space Center has two Crawler Transporters that were built by the Marion Power Shovel Company in the 1960's for the Apollo/Saturn V Program. The Crawler Transporters have been maintained and retrofitted for use in the Shuttle Vehicle Program. The overall Crawler Transporter design and propulsion systems are relatively unchanged from the Apollo/Saturn V program to the present.

The Crawler Transporter is electrically driven by 16 traction motors. The power is provided by four on board diesel gensets. Two 2750 horsepower Alco engines power DC generators and two 1065 horsepower White Superior engines power AC generators. In addition to supplying the needs of the traction motors the on board generated power also supports the needs of the hydraulic leveling and jacking system, steering system, lighting and ventilation systems. The Crawler Transporters are constructed with redundancy built into the design of all major operating systems.

The work force on board the Crawler Transporter during a rollout is approximately 18 engineers, technicians and support personnel. The United Space Alliance (USA), owned by Lockheed Martin and Boeing, is the prime contractor for NASA's space shuttle program. USA retained Noise Solutions Inc. (NSI), a company out of Calgary that provides turn-key noise control solutions with guaranteed results. NSI commissioned Faszer Farquharson to evaluate and develop conceptual noise control measures for NASA's two crawler transporters in order to reduce the noise level exposure of the crawler work force.

2. CRAWLER NOISE SOURCES

The Crawler Transporter is powered by four onboard engine powered electric generators. Two 2750 horsepower Alco engines supply the DC power requirements and two 1065 horsepower White Superior engines supply the AC power requirements. The engines are all housed in a single room within the superstructure of the Crawler Transporter designated as the Crawler Engine and Equipment House. The engines are situated at 90 degrees to the travel of the Crawler Transporter with the Alco engines on the outboard ends and the White Superior engines located on either side of the center of the Crawler Transporter superstructure. The engine and generator sets are mounted on vibration isolators to the floor of the Crawler Transporter Engine and Equipment House. Combustion air for the engines is drawn from the interior of the Crawler Transporter Engine and Equipment House. The engines all exhaust to the underside of the Crawler Transporter superstructure with an exhaust outlet found near each of the four corners of the unit. Large radiators for each of the four engines are situated outside the Crawler Transporter Engine and Equipment House at the travel or drive ends of the Crawler Transporter.

The hydraulic pump systems are housed in the center of the Crawler Transporter Engine and Equipment House opposite the control room. The jacking and leveling hydraulic system (JEL) is located on the opposite side of the Crawler Transporter from the control room. From this location hydraulic fluid under high pressure is directed to the jacking and leveling systems located at the four corners of the Crawler Transporter. The high pressure hydraulic lines take a variety of paths starting inside the Crawler Engine and Equipment House exiting to the underside of the superstructure then turn up the outside of the superstructure to the jacking and leveling system at each corner. The JEL system pumps are skid mounted and sit directly on the floor of the Crawler Transporter Engine and Equipment House. The high pressure JEL lines are rigidly connected to the floor of the house and to the superstructure of the Crawler Transporter. The JEL system return lines follow a similar path back to the large hydraulic fluid reservoir located centrally in the Crawler Transporter Engine and Equipment House.

The steering hydraulic system is similar in design to the JEL system. The steering system pumps are located between the control room and JEL system pumps. This system is also skid mounted directly to the floor of the Crawler Transporter Engine and Equipment House. The supply lines are again rigidly mounted to the superstructure of the Crawler Transporter.

The super charger system supplies hydraulic fluid from the reservoir to the intake of the JEL system and steering system pumps. The supercharger pumps are skid mounted to the floor of the Crawler Transporter Engine and Equipment House. Supply lines leading to the JEL and steering hydraulic pumps are rigidly mounted to the floor and or the skids of the pump systems.

The ventilation for the Crawler Transporter Engine and Equipment House is supplied by a number of 36" and 40" inch diameter fans located along the sides of the Crawler Transporter Engine and Equipment House. The ventilation fans are generally located at positions where the cooling air from the fans will sweep the engine and generator casing areas and the hydraulic pump areas. The cooling air is to exit through grated openings in the floor of the Crawler Transporter Engine and Equipment House. The ventilation fan system is supplemented by a number of portable propeller fans that were moved as required to provide additional air movement inside the Crawler Transporter Engine and Equipment House. The four man doors leading into the Crawler Transporter Engine and Equipment House were open for ventilation purposes during the rollout.

Noise sources associated with the four trucks include the movement of the trucks, the operation of the jacking and leveling system at each leg, the steering system, the truck propulsion motors, mechanical noise associated with the movement of the shoes over the boogies and sound of the crawler way gravel being crushed.

3. CRAWLER OPERATIONS

The rollout operation for a fully loaded crawler requires the services of approximately 18 engineers, technicians and support personnel. The operators are divided between stationary positions and roaming positions. Stationary positions include personnel in the control room, the active steering cab and in the Crawler Transporter Engine and Equipment House overseeing the operation of the engines and hydraulic systems. Roaming positions include each of the four trucks, the underside of the superstructure and the catwalk around the exterior of the Crawler Transporter Engine and Equipment House. Personnel stationed inside the Crawler Transporter Engine and Equipment House are generally the individuals subject to higher than desired sound levels. The personnel located in the control room and the steering cabs are generally more concerned with reduced noise levels for better communication purposes. Communication between Crawler Transporter personnel is facilitated by the use of radio headsets for all personnel outside the control room.

A rollout operation takes from 14 to 16 hours depending on how many stops occur along the 3 mile trip from the Vehicle Assembly Building to the launch pad. Stops for any reason are diagnosed istantly and any repairs required are undertaken immediately by the crawler operating crew.

4. MEASUREMENTS AND DATA ANALYSIS

A requirement of USA was to undertake measurements of a fully loaded working crawler during a rollout without interfering with its operation and design the noise control to reduce the sound levels under the fully loaded conditions. In order to obtain the required detailed sound level data required to design and engineer noise control measures, a combination of sound pressure level measurements and sound intensity measurements were undertaken.

The sound pressure level measurements were conducted with a Brüel & Kjær Model 2260 Precision Real Time Sound Analyzer and a Brüel & Kjær Model 4189 Microphone. The sound intensity level measurements were conducted with two Brüel & Kjær Model 2260 Precision Real Time Sound Analyzers each equipped with a Brüel & Kjær Model 4197 Microphone Pair mounted on a 2683 Sound Intensity Probe. These systems were used to measure and record 1/3 octave band frequency sound intensity and sound pressure level spectra.

Sound pressure level measurements were undertaken of noise sources that did not vary with a fully loaded crawler and that could be isolated by turning equipment on and off. The sound intensity measurements were taken of equipment that produced maximum sound level when the crawler was fully loaded and that could not be isolated from other operating equipment.

The sound pressure level measurements were presented as 1/3 octave sound pressure level graphs and the sound intensity measurements were presented as 2D contours mapped onto an outline of the measured equipment and 3D sound intensity contours providing and indication of sound energy flow. This information was used to evaluate the significant noise sources, determine conceptual noise control measures and predict potential noise reductions for the various systems.

5. NOISE CONTROL MEASURES

The conceptual noise control systems that were presented included an acoustically enclosed walkway and vestibule system that would compartmentalize the crawler engine house, ventilation and radiator fan noise control, JEL hydraulic system noise control including isolation of hydraulic lines, equipment skids and in-line hydraulic silencers, upgraded engine exhaust mufflers, upgrading of the control room/engine room demising wall and engine house acoustical treatment.

The conceptual noise control measures were presented to USA and NASA staff for review and it was decided to proceed to an engineered noise control design stage for all of the conceptual noise control measures with the exception of the radiator fans and the compartmentalization of the engine house.

Upon completion of the engineered noise design stage orders were placed for upgraded mufflers and silenced ventilation fans for one of the crawler transporters. The upgraded mufflers have been fabricated and shipped to the Kennedy Space Centre and are ready for installation.

LARGE HAUL TRUCK MUFFLERS 'THAT WORK'

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1. INTRODUCTION

The Genesee Coal Mine, in operation since 1989, is located approximately 55 kilometers south west of the city of Edmonton, Alberta. Operated by Fording Coal Ltd., the operation is a joint venture between Fording Coal Limited and EPCOR Generation Inc. The average production per year is 3.5 million tonnes of sub-bituminous coal and approximately 23 million BCM's of waste material resulting in an average strip ratio of 6.5 BCM/tonne. The mine is found south of the North Saskatchewan River within the county of Leduc. The mine site is adjacent to the existing EPCOR Generating Station east of Secondary Road 770. The Genesee Mine Permit Boundary covers approximately 7,300 hectares while the actual mine encompasses approximately 1,700 hectares. Lands within the mine permit boundary are mostly owned by the City of Edmonton. Approximately 90 rural residences are located within 1 to 6 km of the power generating station and the mining operations.

Genesee Power Project Advisory Committee (GPPAC) has fulfilled many functions since its inception. Besides keeping local residents informed about project activities, GPPAC has also provided a forum through which local residents, Fording Coal Limited: Genesee Operations, and EPCOR Generation Inc.: Genesee Generating Station discuss issues of importance to them. During 2000, one issue that GPPAC members dealt with was the exhaust noise emitted by the mine's three Komatsu 510E, 150 ton, coal haul trucks. Committee members asked if there was anything that could be done to reduce the truck noise. During a conversation between Whitewood mine manager, Al Brown, and Genesee mine manager, Brad Johnston, a possible solution was discussed and brought forth for the committee's consideration. Noise Solutions Inc. (NSI) of Calgary was identified as the company who had helped Whitewood Operations reduce the noise emitted by the 8200 dragline. Rod MacDonald of NSI was contacted and the company was hired to begin a pilot project to design and install an enhanced muffler system on the mine's trucks.

2. MINING OPERATIONS

Two Marion draglines are used for all waste rock removal and coal production. The larger M8750 model is used in higher strip ratio areas while a M8200 is used in areas of shallower cover. The waste rock mined is spoiled with the draglines into waste piles as part of the regular mining sequence. Exposed coal is lifted with the draglines and placed in piles onto constructed bench floors. The coal is then hauled to the generating station using a 510E Komatsu truck fleet at a rate of approximately 10,000 tonnes/day.

3. MEASUREMENTS AND MODELLING

In order to determine what noise sources at the mine were significant, NSI commissioned Faszer Farquharson to conduct a noise impact assessment (NIA) of the mining operations to the surrounding community. One of the first steps in a NIA is to undertake detailed sound level measurements of all the significant noise sources at the mine. Mining equipment is not the easiest type of equipment to measure because it is always moving. A combination of close-in measurements of various equipment components as well as further away sound propagation measurements were undertaken. The information obtained was reviewed and calculations undertaken to determine sound power levels of the equipment, some as a unit and some as various components of the units.

The sound power levels were used as input parameters for ENM, a sound propagation model along with the terrain and the location of the residences. Detailed mine plan information and locations of the equipment during mining were required as the bench heights with respect to the surrounding topography can create significant sound barriers. Factors such as topography, vegetation and prevailing wind direction needed to be combined with linear distance from the noise source to determine the most affected residences and which noises were the most significant at those points. The topographical features of the North Saskatchewan River valley were examined with particular care: predicting the effects of water and meandering valleys on the acoustics of the area proved to be particularly complicated. The mobile nature of mining also needed to be considered to determine the most significant noise impacts. In the short term, the equipment moves as its working thus creating moving 'point sources' of noise. In the longer term, the mine advances over a period of years thereby changing the distance between individual residences and the mining activities.

The sound criteria used in a given jurisdiction must also be fully understood. Average sound levels as compared to statistical or peak sound levels can significantly change the sound power levels used in the model. The Alberta criteria, outlined in the EUB ID-99-8 Noise Control Directive are 50 and 40 dBA Leq, an energy average, for the daytime and nighttime periods respectively. Thus truck sound power levels were adjusted to reflect the mobility of the trucks in relation to the receivers and number of trips that would be completed during a typical nighttime shift.

3. RESULTS

The results of the model provide both octave band sound pressure levels as well as overall dBA sound levels order ranked by sound level from highest to lowest at the receiver locations. The following dBA sound level order ranked table is extracted from the NIA report.

Order Ranked Sound Pressure Levels
Residence 1, Year 1999 Night Shift

Source	Source Contribution (dBA)
Electric Shovel Ventilation	38.6
Marion 8750 Dragline Bleed Tubes	34.7
Komatsu Dozer	32.5
510 Haul Trucks	32.1
Marion 8750 Dragline Inlet Air House	31.0
Electric Shovel Mechanical Noise	28.8
Genesee Power Plant	28.4
Marion 8750 Dragline Ventilation	24.4
Caterpillar D11 Dozer	23.5
Marion 8750 Dragline House	23.5
Marion 8750 Dragline Vent Fan Casing	20.9
Elk Point Telfordville Gas Plant	14.8
Sum	39.5

The significant noise sources can be readily seen from the order ranked tables, however different sources have different rankings at the modelled residences. Noise modelling was consistent with the resident's testimonials in that the closest residences were not necessarily the most affected with respect to noise from the mine. In this case, the one residence that had a predicted sound level over 40 dBA was to be purchased before that year of mining, thus there was no exceedance of criteria.

4. MUFFLER DESIGN

Based on the sound propagation results indicating no exceedance of the criteria, Fording Coal was able to make a decision to attack the coal haul truck mufflers rather than the highest order ranked source. This decision was based on the haul trucks landing in the top five most significant noise sources source as well as being flagged by the residences as having a high annoyance factor. If there had been an exceedance of the criteria it may have been necessary to attack another source along with the truck mufflers such as the dragline ventilation fans or shovel ventilation fans in order to reduce the overall sound level to meet the criteria.

The noise measurements had indicated that the engine exhaust was the most significant source on the haul trucks and thus this source would receive noise control. Other sources that had been measured were the engine casing noise and radiator fan.

The next step was designing a muffler for the trucks that would provide the required noise attenuation, fit the chassis of the truck and not exceed the engine back pressure requirements.

An NSI muffler designed by Faszer Farquharson that had proved successful for large stationary engines was modified into a dual exhaust design and squished and squeezed to fit the available space under the box of the truck. One unit was manufactured for test purposes. This unit was connected to the existing exhaust outlets using flexible hose. This test was very successful indicating a high attenuation level and low pressure drop. The next step was to install the unit in the truck and undertake a trial period. This again proved successful and the all three trucks have had the mufflers installed. The actual attenuation results were not indicative of the performance of the muffler by itself. To obtain more definitive data, a test was conducted that isolated the exhaust noise from the rest of the truck noise. This was accomplished by parking the truck in the workshop, closing the shop doors and running the exhaust out of the shop with flexible pipe.

The before and after overall sound level results for a truck pass-by including all truck sources indicate a 4 dBA reduction. The engine exhaust source indicated a 31 dBA reduction. Subjectively, the muffler has drastically reduced the engine "bark" that was heard by the surrounding residences and very positive comments have be received regarding the reduced coal haul truck noise levels.

An expansion in the output capacity of the Genesee Generating Station from 800 MW (gross) to 1295 MW (gross) has been approved. The expansion requires a 50% increase in the output capacity of the mine. The mine expansion would be accomplished through the addition of a truck and shovel fleet. Noise control programs being considered for the post-expansion era include, bleed tubes on the 8750 and 8200 draglines, house pressurization fans on the 8750 and 8200 draglines and the waste shovel, and the exhaust noise of the waste haul trucks.

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ACTIVE SOUND RADIATION CONTROL OF CYLINDRICAL STRUCTURES USING PIEZOELECTRIC ACTUATORS

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1. INTRODUCTION

The placement of control actuators in an active noise control system can significantly affect the control performance. Considering the high cost of active control experiments, the optimal design of the physical control system is essential to ensure the efficiency of an active noise control system before it is implemented.

In simple cases like the sound radiation from plates into a free field, the optimal position of the actuators could be determined using sound radiation analysis and modal analysis. However, for a complex enclosure, it would be a wise choice to find the optimal positions of actuators using some optimal approach, because of the inherent complexity of structural acoustic coupling.

Due to their advantages over the gradient-based approach of being robust and highly efficient in dealing with complex multi-model nonlinear problems, Genetic Algorithms (GAs) have been recognized by many researchers as a promising tool in the optimal design of active noise and vibration control systems. However, most of the reported work focused more on the GAs than on the design of the ASAC system itself, and PZT actuators were simplified to be point forces. Obviously, the control effect of PZT actuators is different from that of point forces. The fact that PZT actuators generate distributed effects on the structure over the covered area makes the problem of optimal placement design more critical.

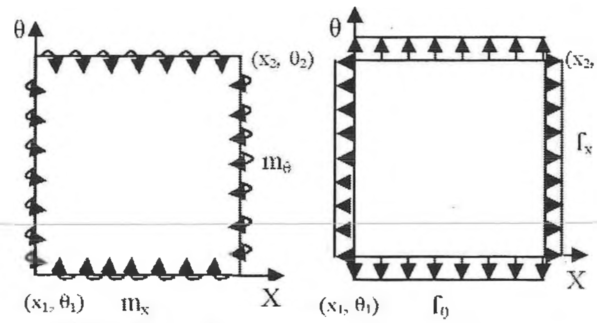
In this paper, the location optimization of PZT actuators in an ASAC system of a cylindrical shell with an internal floor partition is investigated using Genetic Algorithms. The primary physical model was previously developed to simulate the sound field inside an aircraft cabin. In the present work, the effect of PZT actuators was added to the model through a bending model and an in-plane force model [1]. The control performances of both models were assessed and compared. Considering the requirement of practical application of ASAC, the optimal configuration of control system obtained at a single frequency was also tested over the low frequency range below 500Hz.

2. METHOD

2.1 Models of PZT

Two analytical models of PZT actuators (i.e., bending and in-plane force models [1]) were employed here to simulate the effect of PZT actuators attached on opposite sides of the cylinder wall. The bending model simulates the effect of two actuators operating out of phase, which produces an axial stress distribution varying linearly through the thickness of the cylinder wall and creates bending about the middle surface of the cylinder. The loading produced on the cylinder by the bending model is approximated by a line moment distribution acting on the perimeter of the piezoelectric patch area (Figure 1 (a)).

With the in-plane force model, the actuators are assumed to operate in-phase. When this model is implemented on a flat plate, only in-plane displacements are produced. In the case of a cylinder, however, bending displacement is also induced by the in-plane deformation due to the curvature effects intrinsic to shells. Hence, it is possible to employ this model in the ASAC of a cylindrical structure. The in-plane loading can be simulated by a line force distribution on the perimeter of the patch area (Figure 1 (b)).



(a) Bending model (b) In-plane force model

Figure 1: Model of PZT actuators

2.2 Placement optimization of PZT actuators

The vibroacoustic model of the investigated structure was presented in detail in the previous work [2]. In the optimization of the actuator placement using Genetic Algorithms, the reduction of the acoustic potential energy, given by the difference between the acoustic potential energy level before and after control, was employed as the evaluation function to achieve the global control of the interior sound field. For a given configuration of actuator placement, the optimal control input was determined using the quadratic optimization approach minimizing the acoustic potential energy in the enclosure.

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3. RESULTS

The investigated structure and the coordinate system are shown in Figure 2. In simulations reported hereafter, the disturbances were assumed to be point forces; and control actions were provided by PZT actuators. Clearance distance between the disturbance and the control actuator of 0.05m in the longitudinal direction and 10 degrees in the circumferential direction was imposed to avoid placing a control actuator too close to or even overlapping a disturbance force. A complex case involving 10 disturbance forces with random amplitudes and phases and 4 PZT actuators was investigated. The size of the actuators is 0.05m x 0.02m covering a sector angle of 4°.

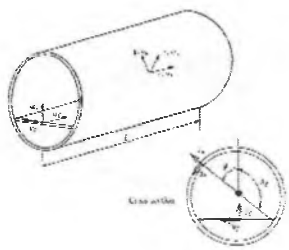
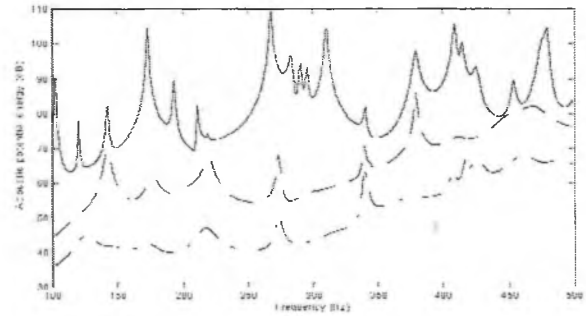


Figure 2: Schematic diagram of investigated structure

PZT actuators were assumed to operate as a bending model and an in-plane model, respectively. Optimization was carried out at the acoustic resonant frequency of 283.7 Hz. The reduction of the acoustic energy with the optimal PZT actuators operating as a bending model was up to 41.66 dB. As the PZT actuators operate as an in-plane force model, the reduction of the acoustic potential energy reached 53.56 dB, which is 11.9 dB more than that obtained in the case of a bending model.

The above result was obtained for one single frequency. Therefore, it is no surprise that the optimal configuration performs well. If it is only effective at one single frequency, a control system would be highly limited in a practical application. Therefore, it would be interesting to verify whether the optimal configuration obtained could also be effective for other frequencies. To this end, the control performance of the same control system was tested in the frequency range below 500 Hz (Figure 3). Since there are no other resonant frequencies below 100 Hz, the frequency range examined was set from 100 Hz to 500 Hz. From Figure 3, one can observe that there is significant sound attenuation over the whole frequency range of interest, whether the PZT actuators operate as a bending model or an in-plane force model. This overall performance could be explained by the structural coupling analysis of the investigated structure [3], which showed that, in this frequency range, the sound field is mainly contributed by a limited number of structural modes with high radiation ability. Therefore, one could predict that the ASAC control system designed at one acoustic resonant frequency could also be effective over the low frequency range under 500 Hz.

From Figure 3, it can also be observed that, over the whole frequency range of interest, the in-plane force model has much better control performance than the bending model. The typical difference between the two configurations oscillates between 10 and 15 dB. This observation can be easily understood. In fact, due to the strong membrane effect, the in-plane motion of the shell is strongly coupled to the out-of-plane motion at low frequencies. When PZT actuators are attached to the surface of a cylindrical shell, the generated in-plane force could produce a more favorable distribution of the low order circumferential modes of the investigated structure and hence couples much more efficiently with the low order interior cavity modes than does the bending model.



Before control - - - - - Control with bending model
-·-·-·- Control with in-plane force model

Figure 3: Control performance of optimal PZT actuators

4. DISCUSSION

This study demonstrated the encouraging performance of PZT actuators in active control of interior noise after placement optimization over the low frequency range under 500 Hz, as well as at the single design frequency. In terms of PZT actuators attached on a cylindrical surface, the in-plane force model has much better control performance than the bending model.

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ACKNOWLEDGEMENTS

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PRELIMINARY INVESTIGATION OF ACTIVE CONTROL OF DIPOLE NOISE SOURCES

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1. INTRODUCTION

Most studies of active noise control (ANC) in open space have concentrated on reducing the primary field generated by a monopole sound source. Although many practical noise sources may be represented by a monopole source at low frequencies, more complex noise sources, such as aircraft propellers, are better represented by multipole sources (e.g. dipole, quadrupole, etc.). Unlike the sound field produced by a monopole source, multipole sound fields are orientation dependent and do not display cylindrical symmetry [1].

This research was divided into two main components: creating a sound source with dipole directivity; and investigating the effectiveness of a multi-channel control system on the dipole source.

During the experiments, the primary source was located on the central axis of the control system, and the control sources and error microphones were placed in parallel arrays, as shown in Fig. 1. The distances between adjacent control sources and error microphones were kept equal, such that $r_{ss}=r_{ee}$ for all test cases. Using the local control method, the quiet zone achieved by the control system in the area surrounding the error microphones was studied.

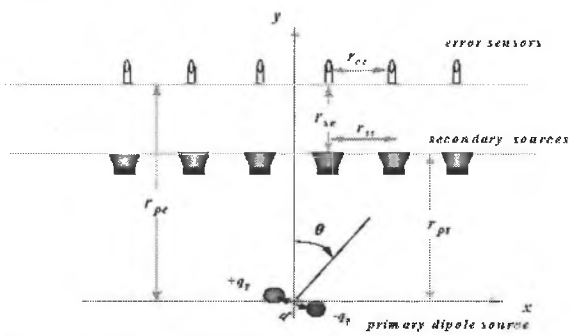

Fig. 1. Multi-channel ANC system for a dipole noise source.

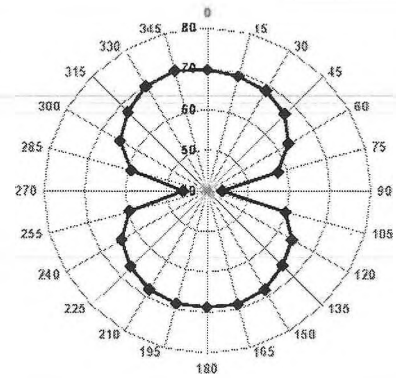

Fig. 2. Directivity at 200 Hz of a speaker placed at the center of an enclosure.

2. CREATION OF A DIPOLE SOURCE

The first sound source that was used was an enclosed loudspeaker with its back cover removed. The directivity pattern had the general shape of a dipole, but was asymmetrical. The radiation from the front of the speaker was about 5 dB stronger than to the back, and the radiation to the sides of the speaker was only 8 to 10 dB lower than to the front and back. In order to improve the symmetry of the radiation pattern, a four-sided enclosure was built that allowed the speaker position to be adjusted. Pushing the speaker back into the box compensates for the stronger directivity to the front, and improves the nulls to the sides of the source. The directivity of the source at 200 Hz is shown in Fig. 2. The pattern is now symmetric, with a drop of about 25 dB to the sides. This sound source is thus a good approximation of a dipole at low frequencies.

3. ANC EXPERIMENTS

ANC experiments were performed in an anechoic chamber at the University of British Columbia. The dimensions of the chamber are 4.7x4.2x2.2 m^3. The primary source was located 0.5 m from the back wall of the chamber at a height of 1.25 m. The control speaker array consisted of three enclosed loudspeaker (monopoles), spaced at equal distances from each other. The error-sensor array of three equally-spaced microphones was placed in front of the control speakers. The distances between the control speaker array and primary source, and between the control source and error microphone array, were kept constant at 1.0 m. The commercially available multi-channel EZ-ANC was used as the ANC controller. The test signal was a pure tone of 200 Hz, which is above the cut-off frequency of the anechoic chamber. The primary source was driven by a signal generator, which also provided a reference signal to the controller. The noise measurements were carried out in a plane at the height of the control system at 0.5 m intervals, yielding 45 measurements.

The noise attenuation was measured for three orientations of the dipole: θ=0, 45, and 90°. As shown in Fig. 1, θ=0° corresponds to when the lobes of the dipole are aligned along the x-axis, such that the null of the dipole is directed towards the control system. The separations of adjacent control channels were determined from simulations, as discussed in [2].

Fig. 3 shows the attenuation for the dipole oriented at 0°. The quiet zone created by the control system is very small, though the sound increase in other areas is also very small. The attenuation achieved in most areas was only about 5 dB.

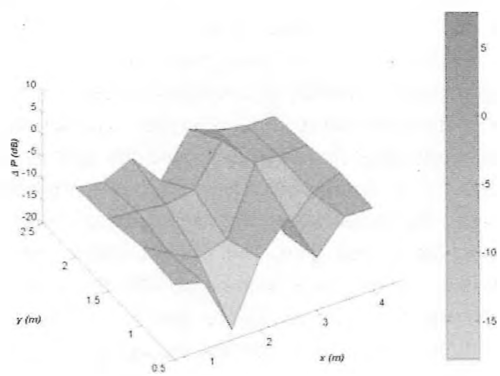

Fig. 3. Measured noise attenuation of a 3-channel control system for a dipole source oriented at 0° (null pointing towards the control system) with r_{ss} = 0.67m (0.4λ).

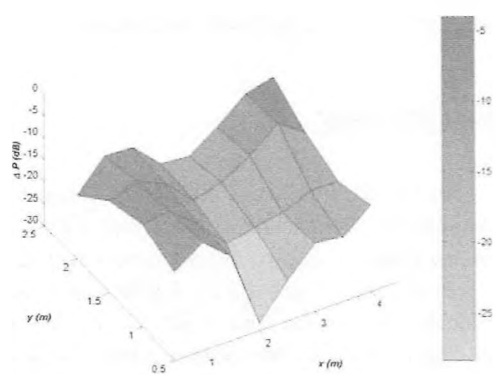

Fig. 4. Measured noise attenuation of a 3-channel control system for a dipole source oriented at 45° with r_{ss} = 1.07m (0.64λ).

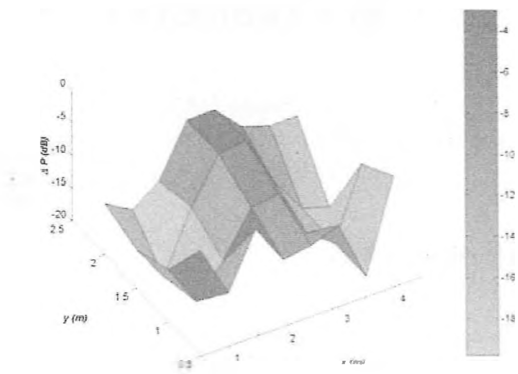

Fig. 5. Measured noise attenuation of a 3-channel control system for a dipole source oriented at 90° with r_{ss} = 1.25m (0.75λ).

The size of the quiet zone increases when the dipole orientation is changed to 45°, as shown in Fig. 4. Attenuation was achieved at all of the measurement positions.

The maximum noise attenuation and largest quiet zone were measured when the dipole was oriented at 90°, as shown in Fig. 5. Attenuation was again achieved at all measurement positions, and the quiet zone is symmetrical.

4. CONCLUSIONS

This preliminary investigation of the effectiveness of a control system on a directional dipole noise source demonstrates that a primary source with non-cylindrical radiation directivity does affect the performance of the control system. The control system works more effectively when it is used to create quiet zones in the area with the strongest primary-radiation directivity. It has also been shown that the quiet zone shifts slightly in the direction with the strongest radiation directivity.

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The authors wish to thank Perry Wong for his assistance with the ANC experiments.

THE EFFECT OF BACKGROUND NOISE ON SOUND POWER IN BOTH A REVERBERANT AND ANECHOIC ENVIRONMENT

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1. INTRODUCTION

Determination of sound power of a source is a useful quantity since sound power theoretically provides a measurement of the amount of sound energy that is radiated from the source independent of its surroundings. While sound pressure level (SPL) measurements are a good indicator of human hearing response, they are highly dependant on the acoustic environment in which they are made, whereas sound power is a characteristic of the source only.

Sound power is not measured directly, but is calculated by sound intensity measurements which is a measure of the radiating power through a surface area. One of the significant advantages of sound power determination, using intensity measurements, is that stationary background sources have negligible influence on the results. This study investigates the validity of such a statement by measuring the effect that an increasing background noise has on a steady broadband noise source through the application of noise intensity measurements in both a reverberant and anechoic environment.

2. THEORY

Sound intensity is calculated from the time-averaged product of the measured pressure and particle velocity. This is derived from:

$$I=\frac{W}{A}=\frac{F\times v}{A}=\frac{F}{A}\times v=P\times v$$

where I is intensity, W is power, A is area, F is force, P is pressure and v is velocity.

Pressure is easily measured by a single microphone but velocity is not. However, with two closely spaced microphones, the particle velocity can be related to the instantaneous pressure change, or pressure gradient, across the distance between the microphones. Knowing the pressure gradient and density of the fluid medium, the particle acceleration can be calculated with Euler's equation shown as:

$$u=-\int\frac{1}{\rho}\frac{\partial p}{\partial r}dt$$

$$I=\overline{p\times u}=-\frac{p_A+p_B}{2\rho\Delta r}\int(p_B-p_A)dt$$

The particle velocity is then derived by integrating the acceleration. Intensity is then given as:

Here $(P_A+P_B)/2$ represents the average pressure which is simply the arithmetic average of the pressure measured by each of the two microphones. Similarly, the term inside the integral is the integration of the finite difference approximation applied to Euler's equation.

As already stated, one fundamental advantage of using intensity measurements for the determination of sound power of a source is that steady background noise has no effect on the intensity measurements. To illustrate this idea, imagine a source within an enclosed surface area for which intensity measurements are conducted. The intensity would then be multiplied by the area to find the total sound power radiated. If the source were now moved outside the surface enclosure, the radiating energy would enter one face of the enclosure then exit from the diametrically opposite face. Given that intensity is a vector quantity, the total net energy contribution from the enclosure due to the external source would then be zero. Effectively, background noise within a measurement environment can be regarded as the external source described above, and therefore, has no effect on the determined sound power. One condition of this is that the background noise must be steady in nature. If it is not, the intensity due to noise entering one side of the enclosure may not equal in magnitude to the intensity exiting the other side, and thus, resulting in a net level other then zero.

3. PROCEDURE

To test the validity of the above statement, intensity measurements were made of a speaker source producing white noise signals. A 6 by 6 grid with a surface area of 1 square foot on the top and a 3 by 6 grid with a surface area of ½ square foot on the four sides was placed around the speaker. Figure 1 shows the speaker with the grid enclosure.

The source speaker was played at a constant sound pressure level (SPL) of 90 dB, measured at a distance of one metre from the speaker. The background noise, also a white

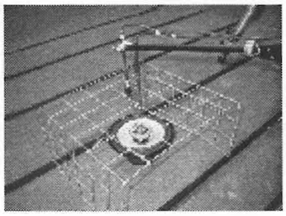

Figure 1: Source Speaker with Grid Enclosure

noise source, was generated by two large loudspeakers placed in opposite corners of the room. The first test involved playing the background noise at an SPL of 80 dB, as measured at the same location as the source speaker, in a semi-anechoic room. Intensity measurements, using the discrete point averaging method, were made of the source speaker using a Hewlett Packard 3569A analyzer which also calculated the overall sound power of the source. The test was then repeated five more times with the background noise increased by 5 dB each time until the background noise was 105 dB, or 15 dB greater than the source speaker. The entire procedure was again repeated, only this time in a highly reflective room. This room was used to investigate the presence of any difference in the results in a reverberant environment.

4. DISCUSSION OF RESULTS

Figure 2 is a graph of the overall sound power level of the source speaker with the various background noise levels in the semi-anechoic room. It can be seen that very little difference exists in the overall sound power of source for the increasing background noise levels. The maximum sound power level was 81.94 dB and the minimum was 81.53 dB giving a negligible difference of about 0.4 dB. In fact, it is usually accepted that "sound power can be measured to an accuracy of 1 dB from sources as much as 10 dB lower than the background noise." [1] Here it is within 0.4 dB with a source 15 dB lower than the ambient. Examination of Figure 2 also illustrates that there was no straight line trend from the low to high background sound level with the maximum level occurring in the middle of the graph suggesting that any differences are most likely random and not influenced by the background noise.

Figure 3 shows the overall sound power level of the source speaker with the six different background noise levels in the reverberant environment. The maximum sound power level was 78.66 dB and the minimum was 77.46 dB with a difference of about 1.2 dB. These differences show a downward trend in source sound power with increasing ambient noise. This is opposite that what would be expected if the background noise influenced the source sound power. Instead, the sound power level should increase. This serves only to suggest that the 1.2 dB difference was due to some other effect other than the background noise. It is suspected that inaccuracies in the source power control influenced these results.

Also noted was that even though the speaker was played at the same SPL measured one metre from the source in both rooms, the power level measured in the semi-anechoic room was greater than the reverberant room. Recall that SPL is influenced by environment. Here the reverberant environment reflected the source energy adding to the SPL while the anechoic room absorbed energy thus requiring more energy output of the source to attain the 90 dB SPL. This reinforces the fact that both environment type and background noise have no appreciable effect on the sound power results.

5. CONCLUSIONS

Sound power determination of a source through intensity measurements have the distinct advantage that they can be determined without influence of background noise and environment. This exercise has clearly demonstrated this fact by showing that a stationary background noise up to 15 dB greater than the source under consideration has no effect on the sound power results in both a reverberant and anechoic environment.

6. REFERENCES

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Figure 2: Semi-Anechoic Environment

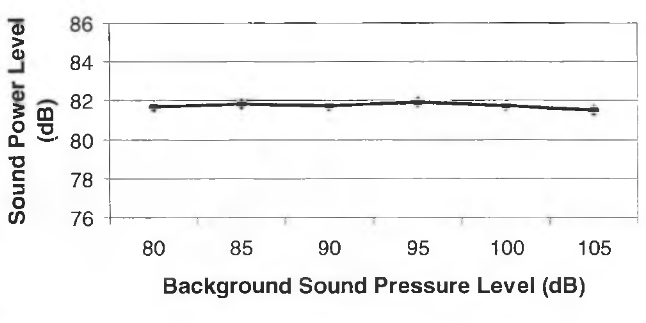
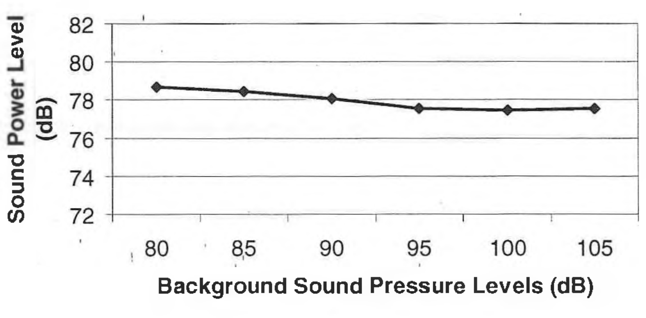


Figure 3: Reverberant Environment



On The Mechanical Behavior Of Vacuum Applied Surface Treatments

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1. INTRODUCTION

Surface treatments are sometimes used to reduce the amplitudes of vibration in large, shell-type structures (Nashif et al., 1985). The pads normally consist of a viscoelastic layer that may or may not be laminated with a thin constraining plate. When the pads are bonded onto the structure, additional damping is provided either through extensional (unconstrained) or shear dissipation (constrained pads) within the viscoelastic layer.

Viscoelastic damping pads are also used as a noise control element, owing to their ability for reducing transverse vibrations. However, in some applications such as aircraft skin assembly, adhesive bonded pads obviously may not be used. In these cases, a vacuum applied damping treatment can be used. It is bonded temporarily, and is removed once the skins are fastened, leaving no vestige on the aircraft.

Vacuum applied damping pads (VA pads) where shown to be quite efficient in reducing impact noises. However, when compared with traditional adhesive bonded pads, a major difference in the acoustic pressure time signal was observed, but was only partly explained (Ross, Amram, Ostiguy, 2001). In this paper, vibration measurements provide additional insight on the matter.

2. EXPERIMENTAL SETUP

Experiments were performed on a simply supported, rectangular aluminum plate. A steel, ball-ended hammer was used to strike the 0,9m x 0,6m x 4,8mm plate at its center point. VA pads were placed symmetrically at 25mm on either sides of the impact point.

One particular configuration was used as an extreme experimental case. In this configuration, the VA pads consisted of the single constraining layer. It was meant to show the effects of added mass and rigidity on the plate, without any viscoelastic damping. The two 1,5mm thick aluminum pads added an extra 31% to the thickness of the plate. A small degree of damping was possible through dry friction on the interface, as the pads were vacuum applied onto the plate.

3. NOISE MEASUREMENTS

During the impacts, the acoustic pressure was sampled at various points on a plane grid positioned 50mm from the surface of the plate. Figure 1 shows the data obtained at a measurement point located close to the impact axis (normal to the plate, passing through the impact point).

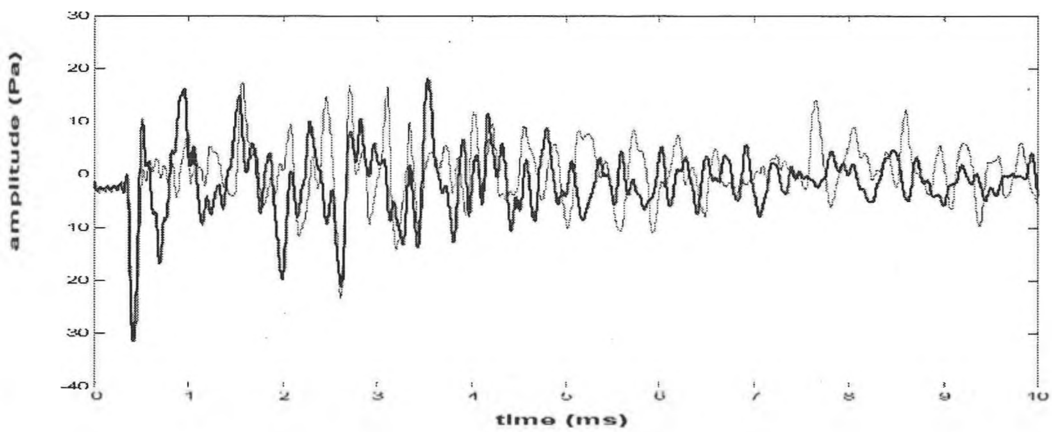

Fig. 1. Acoustic pressure along the impact axis (light: bare plate; dark: with VA pads).

The thin curve is the acoustic pressure due to the impacts on the bare plate (no pads). The acceleration peak is easily identified, and is followed by a "quiet" period before the ringing noise is established.

The thick curve is the acoustic pressure obtained when the VA pads were applied. These VA pads did not have a viscoelastic layer. The acceleration peak is similar to that of the bare plate. However, a large amplitude oscillation immediately follows this first peak and amplifies the radiated noise. This oscillation was previously thought to be the result of a local vibration of the plate between the two VA pads (Ross, Ostiguy, Amram, 2001).

4. VIBRATION MEASUREMENTS

Recent experimental work led to a better understanding of the phenomenon. Lightweight accelerometers were positioned at various locations on the system. Transverse vibrations of the plate were compared, with and without the VA pads. Figure 2 shows the transverse acceleration of the plate, measured at 50mm from the impact point. When the VA pads were used, the acceleration was measured both on the plate (back side) and on the pad (front side), at the same location as above.

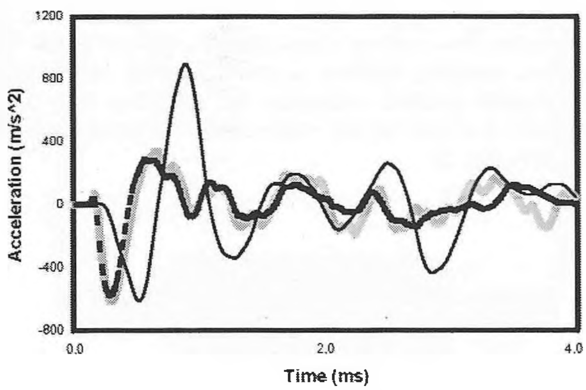

Fig. 2. Transverse acceleration of the plate at 50mm from the impact point (gray: bare plate; - - -: plate with VA pads; : VA pad).

It can be seen that for nearly 2ms, there are insignificant changes in the behavior of the plate, when the VA pads are applied. Changes that do occur seem to affect relatively high frequencies only.

The VA pad itself, however, does not seem to follow the movement of the plate. Its transverse acceleration signal shows that a very large oscillation exists at many but not all locations on the surface of the pad. It seems that the vacuum interface allows a certain amount of freedom to the VA pads, and some regions may temporarily separate from the plate (or may not be in contact at all). This may also hold true for the viscoelastic VA pads, with a lesser degree since these ones are much heavier and more rigid than the constraining layer alone.

The vibration of the VA pads — and not that of the plate — seems to be the cause of the amplified acoustic pressure in the near field of the impacted plate. The VA pads therefore act as a second noise source, in addition to the plate.

5. CONCLUSIONS

It was shown, through experimental analysis, that the transverse acceleration of vacuum applied pads is partly independent of the motion of the impacted plate. In the future, a finite element analysis of the system should yield a complete view and better understanding of the motion of the pads. The use of Nearfield Acoustical Holography on such data would confirm the effect of this vibration on the sound pressure field. Observations and analysis should be performed on viscoelastic VA pads, in order to detect any similar behavior.

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AUTHORS NOTE

The experimental work was conducted by A.R. and by J.-F.R. at the École Polytechnique de Montréal.

Finite Element Modeling of Acoustical Silencers

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1. Introduction

Due to the increasing level of public awareness for noise concerns, the use of acoustical silencers is becoming more prominent. Current methods for design and prediction of performance are only reasonably accurate for specific design cases, and are unable to handle the wide variety of geometrical, environmental, and material parameters available. A numerical method, which can handle all of the various design cases and parameters, and can be implemented along with an optimization scheme is desirable.

The purpose of this paper is to provide an overview of the methods involved in creating a numerical model used to characterize an acoustical silencer. Such areas as geometry, sound absorbing material, and environmental conditions are included in the numerical model, and transmission loss results for various design cases are shown and compared to measured values obtained in physical systems.

2. Theory

The Finite Element Method (FEM) is a numerical modeling technique that can be adapted for use with acoustical problems. Starting with the acoustic wave equation, the acoustic FEM is given as follows [1]:

Stiffness, Wave Number, Mass, Pressure, Density, Velocity

$$\left([K_m] + jk\left(R\Omega/\rho c\right)[M_m] - k^2 K_s \Omega [M_m]\right)\{p_m\} = (j\rho\omega + R)\{Q_m\} \quad (1)$$

Resistivity, Speed of Sound, Structural Factor, Porosity, Frequency

This formulation is analogous to a multi-degree of freedom vibration problem. In this case, the pressure vector is the desired solution. In order to model an acoustical silencer, the geometry is divided up into a meshed grid of acoustical elements (as illustrated in Figure 1).

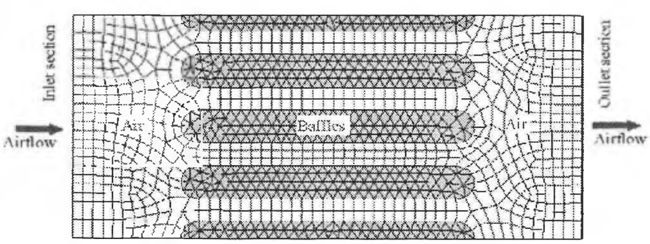

Figure 1. FEM Mesh of Parallel Baffle Silencer

Each element is interconnected with its immediate neighbor, forming a global matrix over the entire geometry.

The next step in the numerical model is to define the fluid and absorptive material properties. Such fluid properties as the speed of sound, the density, and the temperature can be altered. The absorptive material properties which can be included in the model are the flow resistivity, the porosity and the structural factor [2]. The values for these properties can either be known before hand to predict the response of a known system, or can be altered to aid in design and optimization.

The final step in the model is to apply the boundary conditions. In order to excite the model, the elements at the inlet section are given a unit particle velocity. The elements at the exit section are given the characteristic impedance ($z = \rho c$) to mimic an open section (preventing waves from reflecting once they have left the silencing element).

Once all of the elements have been formed and assembled, Eqn. (1) is solved for the pressure vector at each of the element nodes. The three pressure values of interest (p_1, p_2 and p_3) are interpolated from the resulting pressure vector and are then used in the 3-point method for calculating Transmission Loss [3] (TL, defined as the ratio of sound intensity incident to sound intensity transmitted). The 3-point method measures the sound at two points upstream from the silencer and one point downstream (as shown in Fig. 2).

Figure 2. Measurement Locations for 3-Point TL Formulation

The resulting equation can then be used to solve for the TL in the numerical silencer model.

$$TL = 20\log_{10}\left|\frac{p_1 - p_2 e^{jkx_{12}}}{p_3 - p_3 e^{j2kx_{12}}}\right| \quad (2)$$

3. Discussion of Results

Verification of the results was completed by comparing the numerical model results to those obtained using known formulations of simple reactive silencer systems, and more complex physical models of parallel baffle silencers.

One of the first verifications performed, involved the modeling of a simple expansion chamber silencer. A physical model was constructed and it's TL was measured. Also, based on the dimensions, its TL was predicted using the known formula for an expansion chamber [4]. Both of these results were compared to those obtained by a numerical model of the same dimensions. Figure 3 shows the dimensions of the silencer in question while Fig. 4 shows measured, calculated, and numerical results.

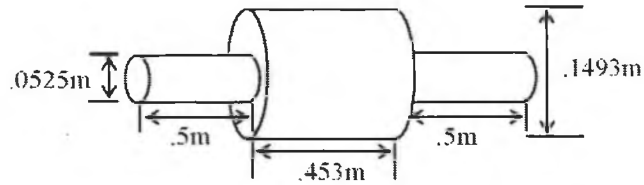

Figure 3. Expansion Chamber Silencer Dimensions

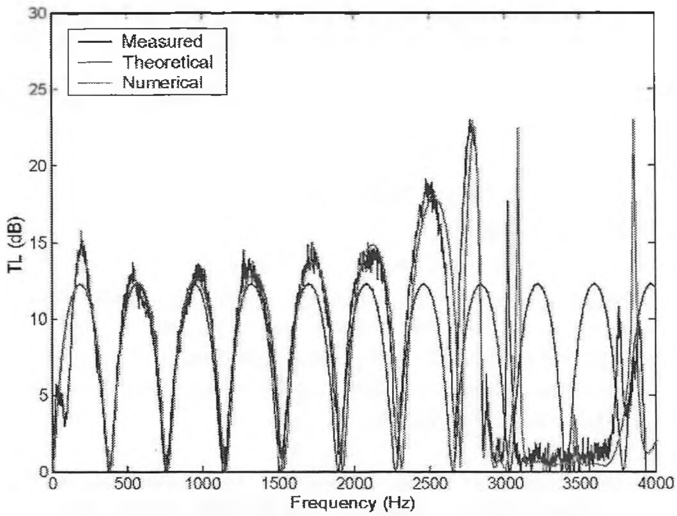

Figure 4. Expansion Chamber *TL* Results

Note that all three curves follow each other until the critical frequency at which the plane wave propagation assumption is no longer valid (approx. 2000Hz). After this point, the theoretical prediction is no longer valid, and can be ignored. The measured and numerical curves, however, follow each other very well over the entire frequency spectrum.

Another verification of the FEM involved the modeling of a scale parallel baffle silencer. The numerical results were compared to those measured from the scale model shown in Fig. 5 which consisted of a source end with a straightening section (for plane wave propagation), a test section with variable parallel baffle configurations and numerous microphone locations, and a termination section with an anechoic termination to prevent reflected waves from returning after the sound has left the test section.

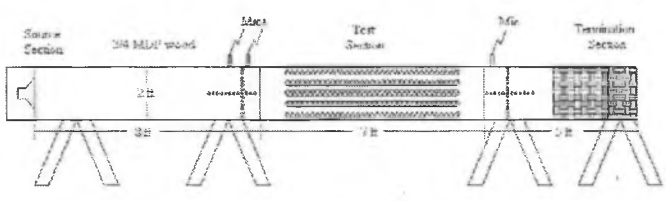

Figure 5. Physical Model of Acoustical Duct Silencer System.

The test section contained 3 baffles of 50mm thick Kaowool Ceramic Fiber with a flow resistivity of 106000mks rayls/m, a porosity of 0.799 and a structural factor of 2.0. Figure 6 shows the results obtained from the numerical and physical models, along with the difference between the two.

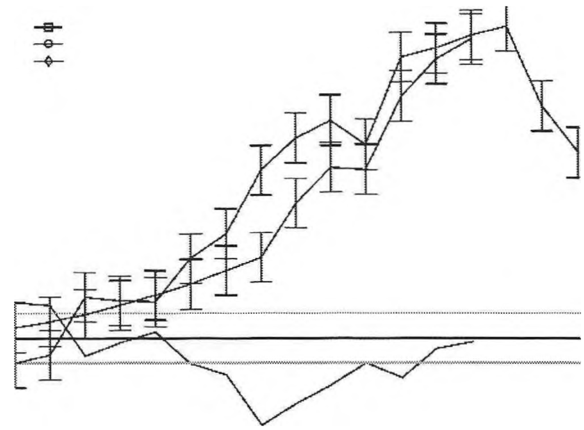

Figure 6. TL results for 3 baffles of Kaowool Ceramic Fiber

Note that for most of the frequency range tested, the difference between the two curves is within ±5dB and the error bars (based on statistical testing) almost always overlap

4. Summary and Conclusions

The results gathered thus far indicate excellent results for purely reactive acoustical silencers, and good results for absorptive silencers. Future work would include a more complicated model for sound absorbing material, and the inclusion of flow. Ultimately, external calculations will be required (either by Boundary Element Methods or Infinite Element Methods) along with vibro-acoustic coupling to calculate the Insertion Loss.

Acknowledgements

The authors would like to thank the Natural Sciences and Engineering Research Council of Canada (NSERC), the University of Alberta Department of Mechanical Engineering and ATCO Noise Management for the financial support for this work.

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A Method For The Inverse Characterization Of Poroelastic Mechanical Properties

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1. INTRODUCTION

Porous materials like polymer foams are widely used for sound absorption and damping vibrations in industries such as buildings construction or aeronautics.

Vibroacoustic efficiency and dissipation mechanisms of these materials depend on their dynamic properties, for assumed isotropic materials : Young's modulus E (or bending rigidity D) and the structural loss factor η. Poisson's ratio, ν, is found constant[Pritz,1994].

This work presents an experimental setup to characterize the frequency dependence of Young's modulus and structural loss factor of a poroelastic material bouded onto a aluminium plate in bending vibration by inversion with the help of an equivalent plate model. Development of such an equivalent plate is interesting to reduce numerical calculation time and memory requirement.

2. EXPERIMENTAL SETUP

A generic experimental configuration is used : the aluminium plate is simply supported and excited with a point mechanical excitation. The mean quadratic velocity of the plate treated by the material is measured over the frequency range of 20 - 820 Hz and is used for a two parameters (D, η) modal inversion.

ρ	ν	ϕ
8.85 (kg.m^{-3})	0.44	0.99

α	σ	Λ	Λ'
1.0	12600 (Nms^{-4})	78 (10^{-6} m)	192 (10^{-6} m)

Table 1. Properties of studied plastic foam.

The parameters of the studied material within the Biot theory[Allard,1993], measured from usual setup [Atalla, 2000], are reported in table 1.

In the following, subscript 1 will refer to the aluminium plate and 2 to the porous material.

3. DISSIPATION MECHANISMS IN POROUS MEDIA

In the low frequency range, among the three dissipation mechanisms occurring within the porous layer, thermal dissipation has been found negligible for various porous materials and thicknesses[Lemarinier, 1997]. Thus, in the chosen frequency range, only structural and viscous dissipations are significants.

In this frequency range, Dauchez's work[Dauchez, 1999] has shown viscous dissipation is mainly related to flow resistivity σ. For high resistivity materials, structural damping is the major dissipative mechanism, whereas for low resistivity material, viscous dissipation can be of greater influence.

Our approach is to include viscous dissipation by taking the effective density[Allard, 1993] $\tilde{\rho}_2$ in place of the material density ρ_2 and set the equivalent plate's loss factor to the material's one to account for structural damping.

4. EQUIVALENT PLATE MODEL

From previous observations, the porous layer can be considered as a monophasic viscoelastic with corrections for viscous effets and structural damping. In addition, the shear effect in the porous layer is taken into account.

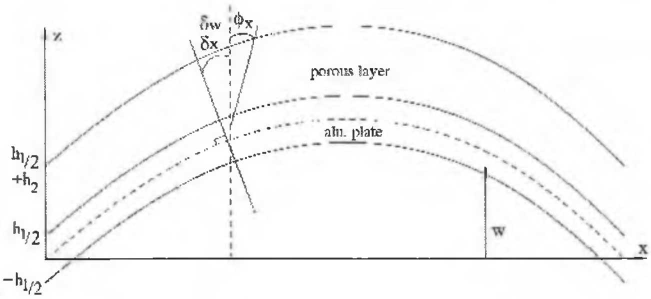

Fig. 1. Displacement components of the two layers plate of thickness h1 + h2.

Thus, displacement components for the porous layer are assumed to be of the form :

$$u_2(x,y,z,t)=-z\frac{\partial w}{\partial x}-(z-\frac{h_1}{2})\phi_x(x,y,t) \qquad (1)$$

$$v_2(x,y,z,t)=-z\frac{\partial w}{\partial y}-(z-\frac{h_1}{2})\phi_y(x,y,t) \qquad (2)$$

$$w(x,y,z,t)=w(x,y,t) \qquad (3)$$

Bending rigidity and surface density of the equivalent plate can be written :

$$D_{12}=(D_1+D_2)C_s(k_2) \qquad (4)$$

$$m_{12}=\rho_1 h_1+\tilde{\rho}_2 h_2 \qquad (5)$$

where $C_s(k^2)$ is a correction factor of the equivalent Ross-Kerwin-Ungar[Ross, 1959] bending rigidity accounting for the shear and depending on the wavenumber k.

Writing kinetic and strain energies and applying Lagrange's equations on the three variables w, ϕ_x and ϕ_y gives the two equations of motion :

$$D_{12}\Delta\Delta w-\omega^2\tilde{m}_{12}w+D_4\Delta\theta=0 \qquad (6)$$

$$D_4\Delta\Delta w+D_3\Delta\theta-\kappa^2 G_2 h_2\theta=0 \qquad (7)$$

where ω is the pulsation, θ is the variable $\delta\phi_x/\delta x + \delta\phi_y/\delta y$ and Δ is the Laplacian operator $\delta^2/\delta x^2 + \delta^2/\delta y^2$. G_2 is the shear modulus of the porous material, κ^2 its correction factor introduced by Mindlin[Mindlin, 1951]. Flexural rigidities D_3 and D_4 are :

$$D_3=\frac{E_2}{1-\upsilon_2^2}\int_{h_1/2}^{h_1/2+h_2}\left(z-\frac{h_1}{2}\right)^2 dz \qquad (8)$$

$$D_4=\frac{E_2}{1-\upsilon_2^2}\int_{h_1/2}^{h_1/2+h_2} z(z-\frac{h_1}{2})dz \qquad (9)$$

5. RESULTS AND PERSPECTIVES

Bending rigidity and loss factor of the foam are given in figure 2. This figure is a typical representation of polymers bending rigidity D and structural damping η in frequency[Corsaro, 1990]. Dimensions used in experiment are :

plate area : 0.48 x 0.42 m.
thicknesses : $h_1 = 3.175$ mm. and $h_2 = 76.2$ mm.

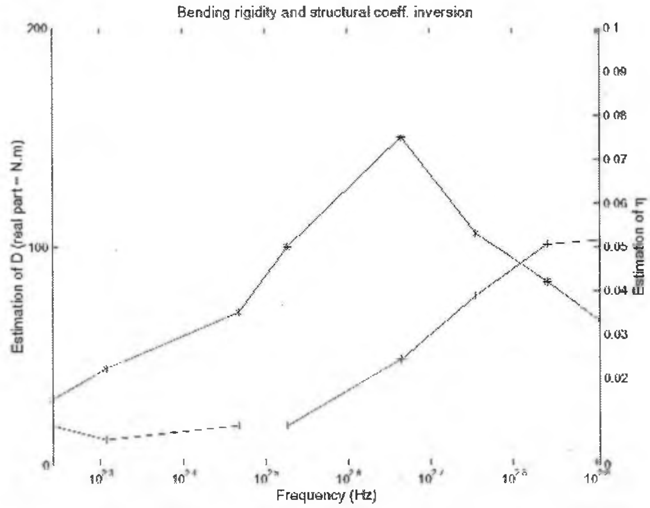

Fig 2. Bending rigidity D_2 (+) and struct. damping (*) frequency dependence of studied porous material.

Our next step is to compare these results with the temperature-frequency superposition principle[Corsaro, 1990].

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Modelling Of Machinery Vibration Isolation Systems

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1. INTRODUCTION

The VVES (Vibration of Viscoelastic and Elastic Structures) suite of MATLAB programs was developed to predict the 3D vibration and isolation characteristics of machinery mounted on one or more layers of elastomeric isolators [1, 2]. The layers can be separated by rigid rafts or flexible rafts formed from a grid of elastic beams. The properties of isolators can be computed within the program for simple flexible component shapes using the viscoelastic material properties. Alternatively, overall mount properties can be entered into the model. This is useful where isolation mounts have complicated geometry and/or are under significant static compression. Measured individual mount properties or detailed finite element predictions can then provide accurate input dynamic characteristics. The suite of codes has a large number of input files and limited visualization capabilities. A graphical user interface called VIMGEN (Vibration Isolation Model Generator) has been developed by DRDC Atlantic and Martec Limited [3] which allows system models to be generated quickly with fewer errors using wizards and interactive detailed 3D graphics. This paper describes the capabilities of the VVES and VIMGEN codes and compares predictions with small-scale experiments [4].

2. THE VVES MODEL

VVES is a 3D finite element modelling tool with a limited number of element types, specialized for efficient modelling of low frequency dynamic behaviour of vibration isolation systems. The program can calculate the vibration modes and natural frequencies of a system, and the steady state response to user defined forces applied to an engine "block". VVES can also calculate the "quasi-static" response of a system caused by motion of the machinery foundation, for example, on a ship in a seaway. The engine or machine to be isolated is assumed to be a rigid body. A centre of gravity, mass moments of inertia, and any number of user-defined attachment points define the engine "element". Elements that can be connected to the attachment points include vibration isolators, elastic beams and point masses. The system can also contain rigid raft elements defined in a manner similar to the engine element. Flexible rafts can be considered using a grid of elastic beam elements.

Three "flavours" of VVES have been developed, each employing a different approach to the modeling of the isolators: i) an approach based on finite element analysis of the viscoelastic isolators using constitutive data of the isolator material; ii) an approach based on the determination of the constitutive data from the results of dynamic stiffness measurements conducted on engine mounts; and iii) an approach based on the results of so called 4-pole parameter data that describes the velocity force transfer matrices measured on full sized mounts in a given direction. A fourth version of VVES has been developed for consideration of active vibration isolators. Only the first version will be discussed further in this paper.

At the global level, each vibration isolator is considered as a single finite element with two attachment nodes, each node having three displacement and three rotational degrees of freedom. The isolator element can have any orientation in space as defined by the attachment nodes and a third point providing the orientation about a line through the attachment nodes. The 12x12 dynamic stiffness matrix $\mathbf{Z}_v$ for the isolator element is formed according to Equation 1 as a function of frequency ω,

$$\mathbf{Z}_v(\omega) = G_1(\omega)\mathbf{K}_{v1} + G_2(\omega)\mathbf{K}_{v2} - \omega^2 \rho \mathbf{M}_v, \quad (1)$$

where $\mathbf{K}_{v1}$ and $\mathbf{K}_{v2}$ are element stiffness matrices and $\mathbf{M}_v$ is the element mass matrix. $G_1(\omega)$ and $G_2(\omega)$ are respectively the complex shear and complex bulk moduli of the elastomeric material. The elastomeric materials typically used in vibration isolators are nearly incompressible and the above formulation eliminates numerical problems that can occur in formulations using Young's modulus and a Poisson's ratio $\mathbf{v}$ that approaches 0.5. The material moduli $G_1(\omega)$ and $G_2(\omega)$ can be measured using dynamic-mechanical testing of small samples of the isolator material. $\mathbf{K}_{v1}$, $\mathbf{K}_{v2}$ and $\mathbf{M}_v$ are only dependent on the isolator geometry and are calculated within VVES for elements with the geometry shown in Figure 1. For isolators of more complicate shaped, if it is assumed that $\mathbf{v}$ is independent of frequency, then $\mathbf{K}_{v2} = 0$ and $\mathbf{K}_{v1}$ can be derived from an external "standard" elastic finite element analysis and imported into VVES.

Fig. 1 Isolator shapes modelled within VVES.

3. VIMGEN MODEL GENERATOR

VVES provides simple line drawings of the engine and rafts. Isolators and beams are drawn as lines between nodes. This drawing is only produced after creation of all ASCII input files defining the VVES model geometry. This is not very useful for checking purposes. VIMGEN has a user-friendly Windows-based interface that allows model data to be entered easily with forms and wizards. As geometric information is entered, the updated model can be viewed in realistic 3D shaded graphics with mouse controlled movement and interaction with model components. VIMGEN can generate input files, run the VVES code and provide 3D animated graphics of system vibration modes and response. It can also provide graphs and tables of results. VIMGEN was developed using a suite of C++ tools call HOOD (Hierarchical Object Oriented Developers tool kit) developed at DRDC to allow rapid generation of engineering modeling and visualization software.

4. SMALL-SCALE EXPERIMENT

As part of testing of VVES and VIMGEN, predictions for some small-scale isolation systems have been compared with measurements made by the Defence Science and Technology Organisation (DSTO), part of Australia's Department of Defence [4]. These systems consisted of 25 mm x 50 mm x 24 mm rubber isolators sandwiched between steel plates. The systems were suspended axially from a soft string in a free-free state. Two VIMGEN/VVES models are shown in Figure 2. The first (A) consists of a single isolator between two 60 mm x 60 mm x 24 mm steel plates. The second (B) has 4 isolators between two 120 mm x 120 mm x 24 mm steel plates. A single "very soft" isolator (shown as a vertical line element) connects the lower steel plate to a rigid base foundation, achieving essentially a "free-free" system. A third two-stage model similar to model "B" but with an additional steel plate and a second layer of four isolators has been built but not tested to date.

The frequency dependent dynamic-mechanical properties of the rubber material were measured at the DRDC Atlantic Dockyard Lab and used in the VVES models. The Young's modulus of the material varied from 3.4 MPa at 1 Hz to 3.9 MPa at 300 Hz. The loss factor of the material varied between 0.03 at 1 Hz to 0.05 at 300 Hz. A comparison of measured and predicted natural frequencies for models "A" and "B" is given in Table 1. The lateral direction refers to the long dimension of the isolators. Figure 3 shows some of the modes shapes predicted by VVES for Model A and viewed with VIMGEN.

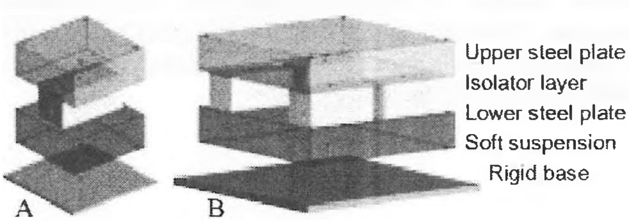

Fig. 2 VIMGEN models of vibration isolator systems.

Table 1. Free-free system natural frequencies

Mode Shape	Model	Pred. (Hz)	Meas. (Hz)	Diff. (%)
Axial rotation:	A	42.0	38.9	5.6
(about the vertical)	B	68.8	69.0	-0.3
Lateral translation:	A	106.9	100.0	6.9
(shear)	B	77.1	76.9	0.3
Transverse translation:	A	96.2	92.7	3.8
(shear)	B	69.1	72.9	-5.2
Lateral rotation:	A	49.0	48.6	0.8
(tension/compression)	B	165.6	157.2	5.3
Transverse rotation	A	100.2	NA*	
(tension/compression)	B	126.8	114.9	10.4
Axial translation	A	139.7	134.9	3.6
(vertical direction)	B	136.4	NA	

*Measurements show "diagonal" rotational modes at 93.4 Hz and 100.4 Hz

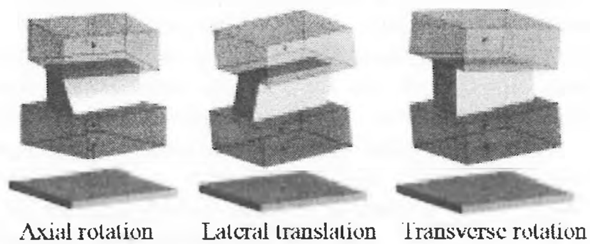

Fig. 3 VIMGEN views of VVES predicted mode shapes.

5. CONCLUSION

Computer codes VIMGEN and VVES have been developed for modelling the 3D characteristics of vibration isolation systems containing elastomeric isolator elements. The codes have been described and a study of the modes of a small-scale system provided. Further work is underway including predictions and measurement of vibration transfer functions for both small-scale systems and for actual diesel engine isolation systems.

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NUMERICAL CHARACTERISATION OF NONLINEAR STIFFNESS PROPERTIES OF PRE-STRESSED VIBRATION ISOLATION MOUNTS

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1. INTRODUCTION

The VVES (Vibration of Viscoelastic and Elastic Structures) suite of programs were developed using a simplified vibration isolation model, in which the isolators were idealised by 2-noded flexible elements, which were connected to the upper and lower attachment points. Under the assumption of constant Poisson's ratio, the dynamic stiffness of each isolator can be represented by the product of complex frequency-dependent Young's modulus and a normalised 12×12 condensed stiffness matrix, which can be obtained through finite element analyses. The VVES program provides a linear finite element capability for this purpose. However, it can only handle vibration isolation mounts with relatively simple geometries under small deformation [1]. In other words, for practical mounts with complex geometries and/or subjected to finite static deformations, the condensed stiffness matrices must be pre-generated using more sophisticated external computational tools and imported to the VVES program. The purpose of the present study is to develop a numerical procedure for computing the condensed stiffness matrices for arbitrary vibration isolation mounts through nonlinear finite element analyses using the VAST program [2]. In this procedure, nonlinear finite element analysis is first performed up to the desired pre-stress level and the resulting tangent stiffness matrix is then used to compute the condensed matrix using a simple method based on prescribed unit displacements and rotations. The present procedure has been utilised to characterise three practical vibration isolation mounts, and for all of these mounts, good numerical results have been obtained.

2. CONDENSED STIFFNESS MATRIX

A typical vibration isolation mount contains top and bottom metal plating and a flexible visco-elastic core. All these structural components need to be included in the finite element analysis. In order to formulate the condensed stiffness matrix with respect to the attachment points, two additional nodes are introduced in the finite element model and multi-point constraint equations are utilised to enforce the displacement compatibility requirements between the attachment points and the surfaces of the top and bottom plates. Once the global stiffness matrix is computed, the condensed stiffness matrix can be obtained by eliminating all the degrees-of-freedom from the system except those associated with the attachment points.

The elimination of these internal degrees-of-freedom can be achieved by using the method of static condensation and the condensed mass matrix can also be obtained by the method of Guyan reduction. However, this procedure involves a large number of matrix operations, which makes it less efficient computationally. For this reason, a simple method, which requires repeated solutions of the global finite element system subjected to prescribed unit displacements and rotations at the attachment points, has been utilised in the present work [3].

3. NONLINEAR FE FORMULATION

In a practical engine vibration isolation system, the isolators are normally subjected to static axial compression caused by the self-weight of the machinery before the application of the small amplitude dynamic load. This static compressive load is often large enough to produce finite deformation of the flexible mount. In order to include the effect of this initial finite deformation in the condensed stiffness matrix of isolators, the global tangent stiffness matrix at the full pre-loading must be employed in the calculation of the condensed stiffness matrix.

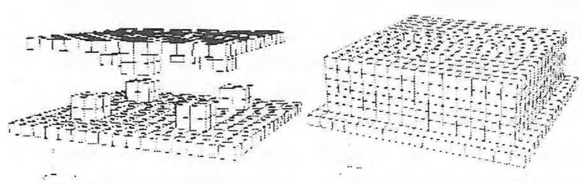
The computation of the tangent stiffness matrix requires nonlinear finite element analyses, which are performed using the general-purpose nonlinear finite element program, VAST. The VAST program provides a number of large strain, hyper-elastic solid elements, developed using the consistent co-rotational and total Lagrangian formulations [4]. A mixed method based on independent interpolation of pressure and displacement fields has been implemented to treat material incompressibility, and incompatible bubble modes have been employed to enhance behaviour of the lower-order element for bending dominated deformations. The hyper-elastic material property was represented using a revised version of Ogden's three-term strain energy function. This strain energy function is formulated in terms of principal stretches and includes 6 material constants in addition to the bulk modulus.

The large strain hyper-elastic capability in VAST has been extensively verified using a large number of test cases, including inflation of rubber cylinder and rectangular sheet and constrained compression of rubber blocks. For all these test examples, VAST results were compared with published analytical and numerical solutions. Close agreement has been obtained.

4. APPLICATIONS

The VAST finite element program has been applied to predict condensed stiffness properties of three practical vibration isolation mounts, including the NETE mount, the PDE engine mount and the PDE raft mount [3]. The nonlinear behaviour associated with the pre-loading was modelled using the large strain hyper-elastic capability in VAST. All these mounts have complicated geometric shapes and are composed of top/bottom steel plating and a soft rubber core. Due to page limitation, only the results for the NETE mount are presented here.

The basic shape of the NETE mount is a nearly rectangular block. Each of the top and bottom steel plates contains four insertions into the rubber core material. All these insertions are of the same size and shape, which is a cylinder with a hemi-spherical top. In the present finite element analysis, a relatively coarse mesh has been utilised, in which the cylindrical insertions were approximated by rectangular ones, as shown in Figure 1. It is believed that this mesh is sufficient for characterising the overall stiffness property of the mount structure.



Top and Bottom Plates Visco-Elastic Core

Fig. 1 Finite element model for nonlinear analysis of NETE mount.

After the construction of the finite element mesh, nonlinear material properties must be determined before finite element analysis. Because the top and bottom plates in this mount were both made of steel, the standard steel properties, $E=2.07\times10^5$ MPa and $\nu=0.3$, were utilised. The selection of material constants for the hyper-elastic material was more difficult. In the present study, the six material constants required by the Ogden's model were first identified by curve-fitting the experimental data for uni-axial tension, and then scaled by matching the predicted and measured initial stiffness of the NETE mount for axial compression. Using these material properties, the VAST program predicted a nonlinear response, which is in close agreement with the experimental data for the complete loading range, as shown in Figure 2.

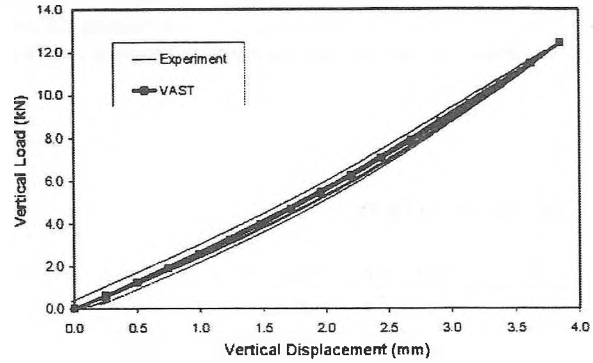


Fig. 2 Measured and predicted load-displacement curves.

After convergence was reached at each load step, the current tangent stiffness matrix was used for the computation of the 12×12 condensed stiffness matrix. In order to investigate the variation of each element in the condensed stiffness matrix with the increase of pre-load, we obtained condensed matrices at a number of pre-load values. This investigation revealed that different elements in the condensed stiffness matrix vary with the magnitude of pre-load in very different ways. This is probably a consequence of the material anisotropy associated with the non-uniform stress distribution in the rubber component. As a result, the earlier method for taking into account the pre-load effect, which involves uniform scaling of the initial condensed stiffness matrix, is inaccurate and may results in errors in the dynamic simulation using the VVES programs.

5. CONCLUSION

A computational procedure has been developed to evaluate the condensed stiffness matrix of vibration isolators required by the VVES program. This procedure is based on a nonlinear finite element program, VAST, which is able to handle complicated mount geometry and nonlinear effect associated with the static pre-loading. Three practical vibration isolation mount structures have been analysed and satisfactory numerical results have been obtained.

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Vibro-acoustic behaviour of multi-layer orthotropic panels

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1. INTRODUCTION

Multi-layer composite panels such as sandwich and honeycomb are widely used in the aerospace and aeronautic industries. A good understanding of their behaviour is necessary to predict their dynamic behaviour under acoustic excitation. A general eigenvalue approach based on wave theory is presented to compute the dispersion curves of such panels. Using these curves, the radiation efficiency, the modal density and the group velocity are computed and used within SEA framework to predict the transmission loss of these panels. Numerical results are presented to validate the proposed approach.

2. NUMERICAL MODEL

The literature [4], [5] proposes various methods on the dynamic behaviour of multi-layer composite structures. It is shown [5] that the analytical resolution of the general problem is not possible unless simplifying assumptions are made which narrows the field of applicability of the suggested models. In order to circumvent these disadvantages, a general numerical method is used to establish and solve the dispersion relation of these structures.

Using a general Mindlin model [1], [2] with transverse shear and the strain - deformations relations, the governing equations of the composites are written in terms of a general hybrid vector made up of the generalized efforts and displacements of the panel :

$$\langle e\rangle = \langle w, \psi_x, \psi_y, M_x, M_y, M_{xy}, Q_x, Q_y\rangle; \qquad (1)$$

with, w the displacement along z axis; ψ_x , ψ_y the rotations around y and x axis respectively; M_x, M_y, M_{xy} the bending moments and Q_x and Q_y the shear efforts. In this context, the system of equilibrium equations efforts - displacements of the panel can be written in the form:

$$[A_0]\langle e\rangle = [A_1]\left\langle\frac{\partial e}{\partial x}\right\rangle + [A_2]\left\langle\frac{\partial e}{\partial y}\right\rangle; \qquad (2)$$

where: $[A_0]$, $[A_1]$, $[A_2]$ are matrices of dimension 8×8 containing the coefficients of the equilibrium efforts - displacements equations system of the panel.

The solution of the system is written in the form:

$$\langle e\rangle = \{e\}\, e^{j\omega t \; jk_x x \; jk_y y}; \qquad (3)$$

with, k_x and k_y the components of the structural wave number k_p of the panel and ω the pulsation of vibration. The real values of k_p correspond to the propagating waves while the complex values to the evanescent waves. In this study our interest is limited to the propagating waves.

Relation (2) becomes following the use of relation (3) and algebraic simplifications:

$$[A_0]\{e\} = k_p\,[B]\{e\}; \qquad (4)$$

with:

$$[B] = -j(\cos\varphi\,[A_1] + \sin\varphi\,[A_2]); \qquad (5)$$

where, one used:

$$k_x = k_p\cos\varphi; \quad k_y = k_p\sin\varphi \qquad (6)$$

φ being the direction of wave propagation in the panel.

The resolution of the system (4) leads to a generalized eigenvalues problem, where k_p represents the eigenvalues vector and $\{e\}$ the matrix of the corresponding eigenvectors (columns). The eigenvalues vector contains the bending and evanescent wave numbers (conjugate complex solutions). The bending wave number of the panel corresponds to the mathematically real eigenvalue of the problem (with an infinitely small imaginary component).

3. GROUP VELOCITY AND MODAL DENSITY

In a structure, the waves having the same phase velocity, transport the energy of vibrations with a velocity called group velocity. It is expressed in the following general way :

$$c_g(\omega,\varphi) = \frac{d\omega}{dk}. \qquad (7)$$

Numerically, in a first approximation, the group velocity is obtained for a given heading φ using a finite difference scheme as the ratio between the infinitesimal variation of the excitation frequency and the corresponding variation of the structural wave number.

Next, the modal density is obtained by integrating its angular distribution:

$$n(f) = \int_0^{\pi/2} \frac{2L_x L_y}{\pi} \frac{k_p}{|c_g|} d\varphi \; ; \qquad (8)$$

with, L_x and L_y the side dimensions of the panel and f the frequency of excitation.

4. RADIATION EFFICIENCY

The radiation efficiency of the panel $\sigma^*(f)$ is computed using Leppington model [3] with the structural wave number, obtained by the resolution of the generalized eigenvalues problem. In order to take into account the total vibration energy of the resonant modes in a frequency band, the radiation efficiency is averaged in the wave number space by the angular distribution of the modal density $n^*(f,\varphi)$. One makes here the assumption of energy equipartition between the resonant modes in a frequency band. The radiation efficiency of the multi-layer composite panel is written as:

$$\sigma(f) = \frac{\int_{k_{min}}^{k_{max}} \int_0^{\pi/2} \sigma^*(f) n^*(f,\varphi) k dk d\varphi}{\int_{k_{min}}^{k_{max}} \int_0^{\pi/2} n^*(f,\varphi) k dk d\varphi} \; ; \qquad (9)$$

where, k_{min} and k_{max} are the wave number field boundary corresponding to the studied frequency band.

5. VALIDATION OF THE MODEL

Like first validation of the model, one considers a thick orthotropic panel (one layer) having the same physical properties in the three directions (isotropic thick panel). The results of the numerical model for the composite panels are compared with the results of the analytical model for isotropic thick panels. The case of an aluminium panel ($E=7.2*10^{10}$ Pa; $\rho=2780$ kg/m^3; $\nu=0.33$; $\eta=0.007$) of thickness h=10cm and side dimensions L_x = 4.6 m and L_y = 2.3 m is considered to validate the new approach. The modal density of the panel is calculated (figure 1) by the analytical model (—o—) and compared with numerical results (—*—). One observe that at low frequencies the effect of bending dominates while at high frequencies shearing become significant. The asymptotic tendencies in bending (—•—) and shearing (— —) are also shown for the same panel.

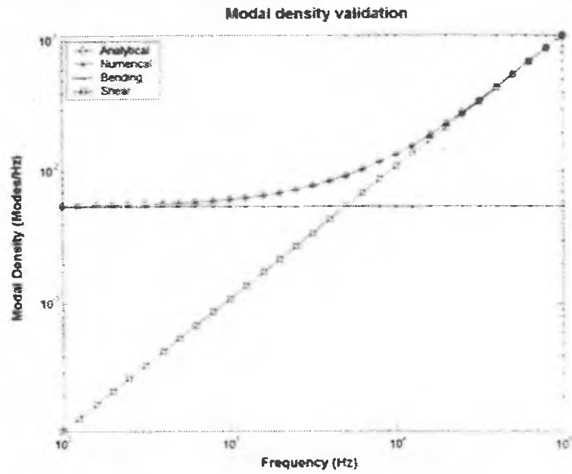

Figure 1. Modal density of the panel

ACKNOWLEDGEMENTS

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COMPARISON OF EXPERIMENTAL AND MODELED INSERTION LOSS OF A COMPLEX MULTI-CHAMBER MUFFLER

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1. INTRODUCTION

Given the greater legislative emphasis on the reduction of automotive noise emissions, the design of attenuators has become a paramount issue in the area of car development. Engine developers have been able to improve engine performance through lowering inlet and outlet valve resistance but at the cost of greater amplitudes of noise propagation downstream of the exhaust valves. As a result, exhaust system manufactures must design their products to achieve greater attenuation levels while not increasing flow resistance which would impinge on engine performance.

The complex multi-chamber muffler is the most common noise control filter used in automotive exhaust applications. Historically, the acoustical design of these mufflers has involved the construction of a prototype based on the initial application of the fundamental equations which would then be experimentally evaluated for performance. This trail and error process is a very costly and time consuming exercise which does not meet the development goals of today's highly competitive automotive market. As a result, the development of powerful computer based design systems for acoustical modeling have given way to the prototyping method of design primarily due to their ability to reliably predict automotive noise. Care, however, must be taken in the utilization of these sophisticated software packages to ensure that any input criteria is applied correctly to ensure meaningful results.

This study investigates the effect of using Ricardo WAVE, a computational simulation using a one-dimensional finite-difference formulation, to determine the realized insertion loss for an 'off the shelf' muffler. The theoretical results are compared to experimental measurements of the same muffler design. The purpose of this study is to investigate the effectiveness of using the theoretical model as a design tool for automotive muffler applications.

2. THEORY

Beranek [1] defined the muffler as any section of pipe that has been shaped in order to reduce the transmission of sound, while at the same time allowing the free flow of gas. The actual muffler used in the experimental portion of this investigation is a commercial reactive muffler. Figure 1 is a cutaway of the muffler showing the multiple chambers. In a reactive muffler, the multiple pipes and chambers provide an impedance mismatch for the acoustic energy traveling through it. "This impedance mismatch results in a reflection of part of the acoustic energy back toward the source of the sound" which prevents some of the energy from transmitting past the muffler.

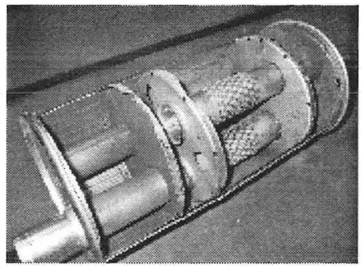

Figure 1: Cutaway of Experimental Muffler

The muffler used in the theoretical portion of this study is a complex multi-chamber muffler. The schematics of this muffler are as shown in figure 2. Time and resources available when the study was conducted did not permit an exact modeling of the muffler used in the experiments, however, the model used is similar in design and still reflects the purport of the study.

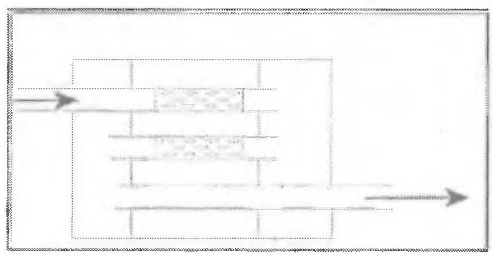

Figure 2: Schematic of Theoretical Muffler

Insertion Loss was used to compare the experimental results to the theoretical model. Insertion loss is the difference, in decibels (dB), between two sound pressure levels measured at the same location before and after the muffler is inserted between the measurement location and the source.

3. PROCEDURE

To experimentally determine the insertion loss of the muffler, a speaker was espoused to one end of a 10 foot exhaust pipe which was inserted through the wall of a semi-anechoic chamber. The other end of the pipe was attached to the muffler inside the anechoic environment which had a 2 foot pipe attached to its output. The speaker played a white noise signal into the system and a microphone located 0.1 metres from the outlet measured and recorded spectral results. Figure 3 shows the muffler in the anechoic room. The procedure was again repeated only with a straight pipe of the same length of the muffler, in its place.

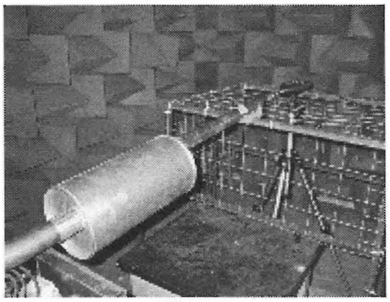

Figure 3: Experimental Muffler in Semi-Anechoic Room

To model the insertion loss, a computer model of the exhaust system complete with white noise source, was created
with WAVE which outputted spectral results, also downstream of the muffler location. The model was run with both the muffler in place as well as the case with it replaced with a straight pipe.

4. DISCUSSION OF RESULTS

Examination of figure 4 shows the spectral results of experimental exhaust system with both the muffler inserted and without. It is clearly shown that the sound pressure level (SPL) without the muffler is greater than the SPL with the muffler. Addition of the spectral information gives a total SPL of 97.4 dB without the muffler and 82.2 dB with the muffler. This gives an insertion loss of about 15 dB.

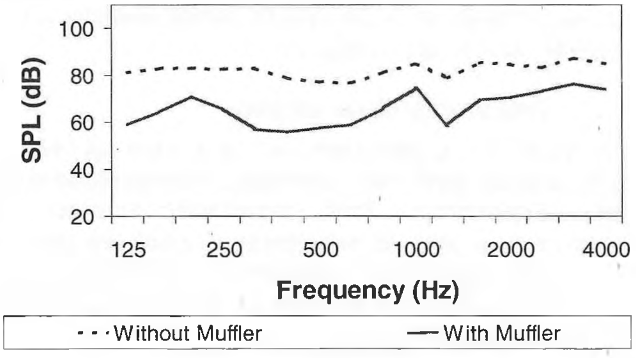

Figure 4: Experimental Spectral Analysis of Multi-Chambered Muffler

Figure 5 shows the results of the theoretical model. Again, it is clear that the muffler improved the acoustical performance of the system. Here, the overall SPL of the exhaust system without the muffler is 97.4 dB which is the same value determined in the experimental exercise. The SPL with the muffler in place is 79.1 dB giving the muffler a realized insertion loss of about 18 dB.

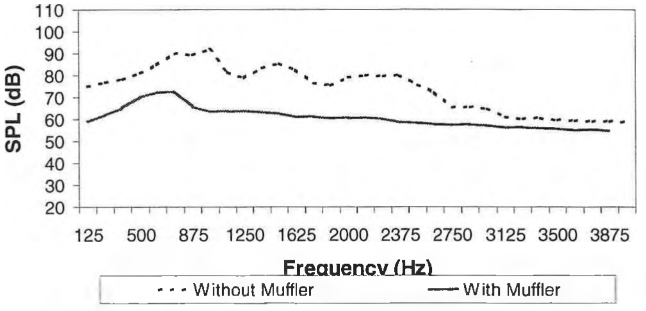

Figure 5: Theoretical Spectral Analysis of Multi-Chambered Muffler

The two studies show similar results with a difference of insertion loss of about 3 dB. A point of validation that can be made is that both studies resulted in identical overall SPL's without the muffler inserted. One should be aware that the modeled muffler was not a perfect match to the actual muffler, but none the less, the similarities still exist giving warrant to pursue the investigation further.

5. CONCLUSIONS

The purpose of this study was to investigate the effectiveness of using a theoretical model as a design tool for automotive muffler applications. Theoretical insertion loss of the computer model compared favorably to those determined from the experimental exercise. While good agreement was obtained in this investigation, some fundamental simplifying assumptions were made which incontrovertibly influenced the theoretical results thus encouraging further investigation.

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A fully hierarchical finite elements code for solving coupled elasto-poro-acoustic problems

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1. INTRODUCTION

The design of multi-layer structures is of significant importance in several industries including automotive, aerospace and buildings. A typical application concerns a foam layer sandwiched between an elastic structure and a limp impervious layer classically known as a septum and coupled to an acoustic cavity. At low frequencies, where the modal behavior of the system is important, finite elements are normally used. Though accurate, the use of linear elements in dynamic problems leads to an important number of unknowns and thus requires important computational resources. This paper presents the performance of hierarchical elements compared to linear elements for dynamic problems. For the sake of simplicity, the performance is demonstrated for a plate backed by a rigid cavity.

2. THEORY

2.1 Hierarchical elements

The p-extension of the finite elements method adds high order polynomial shape functions for the interpolation of the field of variables on the reference elements. In addition to the *node modes* used by linear elements, hierarchical elements are built with *side modes, face modes* and also *internal modes* in the case of tri-dimensional elements. Some are shown in figure 1.

Because the shape functions themselves resemble the vibration modes of the structure, a quick convergence is obtained for the dynamic behavior [1].

2.2 Coupling between different media

At a specific frequency, two different media generally do not have the same wavelength. Consequently, the level of discretization required is not the same. With linear elements, one of the two domains is over-distcretized if incompatible meshes are not used.

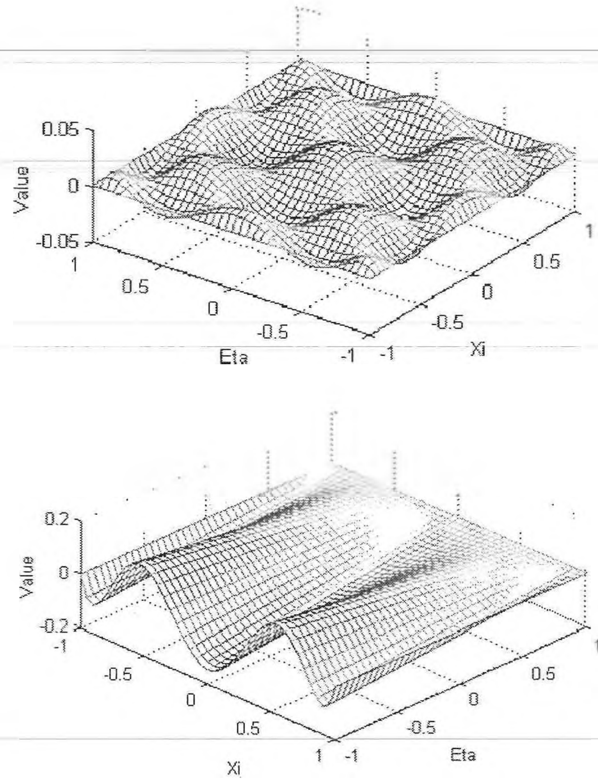

Figure 1 : Side mode and face mode of order 6 on a reference quadrilateral elements.

This problem is avoided by the used of hierarchical elements. The order of approximation is not necessarily the same for the two elements in contact, or for the two phases of one elements as in the case of porous elements arising from the mixed (u,p) formulation.

2.3 Transverse shear locking

The shear locking phenomena is a well know problem for some elements built with variationnal formulation involving only displacement. With hierarchical elements, the approximation space is rich enough to avoid this problem. This does considerably facilitates the developpement of plate, solid and porous elements having both the solid and the fluid phase [2].

3. RESULTS

The case considered here is a plate backed by a rigid cavity. The dimensions of the plate are 0.42m x 0.48 m. The thickness of the plate is 3 mm and the thickness of the cavity is 0.35 m. The properties for the aluminium plate are, E (N/m^2) =70x10^9, ν =0.33, ρ (kg/m^3) = 2700, η=0.015 and the properties of the air are c_0(m/s) = 342.2, ρ_0 (kg/m^3) = 1.213 and η=0.01.

The excitation is a point force located at a quarter of the plate in each direction.

The performance of the hierarchical code are compared to those of the commercial code MNS/Nova®. The mesh of the plate in Nova® is 20 x24 elements and the cavity contains 7 elements in the thickness. The computation time for the hierarchical code was 10:15 min while Nova® took 59:15 min. The order of appromixation per unit length in the hierarchical code is 33 for the plate and 12.5 for the cavity. The hierarchic shape function set used is exactly same as in MSC/Probe®.

Figure 2 compares the results of the two code and also shows that the hierarchical code has converged while Nova® has not yet converged for the plate. Making Nova® converge is a hard task because the number of degress of freedom grows as the square of the number of elements on each side of the plate.

4. DISCUSSION

Here we considered only the simple case of a plate backed by a cavity. Similar performance of the hierarchical elements have been obtained for other structure comprising porous media.

The hierarchical code developed will replace the low frequency module of MNS/Nova®.

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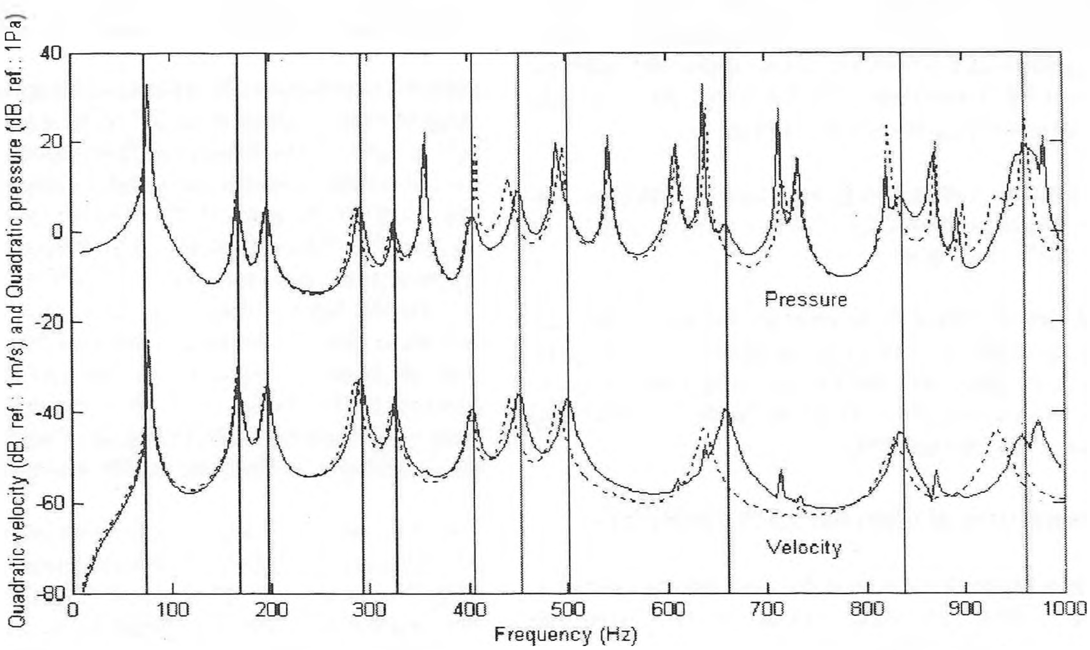

Figure 2 : Comparison of the results given by the hierarchical code to those of MNS/Nova®. The dotted curve is the one of Nova® and the plain curve are the results of the hierarchical code. The vertical lines are the analytical results for the vibration modes of the plate.

CURRENT LOW FREQUENCY NOISE (LFN) ASSESSMENT GUIDELINES AND THEIR USE IN ENVIRONMENTAL NOISE IMPACT ASSESSMENT

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1. The Nature of Low Frequency Noise

Discussed herein are current low frequency noise (LFN) noise assessment rating schemes, and key issues.

LFN is not clearly defined, but generally covers noise in frequencies below 100 to 150 Hz. Infrasound (i.e., sub 20 Hz) is not usually audible but may still produce impacts through perceptibility. Infrasound LFN can produce resonances in human organs and tissues. (Berglund et al, 1996). One feels the noise as pressure sensations (DEFRA, 2001). LFN can also rattle windows, dishes, etc. through sympathetic resonances, increasing annoyance (Bergland). Thus, LFN rating schemes typically go as low as 10 to 16 Hz.
Typical rural sound environments have few man-made noise sources (e.g., traffic and industrial noise) and generally have "flat" frequency spectra. In urban environments, significant levels of ambient low frequency noise exist but are generally less perceptible than in remote areas, due to masking by higher frequency noise within the "urban hum". LFN may be acceptable outdoors, particularly in urban environs. Indoors, building envelopes readily transmit LFN while higher frequency noises are blocked. This removes the masking effect of the high frequency noise, and can therefore increase the noticeability and related annoyance associated with the LFN portion of the spectrum. Closing the windows to block LFN noise only makes the problem worse.

LFN annoyance research suggests that for humans the doubling rate of perceived loudness is 4 to 5 dB for LFN vs 10 dB at 1KHz.(Bergland).

Balanced noise spectra at the receiver are needed to reduce the likelihood of annoyance and LFN complaints. Overall linear SPL and A-weighted SPL differences should be limited to 20 dB for low indoor A-weighted levels (Broner and Levanthall, 1983; Broner, 1994).

2. Comparison of Current LFN Guidelines

Table 1 and Figure 1 summarize the most widely used LFN guidelines. Most are noise criteria curves given the importance of spectral balance.

Table 1: LFN Impact Assessment Criteria

Criteria	Reference	Spectrum	Assessed Location
Overall Sound Level	ANSI B133.8	75 dBC Overall Limit	Outdoor
Vibration in light-weight structures	ANSI S12.2	1/1-Octave for 16, 31.5, 63 Hz bands	Indoor / Outdoor
NCB "balanced" noise criteria curves	ANSI S12.2 (Beranek)	1/1-Octave	Indoor
RC room criteria curves	ASHRAE 1995 and ANSI S12.2	1/1-Octave	Indoor
RC Mark II room criteria curves	ASHRAE 1999 Blazier	1/1-Octave	Indoor
RNC room criteria curves	Schomer	1/1-Octave	Indoor
LFNR low frequency noise rating curves	Broner and Leventhall	1/3-Octave	Indoor
LFRC low frequency room criteria curves	ASHRAE (Broner)	1/3-Octave	Indoor

Figure 1 shows huge discrepancies among the curves with a range of over 30 dB between RC Mark II and NCB at 16 Hz. Who's right? Both Broner and Beranek have compared the various ranking schemes, and found the other party's deficient (Broner, 1994, Beranek, 1997). Broner's and Blazier's curves are based on laboratory tests while Beranek's are based on reported annoyance with real-world HVAC systems. Note that the RC Mark II curves lie well below the threshold of hearing in the 16 Hz band. Beranek has argued against limiting noise to levels below the threshold of hearing (Broner, 1994). Broner and others have responded that LFN impacts at levels below the threshold of hearing are possible, due to sensation of the noise and the feeling of envelopment.

The RNC curves (Schomer, 2000) attempt to bridge the gap between the two systems, by providing base curves similar to the NCB system, for "well-behaved" HVAC systems, and then applying temporal variation penalties for annoyance due to large turbulent fluctuations at low frequencies.

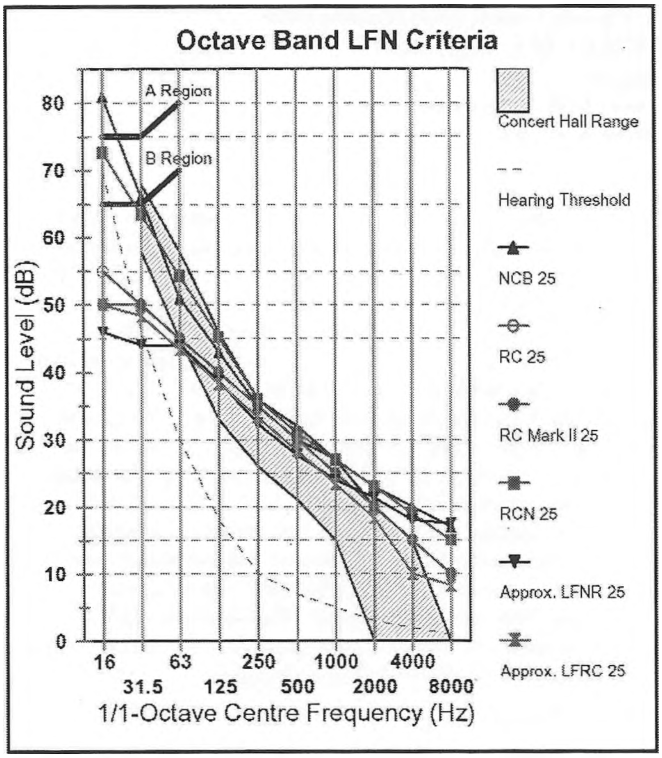

Figure 1: LFN Criteria Curves Compared

Note: LFNR and LFRC curves approximated by assuming that the lowest 1/3-octave value in each 1/1-octave band is representative of the equivalent 1/1-octave band guideline level.

Figure 1 also compares the various criteria against the range of measured ambient noise in eight world-class concert halls (Beranek, 1997). The data set only extends down to 31.5 Hz. However, the results are extremely interesting. The NCB 25 curve lies at the upper boundary of the measured concert hall range for low frequencies down to the 31.5 Hz band, and based on inspection of the spectrum shape, it seems likely that this would hold true for the 16 Hz band as well. The RC curves may be seen in this context as over-design – why should a bedroom have to perform better than a concert hall? While some audience members may fall asleep in concert halls, the primary use of the space is not for this purpose. Therefore, the audience may be more amenable to LFN in this context, rather than in their homes. Many of the long-term effects of LFN seem to be related to duration of exposure (ASHRAE, 1999).

The range of measured concert hall values presented above extends into the 'B' region representing moderately noticeable vibrations in the ANSI S12.2 standard. No complaints have been reported in these halls. Beranek conducted further comparisons of measured HVAC noise versus the 'B' range, and concluded that based on real-world data, there is little justification for including the 'B' range in the specification (Beranek, 1997). Schomer's proposed RNC curves do not include the 'B' range in their specification.

3. Proposed Facilities: Estimating LFN Impact

LFN impacts are best assessed indoors. However, two criteria could be used to limit outdoor impacts:

- ANSI B133.8: limits outdoor sound levels from gas turbine installations to 75 dBC at residential points of reception (to control sound-induced vibrations); and
- ANSI S12.2: ideally restricts LFN to "moderately noticeable" Region B, but in no case should they extend past the "clearly noticeable" Region A.

Predicting LFN indoor impact is more complex, given criteria differences. Our Canadian values suggest a compromise:

- The criteria choice should consider the existing ambient environment. Receptors in urban areas or near existing industry are likely to have relatively "high" existing LFN in their ambient sound environment. LFN from new facilities, may be more readily tolerated since the change from existing conditions would be smaller. Use of the NCB or RNC criteria in these locales seems reasonable. In rural areas with "flat" ambient spectra, and little LFN content, RC or RC Mark II curves would be appropriate.

  The type of noise source should also be considered. For example, a power plant having a few gas turbines and individual exhaust stacks would be less likely to produce time varying noise than one with several large diesel engines and bundled stacks (higher likelihood of beats).
- It seems reasonable to base assessments on total (plant + ambient) levels, requiring representative 1/1-octave band levels and overall dBA measured ambient noise levels. Where possible, maintain a balanced frequency spectrum (dBL - dBA • •20 dB, per Broner, 1994).
- Indoor levels can be estimated using typical building noise reduction characteristics. Where LFN impacts are likely, consider examining specific construction at worst-case receptors, and existing indoor noise levels.
- Limit LFN levels to the NCB 25 curve at low-frequencies (at or below 31.5 Hz), using a "tangency" approach and where economically and technically feasible, mitigate as close to the target RC Mark II value as possible.

4. References

Due to space constraints, a detailed reference list is not provided, but available upon request.

SOUND TRAVEL TIME FLUCTUATIONS CAUSED BY ATMOSPHERIC GRAVITY WAVES

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1. Introduction

Atmospheric gravity waves (AGWs), generated from various random sources (wind shear instabilities, jet streams, meteofronts, orography, convection), significantly contribute to the power spectrum of the mesoscale wind speed and temperature fluctuations in the atmosphere. The temporal scales of wave-associated fluctuations in the lower atmosphere may range from 1 min to several hours, whereas spatial scales are between hundred meters and several dozens of kilometers (Gossard and Hooke, 1975).

Despite an important role of AGWs in formation of a turbulent regime in the middle atmosphere and the stable atmospheric boundary layer (ABL) (Einaudi & Finnigan, 1993; Otte&Wyngard, 2001), the statistics of wave-associated wind speed and temperature fluctuations is poorly understood till now. The AGWs also cause the variations of refraction index for acoustic waves, propagating in the atmospheric wave ducts at long distances from their sources. To predict the mean square values of phase and amplitude fluctuations of acoustic waves one needs to know a form of the power spectrum of the mesoscale wind speed and temperature fluctuations in the atmospheric wave duct. Particularly, this is necessary for the acoustic sounding of stable ABL, acoustic source detection, and prediction of low-frequency sound levels from various explosive and noise sources.

Although a theory of sound propagation through an atmosphere with locally isotropic and homogeneous turbulence is well developed by now (Tatarskii, 1971), there is still a problem to describe the influence on sound propagation of energy-containing mesoscale atmospheric fluctuations, caused by AGWs. Such fluctuations are substantially anisotropic and inhomogeneous in space, and nonstationary in time, and their 4D frequency-wavenumber spectrum, needed for sound propagation modeling, is generally unknown to us.

In this paper the calculations of statistical characteristics of sound travel time fluctuations caused by AGWs are presented. These calculations are used to interpret the experimental data on the statistics of acoustic pulse travel time fluctuations in stable ABL.

2. Model of sound travel time fluctuations induced by AGWs.

The small-scale AGWs, which are often observed in stable ABL with the echo-records of sodars (Kjellaas *et al.* 1974), have rather low horizontal speeds, usually less than 10m/s. Therefore, for such waves are of great importance the effects of advection by the larger scale AGWs which are trapped within stably stratified layers of the lower troposphere, but penetrate down to the ABL (Gossard and Hooke, 1975).

The effects of advection of small-scale waves by a nonstationary wind induced by an entire spectrum of waves was taken into account in the models of AGW spectrum developed in recent works (Chunchuzov, 2002; Hines 2001). The strong nonlinear interactions between AGWs are shown to lead to the universal form for the high-wavenumber tail of the AGW spectrum. The 3D spectrum of both temperature and horizontal wind speed fluctuations decays with a wavenumber k as k^{-5}, whereas 1D vertical wavenumber spectrum takes a universal k_z^{-3}-form (Chunchuzov, 2002). Based on the obtained spectral forms the vaiances, frequency spectra and structure functions of sound travel time fluctuations have been calculated. The frequency spectrum normalized by the Brunt-Vaisaala (BV)-frequency N and the mean square value of travel time fluctuations $<\delta\tau^2>$ is shown in Fig. 1.

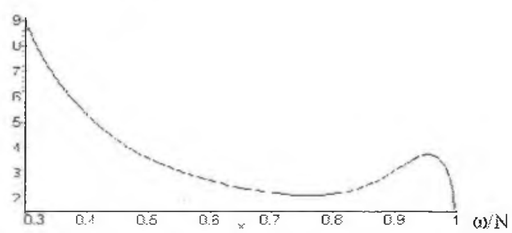


Fig. 1.The normalized frequency spectrum of sound travel time fluctuations caused by AGWs .

This spectrum decays with an increase of frequency ω as ω^{-2}, but when ω approaches the local value of BV-frequency N(z) at a height z the spectrum exhibits a local maximum (or "shoulder") due to in-phase interference at $\omega \sim N$ between incident and reflected gravity waves. Prediction of the spectral peak near N is consistent with the existence of a dominant period of 8 min in the experimental frequency spectra and coherences of acoustic pulse travel time fluctuations in the stable ABL (next section).

3. Measurements of acoustic pulse travel time fluctuations in stable ABL.

Several field experiments on acoustic pulse sounding of stable ABL have been carried out near Zvenigorod (Russia). One case of measurements of travel time of acoustic pulses, generated by a detonation of the air-propane mixture with a repetition period of 1min, is shown in Fig.2. To estimate the horizontal coherences, the speeds and scales of the observed wind speed fluctuations in stable ABL we have placed two acoustic reseivers at the same distance of 2.6 km from a pulse source, but along different azimuths relative to the mean wind speed direction.

The 1.5-h time series of the fluctuations of the pulse travel time $\delta\tau$ are shown in Fig.2 along with the fluctuations of the effective sound speed C_{eff} (measured by the anemometer at a height of 56 m), and atmospheric pressure P (measured by microbarograph) . For these time series, obtained at the three different points 2.6 km apart, the calculated auto-spectra and the coherences are shown in Fig. 3

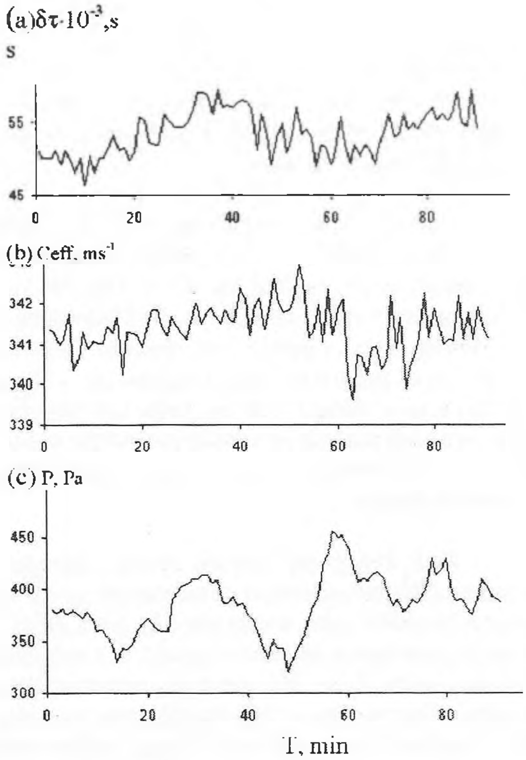
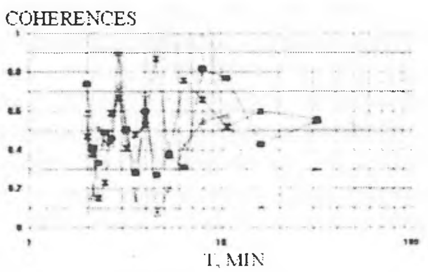


Figure 2. The time series of travel time fluctuations T, effective sound speed Ceff and atmospheric pressure P during October 10, 1995, obtained at the three points 2.6 km apart. a) T; b) Ceff at z=56 m ; c) P.



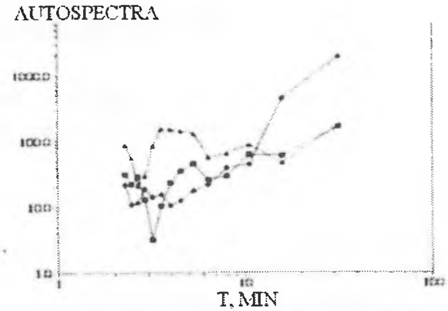


Figure 3. Coherences and Autospectra for the time series shown in Figure 2.

In Fig.3 the obtained autospectra tend to decrease with an increase of period T, but have a broadband maximum ("shoulder") between 5 min and 10 min. As follows from the theoretical model of AGW spectrum in section 2 such a maximum may arise due to contribution to the observed wind fluctuations from the trapped gravity wave modes in the lower atmosphere. The appearance of the "shoulder" in the frequency spectra of travel time and wind speed fluctuations in stable ABL shows that AGWs significantly contribute to the observed fluctuations. This is also confirmed by a presence of the peak of coherences at $T\approx$ 8-10 min seen in Fig.3. For T ranging from 8 to 10 min the sum of the phase differences between each pair of three receivers is close to zero, and this also indicates that such fluctuations are likely caused by AGWs. The intrinsic frequencies of the observed AGWs are close to the typical BV-frequency values ($\sim 2\cdot10^{-3}$ Hz) in the troposphere, where the mean temperature lapse rate is about 6° C per 1 km. Therefore, we suggest that observed AGWs may be trapped in stable layers of lower troposphere, where a profile of N(z) has a typical maximum value close to $2\cdot10^{-3}$ Hz (Gossard and Hooke, 1975). The estimated from the phase spectra horizontal phase speed and wavelength λ are about 7m/s and 3.4 km, respectively.

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EVALUATION OF WIND TURBINE NOISE

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1. INTRODUCTION

Virtually anything with moving parts will make some sound, and wind turbines are no exception. Well-designed wind turbines are generally quiet in operation. Wind turbine noise is very low compared to the noise from road traffic, trains, aircraft and construction activities and can also be compared to the sound level inside a typical living room, the reading room of a library or in an unoccupied, quiet, air-conditioned office. Typically, the sound of a working wind farm is actually less than normal road traffic or an office.

2. NOISE MEASUREMENT

The noise a wind turbine creates is normally expressed in terms of its sound power level. Although this is measured in dB(A), it is not a measurement of the noise level which we hear but of the noise power emitted by the machine. The sound power level from a single wind turbine is usually between 90 and 100 dB(A). This creates a sound pressure level of 50-60 dB(A) at a distance of 40 metres from the turbine (about the same level as conversational speech). At a house 500 metres away, the equivalent sound pressure level would be 25-35 dB(A) when the wind is blowing from the turbine towards the house. Ten such wind turbines, all at a distance of 500 metres, would create a noise level of 35-45 dB(A) under the same conditions. With the wind blowing in the opposite direction, the noise level would be about 10 dB(A) lower. It must be noted that the turbine noise estimates do not take into consideration the wind noise that is providing a masking effect.

3. TURBINE TECHNOLOGY

3.1 Turbine Description

Almost all wind turbines that produce electricity for the grid consist of a tower between 40 and 80 metres high, a nacelle (housing) containing the gearbox and generator mounted on top of the tower, and three blades that rotate around a horizontal hub protruding from the nacelle. This type of turbine is referred to as a horizontal axis machine.

There are two potential sources of noise: the turbine blades passing through the air as the hub rotates, and the gearbox and generator in the nacelle. Noise from the blades is minimised by careful attention to the design and manufacture of the blades. The noise from the gearbox and generator is contained within the nacelle by sound insulation and isolation materials. Standing next to the turbine, it is usually possible to hear a swishing sound as the blades rotate, and the whirr of the gearbox and generator may also be audible. However, as distance from the turbine increases, these effects are reduced.

3.2 Turbine Types

Wind turbines may be designed in different ways and many of the differences have come about from a desire to minimize noise emissions.

Upwind & Downwind Machines: The majority of horizontal axis turbines are designed in such a way that the blades are always upwind of the tower. This has the effect of minimizing any airflow changes as the blades pass the tower. Some turbine designs, particularly some of those installed in the USA, have the turbine blades downwind of the tower. With this type of design, a strong pulse can sometimes be heard with each passing of a blade behind the tower. However, most turbines currently operating in Alberta are of the upwind design.

Twin Speed and Variable Speed Machines: Most horizontal axis turbines rotate at a constant speed, usually between 25 and 50 rpm, irrespective of wind speed. However, twin speed machines operate at a reduced speed when the wind is light. This produces less noise and means the noise of the turbine is also significantly lower by up to 10 dB(A). Variable speed machines change speed continuously in response to changes in wind speed and, although noise output may be higher at higher wind speeds, it is lower at low wind speeds where the low background levels occur.

Direct Drive Machines: Direct drive turbines are the latest design concept in turbine technology. Simply put, these machines have no gearbox or drive train, and consequently no high speed mechanical (or electrical) components. Direct drive turbines are therefore much quieter than gearbox machines as they do not produce mechanical or tonal noise. An example of this type of turbine is the 1.5MW 'Ecotricity' turbine.

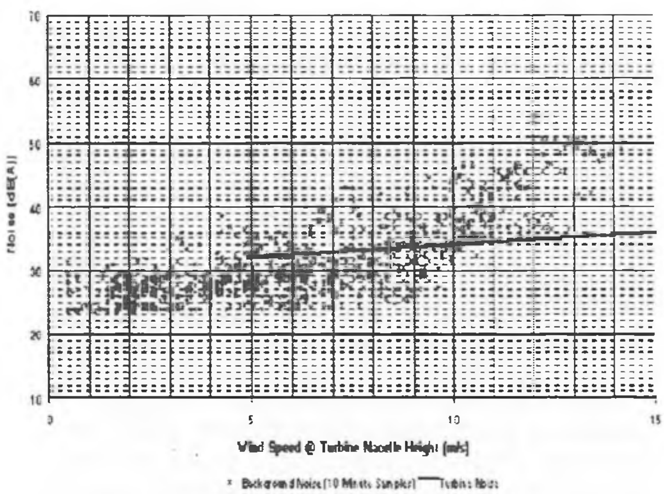
3.3 Considerations for New Wind Projects:

When planning a wind turbine project, careful consideration must be given to any noise that might be heard outside nearby houses. Inside, the level is likely to be much lower even with windows open. The potential noise impact is usually assessed by predicting the noise that will be produced when the wind is blowing from the turbines towards the houses. This is then compared to the background noise that already exists in the area, without the wind farm operating.

There is an increase in turbine noise level as wind speed increases. However, as seen in the Graph below, the noise from wind in nearby trees and hedgerows, around buildings and over local topography also increases with wind speed but at a faster rate. Thus, it is difficult to detect an increase in turbine noise because of the increase in the background sound level. Also, wind turbines do not operate below a specified wind speed referred to as the cut-in speed (usually around 15 km's per hour). Wind data from typical sites suggests that wind speeds are usually below the cut-in speed for about 30% of the time.

It has been suggested by some regulators that turbine noise level should be kept within 5 dB(A) of the average existing evening or night-time background noise level. This is consistent with standard approach the EUB uses for noise impact assessment of energy industry sources, except for construction related noise that currently has no specific limit.

Background Noise and Turbine Noise vs. Wind Speed



3.4 Resident Views:

Despite technical evidence that suggests otherwise, a small number of residents living near wind farms in Europe have stated that in their experience mechanical noise is insignificant compared to the aerodynamic noise, and characterized by some as "blade thump". Residents have also pointed out that the mechanical noise is usually only audible within 100 meters of the turbine, but the blade thump can be heard at distances of up to 1.5 km away. Although measurements indicate the noise is not particularly loud at this distance, residents claim it can be extremely irritating when exposed to it for any length of time because of its rhythmic nature making it hard to disregard.

It is believed that the blade thump is caused by the blades passing the tower of the turbine. The rotational speed of 3-bladed turbines is about 28 rpm at maximum rotational velocity. This results in a rhythmic sound comprised of about 84 beats per minute from each turbine. This sound rises and falls in volume due to slight changes in wind direction. Some residents have described the effects of this sound as a feeling of anxiety, and sometimes nausea. Some have expressed belief that the frequency of the blade thump is close to the human heart rate; and some residents feel that their own pulse rate is trying to match that of the turbines.

Statistics however, taken from recent surveys suggest that the number of those concerned about noise dropped from 86% immediately after construction to 20% a year later. Of these only 3% of respondents indicated that they were bothered by the wind plant's noise.

4. Conclusion

Wind and other renewable energy sources enjoy strong popularity with the public, a logical outgrowth of increasing concern about the environment and the perception that renewable energy sources have less environmental impact than their fossil and nuclear counterparts. As its visibility grows, wind is likely to add to its already strong latent public support and to become one of the most preferred electricity generation options of the next decade.

With respect to noise produced by typical wind farms, the level is so low that they would not be noticeable in most urban residential areas. However, the areas suitable for such developments tend to be in quiet but exposed areas of countryside. Therefore some amount of effort must be put into minimizing any noise impact. Nonetheless, it should be emphasized that typical noise levels from wind farms are so low that at a carefully considered site they would normally be masked by other background sources.

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Note: *The above is taken directly from an extensive literature search into wind turbine noise and the findings in the above publications validated by field research conducted by the EUB in cooperation with wind power generators Canadian Hydro, and Vision Quest Wind Electric Inc.*

PROPAGATION LOSS MEASUREMENTS FOR LOW FREQUENCY SONAR IN EMERALD BASIN AND EXUMA SOUND

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1. INTRODUCTION

Knowledge of the characteristics of sonar propagation in a given environment is essential for the proper employment of a sonar system. A low-frequency hyperbolic frequency modulated signal (HFM) was employed to measure propagation loss in both a deep water environment and in a shallow water surface duct environment. The signal was processed both incoherently, using time-domain energy-detection, and coherently, using a matched filter to determine the degree of coherence loss. The experimental propagation loss results were then compared to computational models, including the Generic Sonar Model (Weinberg, 1985), an eigenray model, and SWAMI (for the shallow water environment), which uses a normal mode model.

2. METHOD

Two locations were chosen for measurement of propagation loss. The first location was Exuma Sound, approximate location latitude 24° 23′N, longitude 76° 9′W. The bottom depth measured at the beginning of the data collection run was 1762 m, with a sound speed profile given in Figure 1. The transmitter was at depth 31 m. A series of three 0.5 second HFMs were transmitted with bandwidth 25 Hz, start frequency 1125 Hz, 1175 Hz, and 1225 Hz, and dwell time 0.1 second. The series was transmitted at one minute intervals.

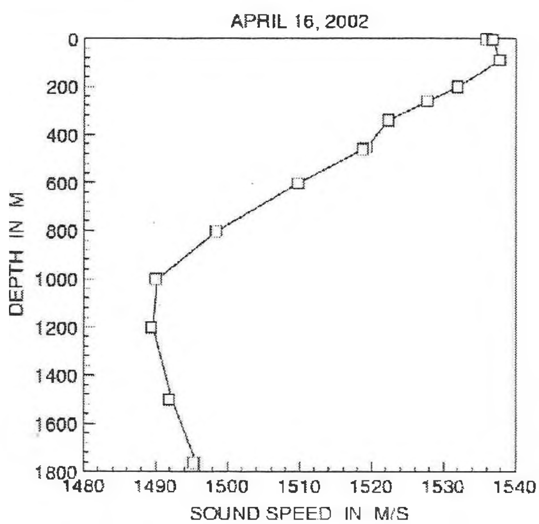

Figure 1. Profile of sound speed vs. depth for Exuma Sound.

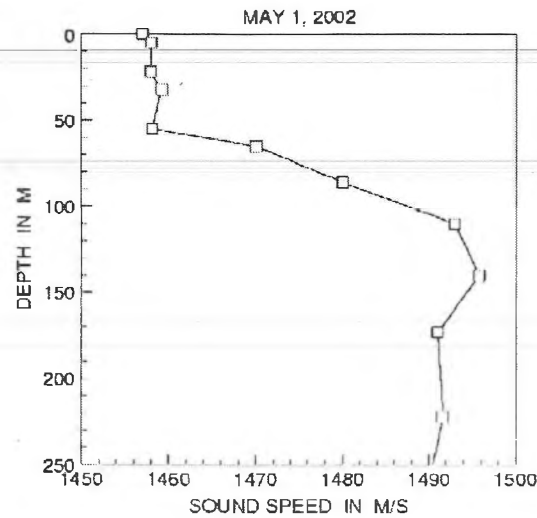

Figure 2. Profile of sound speed vs. depth for Emerald Basin.

The receiver was a Combined Omni Resolved Directional Sensor (CORDS) used in omnidirectional mode, being towed at 50m depth at a speed of 8 knots.

The second location was a comparatively shallow-water site, Emerald Basin, latitude 44° N, longitude 62° 53′W. The bottom depth measured at the beginning of data collection was 260 m and the sound speed profile was as shown in Figure 2. The source in this case was a two-element free flooding ring vertical projector array (VP2) at depth 58 m. The transmit signal was a sequence of one-second HFMs with bandwidth 50 Hz, centre frequency 1150 Hz, 1200 Hz and 1250 Hz and a 30 second dwell time. The receiver was the DRDC Atlantic UAT (Underwater Acoustic Transponder) at a depth of 56 m, using data from one omnidirectional hydrophone.

3. ANALYSIS

3.1 Exuma Sound data

Approximately 4 hours of data was collected in Exuma Sound, translating to about 60 km of range measurement. Figure 3 summarizes the results for the 1125 Hz signal. Both the coherent (matched filter) propagation loss and the propagation loss measured using incoherent processing are very close to those predicted from the model. The model in this case is a range-independent multipath expansion eigenray model. The model results are shown for

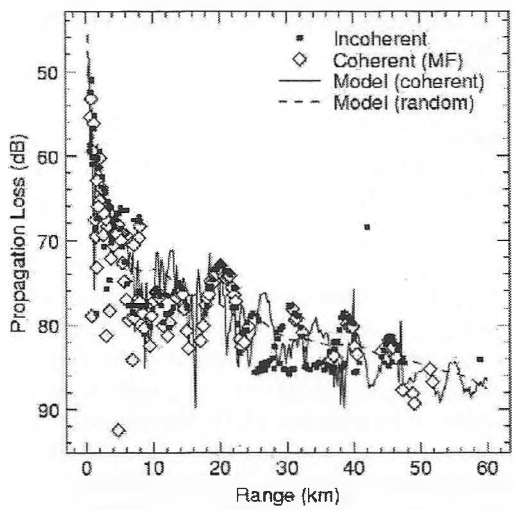

Figure 3. Propagation loss measurements and model for Exuma Sound.

both coherent and random eigenray phase summations. Although the exact nature of the bottom is not known, the model results are quite insensitive to the bottom reflection model used.

There is very good agreement with the data collected and the model; the coherently summed multipath model accurately represents the overall behaviour of the data, although some of the peaks and troughs of propagation loss are not in the same location as they appear in the data, as can be expected from a homogeneous, range-independent model. The difference between the incoherent detector loss and the matched filter loss is fairly small: figure 5 has a comparison for Exuma Sound and Emerald Basin.

3.1 Emerald Basin data

The data from Emerald Basin has measurements over about 55 km of range. The propagation loss data for the 1150 Hz signal and two models are shown in Figure 4.

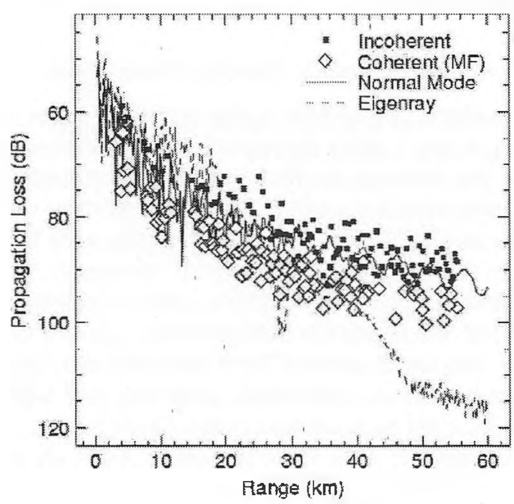

Figure 4. Propagation loss measurements and models for Emerald Basin.

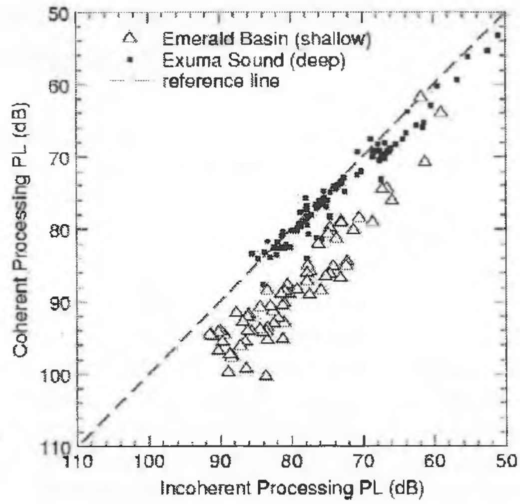

Figure 5. Comparison of propagation loss measured using coherent vs. incoherent processing for deep and shallow water.

In this case, the eigenray model used previously shows a much greater propagation loss occurring at longer ranges than is evidenced by the data. The normal mode model, however, shows a very good agreement with the data. The model used here is a range-independent one. Both the normal mode and eigenray models are very sensitive to surface loss parameters (wave height or sea state).

A comparison of the propagation loss measured using the different types of processing is given in Figure 5. It is evident that the shallow water environment leads to significantly more loss of coherence and thus propagation loss, particularly at extended ranges.

4. CONCLUSIONS

For the data collected here, matched filter coherent processing gives close agreement, to within 2 dB to 3 dB over 60 km of range, to incoherent processing for the Exuma Sound deep water environment. However, the shallow water measurement seems to indicate that loss of signal coherence increases the propagation loss measured, in this case by 5 dB to 10 dB over this range.

The modelling of propagation loss for both coherently and incoherently processed signals also depends on the environment. Here, it is seen that a multipath eigenray model gives excellent agreement to measured propagation loss in a deeper water environment, with little dependence on bottom type. On the other hand, the surface-ducted shallow water environment is more accurately modelled using normal modes, and is quite sensitive to surface conditions. Ideally, the shallow water environment might be modelled using a range-dependent normal mode technique.

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MEETING INTERNATIONAL STANDARDS FOR LOW FREQUENCY UNDERWATER RADIATED NOISE FROM SHIPS

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1. INTRODUCTION

There is a renewed concern to minimize the underwater radiated noise from commercial vessels and, in particular, fisheries research vessels. The International Council for the Exploration of the Sea (ICES) has issued recommendations for maximum radiated noise levels over a broad spectrum [1]. This includes very low frequency (less than 50 Hz) narrowband noise, which is not amenable to prediction with energy-based or empirical methods. DRDC has developed methods for predicting such radiated noise using a combination of finite element and boundary element techniques incorporated into a computer code called AVAST [2]. This paper will demonstrate the procedure for such a noise prediction using a generic ship model with excitation provided by sample vibrations measured on engine mounts. The analysis will proceed from a coarse MAESTRO [3] model of the vessel, through to a finite element model where the loads will be applied and the natural frequencies calculated, to a radiated noise prediction using the AVAST software. The resulting prediction will be compared to the ICES recommendations up to a frequency of about 50 Hz.

2. NUMERICAL MODELLING

A generic ship model was constructed using the MAESTRO software, which allows for the rapid generation of a coarse finite element (FE) model of a vessel. The vessel has a length of 75m (BP), a beam of 12m, and a draft of 3.5m. The FE model is shown below in Figure 1 and contains 3566 nodes and 8041 elements.

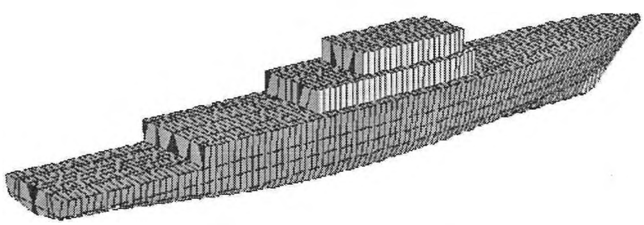

Figure 1: Structural Finite Element Model

This specialized MAESTRO model can be converted to a standard FE model using available translators. An in-air dynamic analyses of the model was performed to determine the *dry* natural frequencies of the ship which are required input for the AVAST analysis. As well, loads were applied to the model to simulate dynamic engine loads from a diesel generator. To determine the appropriate load levels, unit loads were applied, a modal frequency response analysis performed, and the resulting accelerations of the engine mounts (with the ship in water) were compared to some values available in the literature [4,5]. The applied loads were scaled to correct for the differences between the predicted and measured amounts. This procedure was iterated until convergence. The resulting loads were then available for the radiated noise analysis. While there was some variation in the measured data, the loads were assumed to be constant with respect to frequency to ease the analysis. An average over the frequency range was used resulting in a vertical displacement amplitude of 1.3 E-06m and a transverse amplitude of 5.0 E-07m at each engine load point. Modal damping factors were assumed to be a consistent value of 0.002 based in part on data measured on a scientific research vessel.

Given a description of the dry natural frequencies and a load file, the boundary element based AVAST code can be used to predict low frequency radiated noise. The program requires only a model of the wetted surface of the vessel and the required model (containing 1132 panels) is shown in Figure 2.

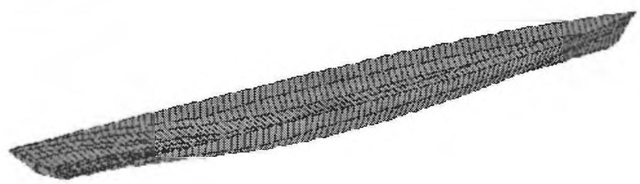

Figure 2: Wetted Surface Boundary Element Model

Once the model is read and the modes and forces input, the user must generate a set of field points for the radiated noise prediction. For this analysis, the field points were located at a depth of 10m from the surface and a range of 100m from the approximate X-Y centre of the fluid model (with Z vertical). 36 field points spaced every 10° were used. The user indicates that there is a reflecting surface (located at the draft line) and also inputs the fluid properties (density of 1025 kg/m^3 and sound speed of 1500 m/s) and the frequency for this analysis. As the fluid matrix properties vary with frequency, multiple analyses were run with a frequency increment of roughly 2 Hz. Once this data is input, the type of analysis is selected. Upon choosing elastic radiation, the user may then vary the modal damping ratios from the default values of 0.002.

AVAST calculates the resulting acoustic radiation at the input frequency and at all modes above that frequency. The resulting radiated noise pattern may be viewed onscreen or, as in this case, the results may be copied to a spreadsheet for further analysis.

3. RESULTS & DISCUSSION

The predicted radiated noise level at the 100m distance is plotted versus frequency in Figure 3. The lower three curves show the sound level for the broadside and bow aspects, as well as the maximum at each frequency. The strongly radiating resonances can clearly be seen as can the variation of the maximum with respect to aspect (no single aspect dominates).

The primary reason for this analysis was to compare the ship's signature with that recommended by ICES. In the frequency range used here, the recommended ICES source level limit (SL) in dB is given in [1] as SL = 135–1.66 log(f) where f is the frequency in Hz. The above results in Figure 3 are not given as source level, which is in dB re 1 μPa at 1m. If one assumes simple spherical spreading, one can convert an omni-directional pattern to a source level by adding 20 log (r) where r is the field point radius in metres (40 dB in this case). The assumption was thus made to use the maximum at each frequency and use spherical spreading. The results are also shown in Figure 3, together with the ICES level for this frequency band. Note that, at first, only the resonant peaks exceed the ICES level, but after about 10 Hz, the radiated source level completely exceeds the recommended level. As this was an artificial analysis to some extent, there is no real concern, however, this type of analysis can show where problems might occur and possible solutions could be examined at the design stage.

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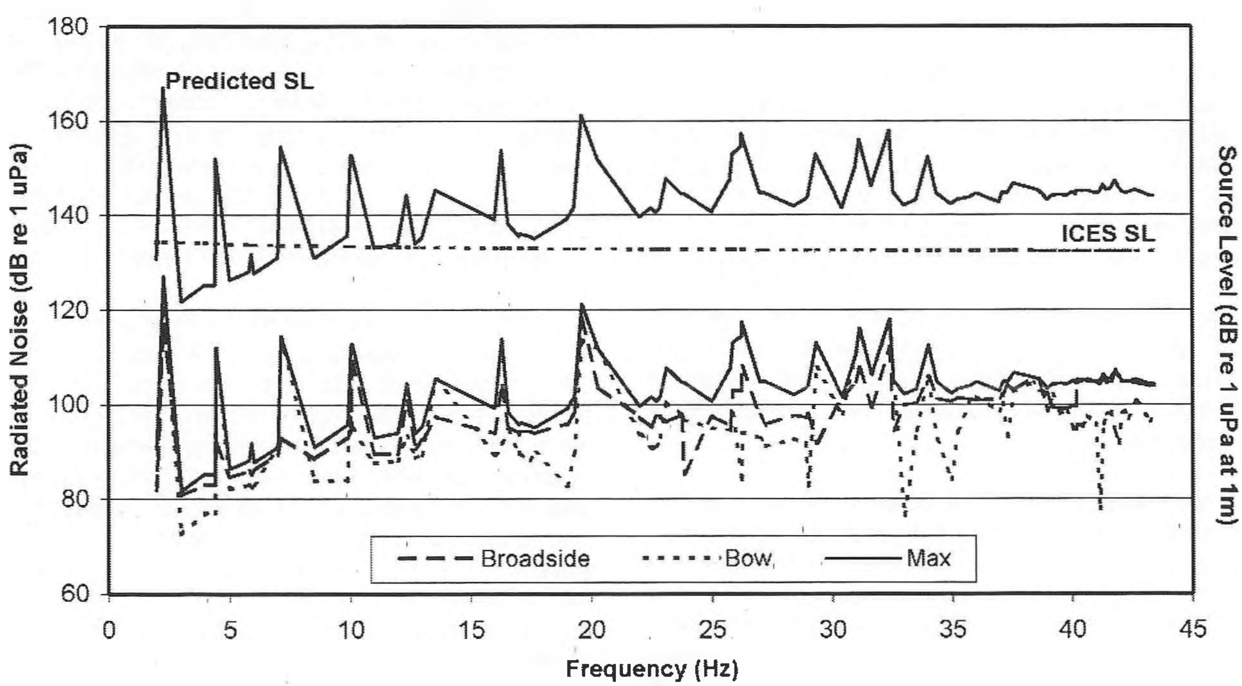

Figure 3:Radiated Sound Level and Source Level vs. Frequency

COMPLEX SHOCK LOADING ON SUBMARINE OIL PIPELINES

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1. INTRODUCTION

Underwater explosions are one of the most dangerous accidental loads that a submarine pipeline can experience. Not only can they completely destroy a structure, they can also cause a significant damage to the marine environment. Therefore, careful dynamic analysis of a pipeline when it is being subjected to a hydrodynamic shock wave is one of the primary goals when the safety of an offshore installation is a concern.

The paper concerns with a structural analysis of a fluid-filled submerged elastic circular cylindrical shell subjected to a shock wave. The interaction between a circular cylindrical shell and a hydrodynamic shock wave has been a subject of intensive investigation for the last few decades. In the vast majority of the published works, a step-exponential shock wave was considered, i.e. a shock wave with exponentially decaying pressure behind the front. Although this well-studied classical model allows a very accurate analysis in some cases, quite often more complex models are required. This occurs, for example, when reflections of a shock wave from rigid walls and/or the free surface are present. In this case, the pressure pattern behind the wave front can become quite complicated.

The situation when the free surface has an influence on the interaction process is of special practical interest. In this case, along with the first (primary) peak of pressure, a shock wave has a few secondary peaks [2]. The most noticeable of those is the second (negative) one. It is associated with the reflection of a shock wave from the free surface, and its magnitude can be of the same order as the magnitude of the primary one. Therefore, it is almost certain that the influence of a shock wave with such a complex pressure profile will differ quite significantly from the case when a step-exponential shock wave is analyzed. Thus, addressing shock loads with multiple pressure peaks appears to be worth pursuing.

There is another reason for the discussed study to receive some attention. As it has recently been found [3], the stress state of a submerged fluid-filled shell is determined by multiple wave effects in both interior fluid and a shell. This observation was made for a step-exponential shock wave. Since shock waves with multiple pressure peaks bring the next level of complexity into the wave patterns of the process, it is reasonable to expect that resonance-like phenomena are likely to happen in this case. Obviously, the study of these phenomena is of considerable practical interest. Therefore, there is a need to extend the previously accomplished research to the case of a shock wave with a more complex pressure profile.

2. MATHEMATICAL APPROACH

The equations of shell dynamics are derived using Hamilton's principle and Love's classical expression for strain-energy [4]. The fluids are assumed to be linearly compressible, and driven by the wave equation. Both fluids and a shell are coupled through the dynamic boundary condition on a shell surface. Therefore, we are dealing with two wave equations for fluid potentials, coupled with the system of equations for shell displacements, all of them being time-dependant.

As to the used methodology, the problem was solved in two steps. First, hydrodynamic pressure was obtained under the assumption that the normal displacements of a shell are known. At this stage, separation of variables was used to eliminate the space coordinates, and the Laplace transform was applied to the time one. As a result, a series representation for the total hydrodynamic pressure at the shell surface was obtained, containing, in integral form, the normal displacements (which were still unknown). Then, the derived analytical solution for the pressure was numerically coupled with the shell equations, and the spectral technique was used here. Finally, the developed hybrid analytical-numerical solution was used to simulate the interaction, and, in particular, the stress-strain state. More detailed discussion on the proposed solution scheme can be found in [3].

3. RESULTS AND CONCLUSIONS

A steel shell submerged into water and filled with oil was considered. The radius of a shell and the wall thickness were 0.50 m and 0.005 m respectively. A shock wave with one primary (positive) pressure peak and one secondary (negative) peak was considered to model a real underwater explosion similar to the one addressed in [2]. The magnitude of the secondary peak was chosen to be equal to a half of that of the primary peak. The influence of three different shock waves was analyzed. All of them had the primary peak at 0 ms (i.e. at the moment of the initial contact between a shock wave and a shell), and the secondary peaks at 0.42 ms (SW-A), 0.14 ms (SW-B), and 0.82 ms (SW-C). The magnitudes of the pressure peaks

(1250 kPa and –600 kPa) were adopted from the experimental data [2], and were the same for all the considered shock waves. The primary and secondary peaks were assumed to have the same rate of exponential decay [1]. The results were compared to a step-exponential shock wave with only one positive peak at 0 ms (SW-0).

Figure 1 shows the dynamics of the transverse stress for SW-A, SW-B, and SW-0 at the rear point of a shell. One can see that the maximum stress for SW-A is about 35% higher than that for SW-0, whereas the difference between the stresses caused by SW-B and SW-0 is insignificant (in terms of the maximum magnitude). Therefore, the resonance does take place, and it is not the magnitude of the secondary peak that determines the destructive effect that a shock wave has on a structure but the timing when the peak occurs. Namely, it has been observed that the resonance happens only if the secondary peak occurs at times close to 0.4 ms. For all other timings, the secondary peak does not cause any significant increase of stresses. Note that the maximum stresses for SW-A and SW-0 have different signs.

The observed timing of resonance has a clear physical interpretation. First, we recall that it takes $\pi r_0/c_1$ for an elastic wave originated at the front point of the shell to reach the rear point (here c_1 is the sound speed in the shell material, and r_0 is the radius of a shell), and $2r_0/c_2$ for a hydrodynamic wave in the interior fluid to come to the same point (c_2 is the sound speed in the interior fluid). Then, it becomes clear that the hydrodynamic wave in the interior fluid (caused by the primary peak of pressure) and the elastic waves in a shell (caused by the secondary peak) will arrive at the rear point at the same time, superposing and causing much higher stresses, only if the secondary peak occurs at about $t_s=r_0(2/c_2-\pi/c_1)$. For the considered system, this formula gives $t_s \sim 0.48$ ms. The observed timing is slightly different because the elastic waves, as long as they have reached the rear point, need some time to actually superpose to cause a significant increase of stresses.

For the considered geometry, we define R_0 as the distance between the source of the shock wave and the free surface. Then, it is easy to show that the resonance only happens when

$$r_0=2R_0c_1c_2/(c_0(2c_1-\pi c_2)), \qquad (1)$$

where c_0 is the sound speed in the exterior fluid. In particular, for the considered system (water-steel-oil) we have $r_0 \sim 1.5R_0$. This formula allows one to determine the location of an explosive that is particularly dangerous for a specific pipeline, and also to predict the radius of a pipeline that will be most sensitive to an explosion with specific parameters. It should be especially noted that the distance between the source of a shock wave and a structure is not present in formula (1). Similar formulas can be derived for other geometries common in offshore engineering (rigid walls, sea bed etc.), allowing a preliminary analysis of maximum stresses without any complicated computations.

The conducted study results in the following conclusions. (1) A destructive influence that an explosion can have on an underwater pipeline significantly depends on the location of an explosive with respect to a structure and the free surface and/or walls and other obstacles. (2) When a submarine oil pipeline is being designed, a dynamic analysis of the whole system is important: it is necessary to make sure that, for the particular conditions of installation, all the resonance phenomena caused by the reflections of a potential shock wave are taken into consideration as a possible risk factor.

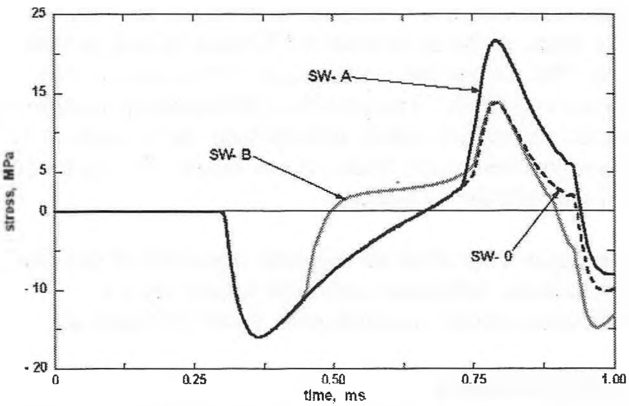

Figure 1. Transverse stresses in a submarine oil pipeline for three different shock waves.

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ACKNOWLEDGEMENTS

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ICESHELF 2002 – UNDERWATER ACOUSTICS IN THE ARCTIC ENVIRONMENT

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1. INTRODUCTION

Iceshelf 2002 was an Arctic trial organized by Defence R&D Canada Atlantic (DRDC Atlantic), in collaboration with several government, industry, and university partners. The overall purpose of the trial was to test components and algorithms in support of the Rapidly Deployable Systems Technology Demonstration Project.

This Arctic trial was conducted on shore-fast ice west of CFS Alert, on the north coast of Ellesmere Island, in March-May 2002. Living and working in an Arctic environment is always a challenge. This trial involved setting up a camp on the ice, several kilometers from the base, and working from this camp over several weeks. It was logistically, physically, and scientifically a challenge.

This paper is not about the scientific objectives of this trial, but about the difficulties associated in carrying out underwater acoustic research in an Arctic environment.

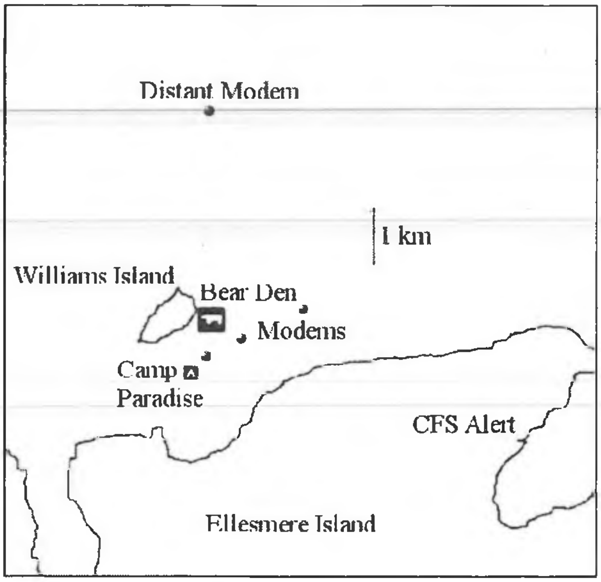

Fig. 1. The ice camp location near CFS Alert (Camp Paradise). The modem locations represent locations where gear was deployed through the ice.

2. LOGISTICS

For this trial we shipped approximately 12 tons of gear to our storage facilities in Alert. From there, we used skidoos and BV 205 tracked vehicles to carry everything necessary to set up a camp for 20 people on the ice, and conduct scientific experiments over a period of four weeks. The task was daunting.

Figure 1 shows Canadian Forces Station Alert, the most northerly permanently inhabited place in the world. The station was established in 1958; its population has varied widely since its inception, with ~220 people at its peak. Nowadays, approximately 60 people occupy the base year-round. Though some civilians live on the base (a contingent of Environment Canada maintain a weather station near Alert), it remains a military town that is resupplied almost entirely by military aircraft. Because of its northern status, the base has been the launching point of many North Pole or other northern expeditions, but needless to say, its access to civilian organizations requires a great deal of interaction with the military. DRDC is privileged in this respect because of its ties with the military.

Following a joint US-Canada trial of the 1990s, the Spinnaker project, DRDC maintains a storage facility in Alert. The Spinnaker building is a 446 m^2 area filled with Arctic gear: everything from skidoos to Arctic tents and stoves, scientific gear and ice drilling devices.

Upon our arrival in Alert, a small skidoo party went on the ice to find a suitable location for the camp. We have to drill many holes through the ice to conduct our experiments, therefore it is worthwhile to spend extra time to find an area of annual ice. One-year old ice is roughly 1.5 to 2 m thick, and multi-year ice and ice ridges near Alert can be much thicker. Annual ice also tends to be smooth, simplifying the camp set up.

In 2002, we found an area 6 km from Alert, shown in Figure 1. Thirty-eight people participated in various parts of the Iceshelf 2002 trial, and a camp was set up for approximately 20 people on the ice. This camp, shown in the background of Fig. 2, included two scientific tents, a double battery/workshop tent, a tent for our remotely-operated vehicle, a fully-equipped kitchen tent, a food storage tent, 6 accommodation tents, and an under-appreciated "Hurritent" with toilet facilities. All of these tents are strong insulated nylon tents and, except for the food storage tent and the Hurritent, are heated with small oil stoves.

It took approximately 10 days, 15 people, a fleet of skidoos, and help from the base BV tracked vehicles to carry all the necessary supplies and set up the ice camp, at a temperature below –30°C.

3. THE ENVIRONMENT

The mean daily temperature in Alert varies from about –33°C in March to –12°C in May. To work in these temperatures requires many layers of bulky fur-laden clothing, making physical labor difficult and slow. High-caloric diets are required. Long restful pauses around warm meals punctuate the long days: by late March, the sun is up 24-h a day, and one often forgets how late it is.

The cold is hard on the people and on the equipment. At –35°C, many materials become too brittle to use. Tent covers shrink, wires and fuel lines break, oil thickens. As temperatures reach –40°C even some kerosene fuels gel.

Batteries lose capacity quickly in the cold. For small gear, the use of Nickel Metal Hydride batteries instead of the traditional alkaline batteries helps. Either type must be kept warm. For larger pieces of gear, car batteries or generators have to be used. Generators are difficult, if not impossible, to start after spending even a short time in the cold.

Years of experience are required to find out what will work and what won't in the Arctic environment. Equipment has to be built very rugged, with simple designs – what breaks will need to be fixed by someone wearing thick gloves.

4. THE SCIENCE

By far, the biggest challenge remains the sheet of ice between our camp and the ocean. To investigate acoustic propagation underwater, one needs to drill holes through thick ice. Over the years, DRDC has designed many tools to solve this problem.

For small holes (a few inches in diameter), DRDC has been using mechanical drills, either powered manually, or electrically. Drill stems are 2 or 4' long, with quick disconnect mechanisms. If the ice is very thick, tripods, such as the one shown in Fig. 2, are used to support the weight of the drill.

Fig. 2. A tripod is setup to support the weight of an ice drill; the ice camp is in the background.

Fig. 3. A hot water drill is used to cut blocks of ice. The cut area will be large enough the accommodate the Phantom: a large, remotely-operated vehicle used to deploy our gear underneath the ice.

For larger holes, a hot water drill was designed by DRDC. Cold water is pumped through a reservoir using a small electric motor, heated with a furnace burner, and redirected to the ice through various sized "cookie cutters". In Fig. 3, a 4-ft horizontal straight edge is used to cut the ice in blocks. These blocks are then pulled out with skidoos and an A-frame lifting device.

Once the holes are ready, the gear is lowered in the water. If the gear is meant to be deployed over a distance, such as a horizontal acoustic array, lines are used to pull the gear between holes. A remotely operated vehicle is used to carry the line between the holes.

For acoustic studies, it is vital to reduce our own contribution to noise. Electrical noise, from generators for example, is particularly difficult to isolate. During Iceshelf 2002, we used 50 car batteries wired in series-parallel to run the equipment. Generators ran at night to recharge the batteries. Our own acoustic noise, which can couple through the ice to the ocean, can also be a nuisance. Skidoos, generators, even people walking on the ice can trigger noise that will contaminate our acoustic datasets. Only a great deal of coordination can solve this problem.

5. CONCLUSION

You might conclude that Arctic work requires considerable expertise and logistical capability. Sometimes that may be difficult for small groups to achieve. DRDC has over 30 years of experience working in arctic conditions, and is always open to collaborative projects.

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ACKNOWLEDGEMENTS

The authors wish to thank all of the contributors to Iceshelf 2002 and CFS Alert for their support.

USING SEDIMENT POROSITY AS AN EFFECTIVE GEOACOUSTIC PARAMETER FOR SONAR PERFORMANCE PREDICTION

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1. BACKGROUND

Performance prediction modelling for active sonar requires the geoacoustic properties of the surficial seabed sediments in order to account for the transmission loss and reverberation due to seabed interaction. Calculating the transmission loss and scattering of acoustic energy typically require the geoacoustic properties to be parameterized in terms of their density, compressional and shear sound speeds, and their associated attenuations. These parameters are often measured on physical samples, such as cores, or obtained from inversions of purpose designed acoustic experiments. The former are costly to collect and prone to artifacts due to disturbance during collection, handling and storage. In addition, the measurements must be corrected to the *in situ* conditions, notably temperature and pressure, and possibly for dispersion effects as the sound speed on cores is typically measured at much higher frequencies (hundreds of kHz) than that of interest. Numerous acoustic techniques have been developed to obtain geoacoustic parameters and can yield excellent results. However, these techniques typically require specialized equipment and detailed analysis that preclude their widespread use. This challenge provides the motivation to develop instruments or techniques that reduce the number of independent parameters that must be measured to effectively parameterize the seabed. This is particularly true in a rapid environmental assessment scenario in which the number and type of measurements must be limited and the analysis streamlined.

2. INTRODUCTION

The porosity of a marine sediment is the volume of the interstices, that is the pore space, between the sediment grains, per unit volume of sediment. In marine sediments, the pore space is typically filled with seawater though it is also possible to have gas. Empirical studies suggest that the porosity of the surficial sediments is a physical property of the seabed to which the more traditional properties (density, attenuation, and sound speeds) may be related [1, 2]. For example, using data from over two hundred cores collected in littoral waters, using divers or box cores to minimize artifacts, Briggs and Richardson [2] have established a regression relationship between porosity, n, in percent, and compressional sound speed, V_p, given by $V_p = 1.574 - 0.015n + 0.001n^2$ with a coefficeint of determination of $r^2 = 0.954$. In their research, this and other relationships are used to determine seabed properties from normal incidence measurements of acoustic impendance. Noting that porosity strongly controls geoacoustic properties of sediments, Prior and Marks [3] have developed a series of nine seabed geoacoustic models representing regular increases in porosity that start from rock, move through coarse grained sediment and end with fine grained mud. They argue that these models provide a satisfactory parameterization for the majority of ocean sediments and use them as the basis for a 'pragmatic approach' to modelling transmission loss and inversion for seabed properties.

The energy scattered at low grazing angles on 'smooth' seabeds is dominated by scattering mechanisms within the near surface sediment volume. Scattering measurements in this regime have been successfully modelled using a theoretical framework that attributes the scattering to inhomogenities that are represented physcially by variations in sediment porosity [4]. There is also emperical evidence using backscatter data from swath bathymetry systems on the Eel River and New Jersey ONR Strataform sites that the backscattered intensity in sand sediments decreases with increasing porosity [5]. On 'rough' seabeds, the scattering from the water-sediment interface is appreciable and degrades the correlation between porosity and backscattered energy.

This review of the literature suggests that *in situ* measurements of porosity may serve as an effective single parameter for characterizing seabed properties for both transmission loss and reverberation in sonar performance prediction. To this end, DRDC Atlantic is developing an *in situ* probe that can measure geotechnical (large strain) and geoacoustic (small strain) properties of the seabed, including porosity.

3. SEDIMENT PROBE

The DRDC Atlantic free fall cone penetrometer (FFCPT) test probe consists of a nose cone instrumented with geotechnical sensors, power supply, electronics, and tail pressure sensor (Fig. 1). As the probe penetrates into the seafloor, it measures acceleration and dynamic sediment porewater pressure as a function of depth. It also records hydrostatic pressure in the water and has an optical backscatter sensor for mudline detection capability. This combination of sensors permit the direct application of geotechnical analysis methods and parametric-based correlations already long established in engineering practice [6]. The DRDC Atlantic FFCPT has been developed in collaboration with Brooke Ocean Technology (BOT) Ltd. and Christian Situ Geoscience (CSG) Inc. (both in Dartmouth, Nova Scotia). It incorporates the basic sensor suite from an earlier 11.43 cm (4.5 inch) O.D. prototype into a modular 8.89 cm (3.5 inch) O.D. design (Fig. 1). Additional ballast or geoacoustic sensors can be integrated into the probe because of its modular design. The first module being developed measures resistivity as a means to determine porosity.

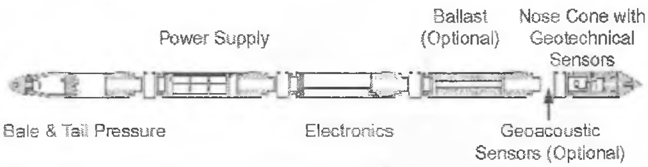

Figure 1: Schematic diagram of the DRDC Atlantic FFCPT.

The resistivity module has been developed by BOT and ConeTec (Vancouver, British Columbia) using the design principles of a static resistivity CPT system as described by Campanella and Weemees [7]. Conformal with the O.D. of the probe, there are two cylindrical brass electrodes, 1.5 cm wide, separated by 6.7 cm and isolated by sections of an insulating material—Delrin (Fig. 2). Once the probe has penetrated the seabed, a static resistivity measurement is made by generating a current-switched AC sinusoidal wave, at a frequency of about 1 kHz. Dynamic resistivity measurements during penetration [8] are ultimately desired, however, the high rate of initial penetration poses several technical challenges, such as an excitation rate of several hundred kHz, that have yet to be overcome.

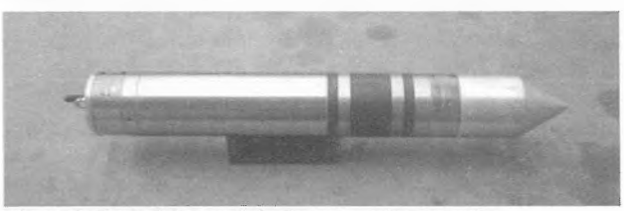

Figure 2: Resistivity module for measuring porosity

The measured bulk resistivity is a function of the resistivity of both the pore fluid and the sediment grains as well as the shape of the pore spaces. Assuming that the resistivity of the pore fluid is low, as with seawater, and there is not an abundance of clay minerals, then Archie's [9] law may be applied. It is $\rho_b/\rho_f = an^{-m}$, where ρ_b is the bulk resistivty, ρ_f is the fluid resistivity, a is a constant (usually 1 for unconsolidated sediments), n is the porosity, and m is a function of grain shape (~1.5 for sands).

4. RESULTS

The DRDC Atlantic FFCPT and resistvity module were deployed at a number of locations in St. Margaret's Bay, Nova Scotia, in June 2002. Some of the measurement locations are co-located with high quality sediment cores as well as drops from two other types of penetrometers that only measure acceleration. At the time of preparation of this manuscript, the data are still being analyzed. However, it is already clear, and encouraging, that a systematic increase in bulk resistvity has been observed as the test locations progressed from high to low porosity sediments.

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MATCHED BEAM PROCESSING SENSITIVITY TO ARRAY ELEMENT LOCALIZATION

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1. INTRODUCTION

Modern developments in underwater acoustics include the assessment of employing rapidly deployed, bottom moored vertical and/or horizontal hydrophone arrays for target detection and localization. While these arrays are economical in both cost and time for deployment, their autonomy presents a problem in effectively processing data recieved from a sensor which is not precisely located. Because these rapidly deployed systems (RDS) are deployed under tension from surface vessels and then lowered to the bottom, deployment geometry is vulnerable to the effects of wind and waves at the surface, as well as currents during array descent. The resulting uncertainty in the deployed array position can have a detrimental effect on subsequent processing of received acoustic signals.

This paper will discuss the array element localization (AEL) of a bottom-moored ultra-light (ULITE) horizontal array deployed during the RDS-2 trial. As well, the sensitivity of source localization to improper AEL will be demonstrated with a synthetic example of matched beam processing (MBP), an array processing technique which compares measured and calculated plane wave beams from a linear array to determine target position.

2. EXPERIMENT

In November, 1995, a multi-national trial was conducted to 'test and demonstrate advanced deployable array technologies and advanced data recovery methods and to test rapid array deployment techniques.' [1] The trial, called RDS-2, was conducted in the Timor Sea, approximately 160 km west of Darwin, Australia. Numerous arrays were deployed including an ultra-light (ULITE) Y-shaped horizontal array with three 468 m arms, each containing a nested configuration of 32 hydrophones.

Deployment of the ULITE array was carried out by three surface vessels, each paying out an arm under tension as they diverged from the central node position, and then lowering the fully extended array 107 m to the bottom. As is shown in Fig. 1, the intended and actual array positions differed significantly. Unequal cable tensions between the deploying ships as a result surface conditions, combined with strong currents resulted in individual hydrophones being in excess of 300 m from the planned positions.

After array deployment, an array element survey was conducted in which light bulbs were lowered from a surface vessel and imploded as sound sources at selected locations at an estimated depth of 52 m (Fig. 1). The light bult geometry was based on the intended array position. However, because of the disparity between intended and actual array positions, the source locations were less than ideal.

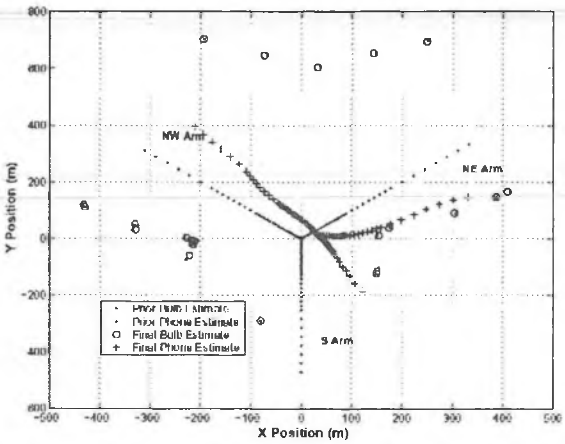

Fig. 1. Plan view of intended and recovered ULITE hydrophone positions. (Prior and final estimate of light bulb positions included)

3. ARRAY ELEMENT LOCALIZATION

Relative arrival times were measured for the light bulb implosion transients at each hydrophone by peak picking of direct path arrivals. The inverse problem of determining source and receiver positions from the arrival times is solved using the method of linearized inversion [2], [3]. *A priori* estimates are assigned to source and receiver positions in an iterative algorithm which seeks to minimize the modelled and real data misfit. Convergence criteria of the algorithm stipulate that the χ^2 misfit reduce to N (the number of equations generated for n transient arrivals at m hydrophones), and that the rms change of hydrophone positions between iterations be small in comparison to the expected solution accuracy. The inversion solution provides the best fit to the data, while explicitly minimizing array structure to only that which is resolvable from the acoustic information.

Hydrophone positions were located to within an average absolute rms error of 2.4 m horizontally, and 0.6 m vertically. Relative uncertainties were determined by a Monte Carlo appraisal [2],[4]. In the appraisal, the recovered position of the array is treated as the 'true' model to generate travel time data for simulated implosions at recovered source positions. Simulated data is then perturbed by adding random errors to create numerous data sets. Each set is inverted using the linearized algorithm and the resultant rms errors are averaged, providing average relative errors of 0.5 m horizontally and 0.6 m vertically. Fig. 3 depicts absolute and relative errors for each element.

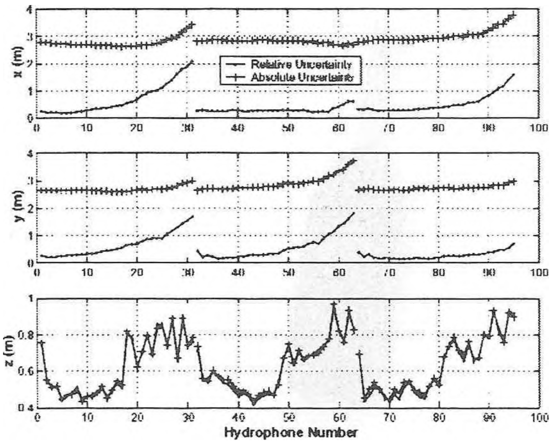

Fig. 2. Relative and absolute uncertainties (1 standard deviation) for ULITE element positions.

4. AEL IMPACT ON MBP

To demonstrate the impact of inaccurate AEL, a example is presented in which MBP is applied to simulated receptions from the recovered positions of the NE and NW ULITE arms. For the simulation, a 200 Hz source is located at 50 m depth, 80° (ref. true north) from the ULITE node, at a range of 3 km. Simulated acoustic data were generated using the ORCA normal mode propagation model [5], to which random noise was added resulting in a signal to noise ratio of 20 dB.

Fig. 3 depicts the effect of range, bearing, and depth correlations between the simulated *true* model (receiver positions are exact), and *estimated* model in which normally distributed errors of specified standard deviations have been added to the receiver positions. For the first run (solid line), *estimated* receiver positions are the same as *true* positions, thus correlations of 0.99 are achieved at correct bearing, range, and depth. Random horizontal errors drawn from a Gaussian distribution with standard deviation equal to that of the relative errors for the AEL inversion are added to the *estimated* receiver positions for the second run (dotted line). The correlation is reduced to 0.92, peaks remain at the correct bearing, range and depth. Doubling the standard deviation of the hydrophone perturbations begins producing range and depth estimation errors, and by the third run (dashed line) in which the standard deviation of induced errors is tripled (≤ 7 m), significant degradation is seen in both range and depth. The source is falsely located at range 2.75 km and depth 10 m. Finally, using the prior hydrophone positional estimates in the MBP precluded any meaningful localization in range, bearing, or depth (not shown).

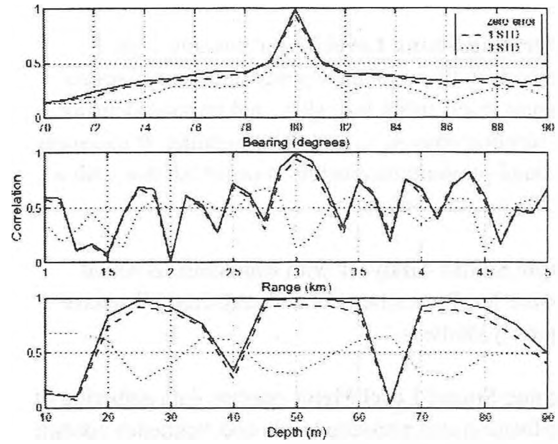

Fig. 3. MBP bearing, range, and depth correlations as a function of receiver positional error.

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AUTHOR NOTES

This work was conducted as part of the primary author's MSc thesis research in matched beam processing of ULITE acoustic data collected during RDS-2. AEL algorithms were developed and implemented by the second author.

BAYESIAN APPROACHES TO GEOACOUSTIC INVERSION

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1. INTRODUCTION

Great effort has been applied to estimating seabed geoacoustic properties using measured ocean acoustic fields. This amounts to an inverse problem: determine a model **m** given observed data **d**, where the model $\mathbf{m}=\{m_i, i=1,M\}$ represents the unknown geoacoustic and geometric parameters. In this paper two inversion algorithms are applied to ocean acoustic data. Adaptive simplex simulated annealing (ASSA) [1] is an optimization inversion algorithm that determines the model **m** that minimizes the objective function. While ASSA is an efficient nonlinear inversion algorithm for determining model parameter estimates, it does not provide rigorous parameter uncertainty estimates. A Bayesian sampling algorithm, the fast Gibbs sampler (FGS) [2], is also applied here. Through Bayesian inference, parameter estimates, parameter uncertainty estimates, as well as other information about the problem, can be determined.

To assess the abilities of ASSA and the FGS when applied to data from range-dependent environments, the algorithms were applied to synthetic benchmark data generated for the 2001 Inversion Techniques Workshop [3]. Some results using Test Case 1 (TC1) data for a shallow-water down-slope environment are presented here (see also [4] and [5]). An under-parameterized approach was applied to determine the optimal model parameterization for the environment.

2. BAYESIAN APPROACH

For the Bayesian approach to inverse problems [2], **d** and **m** are considered random variables. Bayes' rule states that the posterior probability density (PPD) $P(\mathbf{m}|\mathbf{d})$ is proportional to the likelihood function $L(\mathbf{d}|\mathbf{m})$ multiplied by the prior probability distribution of **m**, $P(\mathbf{m})$. The PPD embodies the general Bayesian solution to the inverse problem. Due to the PPD's multi-dimensionality, its properties can only be assessed indirectly, using, for example, the marginal probability densities, the posterior mean model, and the model covariance matrix. In addition, highest posterior density (HPD) intervals can be used to quantify parameter uncertainties. The smallest interval of each marginal density containing α% of the distribution's area defines the α% HPD. Also, the model that maximizes the PPD, the maximum *a posteriori* (MAP) solution, provides alternative parameter estimates. Under certain conditions, a sampler which samples a Gibbs distribution (a Gibbs sampler (GS)) can be used to estimate the PPD properties.

3. INVERSION METHODS

ASSA, one of the inversion methods used here, is a hybrid algorithm that combines the global and local inversion methods of simulated annealing and downhill simplex to search the parameter space for the optimal model. Each proposed model $\mathbf{m}'$ is assessed by evaluating the mismatch E between the measured fields **d** and modeled fields $\mathbf{d}(\mathbf{m}')$. The MAP is located when an objective function related to the PPD is used [5]. For a more complete solution, the FGS, which combines a GS with features that increase its efficiency, can be applied to determine the PPD properties.

The appropriate model parameterization is usually unknown in geoacoustic inverse problems. An under-parameterized approach is applied here to determine the appropriate number of sediment layers needed to represent the seabed. This approach includes repeatedly applying the inversion algorithm and increasing the number of layers each time. The solution model that minimizes both mismatch and structure will have the optimal parmeterization.

4. RESULTS

The results of applying the under-parameterized approach to TC1 using ASSA are shown in Figure 1. In Figure 1a, the mismatch E decreases significantly between L=1 and L=3, and then plateaus. Therefore, the model should include at least 3 sediment layers. As a measure of structure, the l_1 norm of variation between like parameters was calculated. In Figure 1b–d the variations of the sediment compressional

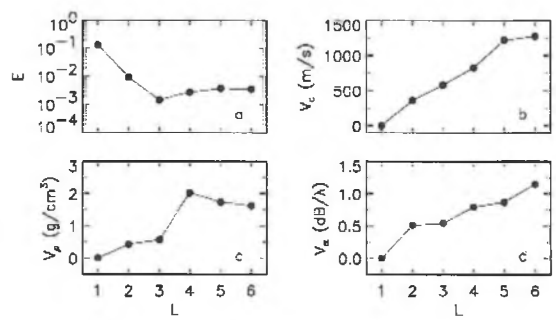

Figure 1. Under-parameterized approach to determining the appropriate number of sediment layers L using (a) the mismatch E and (b)–(d) the l_1 norm of variation for compressional speed V_c, density V_ρ, and attenuation V_α.

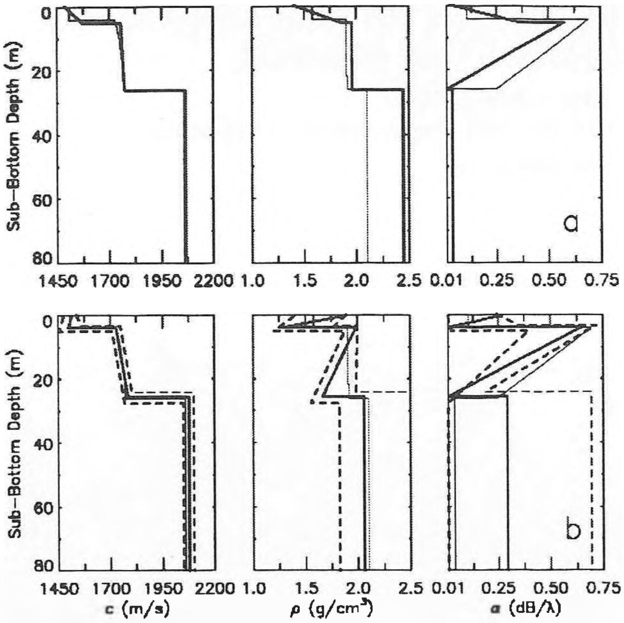

Figure 2. MAP profiles of compressional speed c, density ρ, and attenuation α using (a) ASSA and (b) the FGS. The thin line represents the true model and the thicker line represents the MAP estimates. The dashed lines represent the 95% HPD interval bounds.

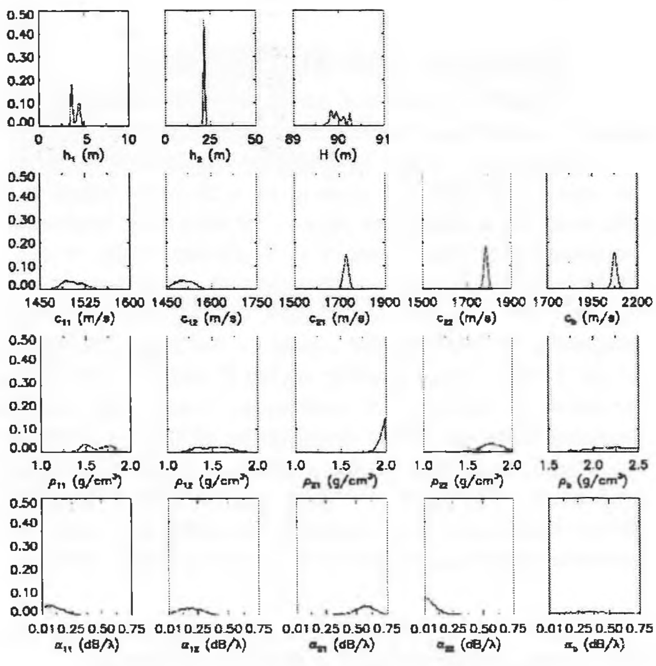

Figure 3. Estimated marginal probability densities for all parameters: layer thickness h, water depth at the source H, c, ρ, and α. For parameters x_i and y_{ij}, i represents layer (1, 2, or b (basement)) and j represents the top (1) or bottom (2) of the layer. The abscissa limits represent the bounds used in the inversion.

speed V_c, density V_ρ, and attenuation V_α typically increase with L. The model parameterization that minimizes the structure and the mismatch is, therefore, the L=3 parameterization.

Figure 2a shows the L=3 ASSA MAP estimate through parameter profiles. The true model is included for comparison. The compressional speed profile, including layer thickness, approximates the true profile extremely well. The density and attenuation profiles are also very well determined.

The estimated TC1 1-D marginal probability densities generated using the FGS for a three-layer model are shown in Figure 3. Most distributions are unimodal and symmetric. The layer thickness and compressional speed parameters have generally narrow distributions and are, therefore, well determined parameters. Density and attenuation parameters are not as well determined.

Figure 2b shows the parameter profiles for the FGS MAP estimate and the true model. The FGS MAP solution is a very good estimate of the true solution. Also included in this figure are the schematic representations of the 95% HPD intervals used to quantify the parameter uncertainties.

5. CONCLUSIONS

ASSA and the FGS were successfully applied to the range-dependent benchmark data. The appropriate number of sediment layers and good parameter estimates were determined using the under-parameterized approach and ASSA. While not as efficient as ASSA, the FGS provides uncertainty bounds which are crucial for assessing the quality of the final estimate.

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THE EFFECTS OF AZIMUTHAL VARIABILITY ON MONOSTATIC ACOUSTIC BACKSCATTER FROM THE SEABED

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1 INTRODUCTION

By measuring the effects of azimuthal variability on monostatic acoustic backscatter from the seabed, the accuracy of sonar performance prediction (SPP) models and their detection ranges can be improved. SPP models require a number of inputs including bottom loss, bottom backscatter, surface loss, surface backscatter, ambient noise level and the sound speed profile of the water column. The sea surface loss, sea surface backscatter and the ambient noise level can all be predicted easily using the wind speed [1, 2, 3], whereas the sound speed profile is generally measured using a bathythermograph. Conversely, estimating the seabed properties required for SPP models has remained difficult.

Defence Research and Development Canada (DRDC) – Atlantic (formerly DREA) has addressed these issues by developing a wide band sonar (WBS), to quantify the geoacoustic properties of the seabed. The WBS consists of a parametric transmitter and a superdirective hydrophone line array. Due to the nature of the signal generation of the parametric transmitter, no side lobes are formed and the beam width of the difference frequency is only 3°. These properties, prevent interference from other undesired boundary returns thereby enabling accurate measurements of the azimuthal variability of the backscattering strength.

2 EXPERIMENTAL METHODOLOGY

In June 2002, the WBS was used to measure the azimuthal variability of the backscattering strength. The system collected acoustic backscatter data as a function of azimuth and grazing angle at three shallow water sites - two sites near Sable Island and one site in St. Margaret's Bay, Nova Scotia. The experiments were performed at grazing angles ranging from 7° to 15° and frequencies of 2, 4 and 8 kHz. The experimental geometry is shown in Fig. 1. During the monostatic backscattering measurements the parametric array head was held at a fixed grazing angle while a series of 50 pings were transmitted at a pulse repetition frequency of 4 pings/s. The measurements were repeated in 4° azimuthal increments through 360°. This procedure was repeated at a number of grazing angles.

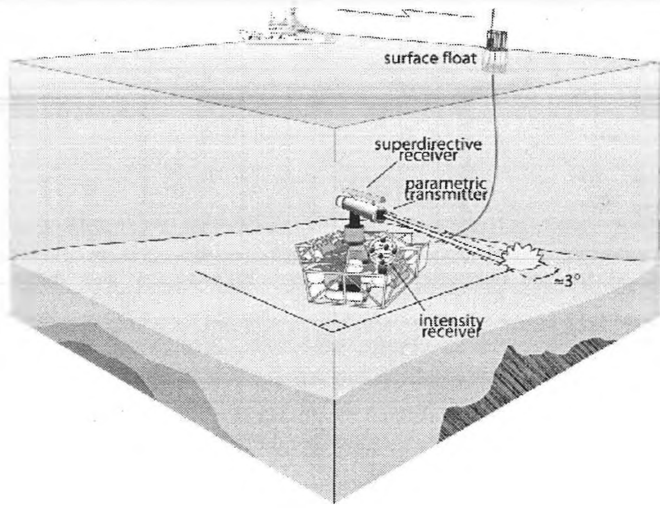

Figure 1 Geometry for backscatter azimuthal variability experiments.

3 RESULTS AND DISCUSSION

Figure 2 shows a color contour image of the azimuthal dependence of the backscattered energy vs. range as measured on a single hydrophone on the superdirective line array. The data are from a site near Sable Island at 8 kHz along 90 different azimuths. The data were taken with the parametric array pointed at a grazing angle of 8.5°. References for ranges corresponding to grazing angles 5°, 8°, 10°, 20° and 30° are shown on Fig 2 and 3. The monostatic return from the center of the beam (yellowish coloured ring between grazing angles 8° and 10°) dominates the energy contour. The monostatic return shows rich structure in the azimuthal dependence of the data as there are many singularities in the seabed structure both inside and outside of the center of the beam. The first 5 ms of the energy time series are discarded because the data are contaminated by signals arising from the structural platform.

From the received levels ($RL(\theta_g)$) of the monostatic arrival, the scattering strength ($BSS(\theta_g)$) may be determined,

$$BSS(\theta_g) = RL(\theta_g) + 40\log[R(\theta_g)] - 10\log(\tau) - 10\log[dA(\phi,\tau)] - BP(\theta_g) - G_{pa}[f, R(\theta_g)] \quad [1]$$

where $R(\theta_g)$ is the one way distance of the monostatic path; τ is the pulse length; dA is the area ensonofied by the 3° beam width ϕ and pulse length τ, $BP(\theta_g)$ is the beam pattern of the source; and $G_{pa}[f,R(\theta_g)]$ is the parametric array gain. The energy data from Fig. 2 were converted to backscattering strength using [1] and are shown in Fig. 3.

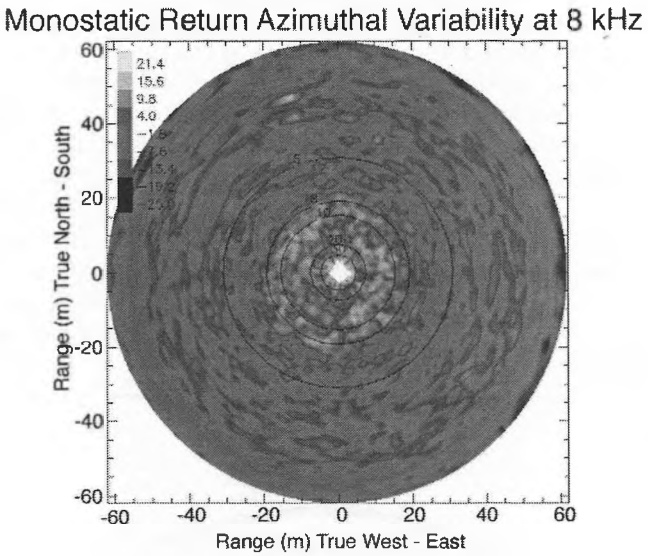

***Figure 2** – Color contour images displaying the azimuthal variability of the energy time series*

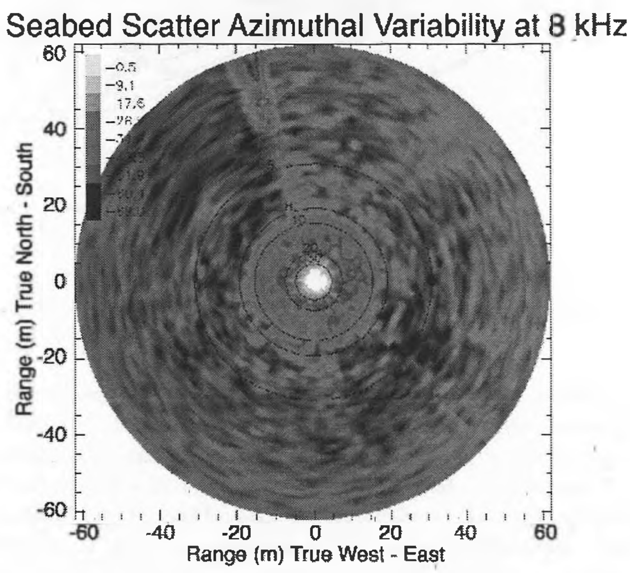

***Figure 3** – Color contour images displaying the azimuthal variability of the seabed scattering strength*

Figure 3 shows considerable variability in the scattering strength of the seabed. At shallow angles, near the center of the beam and further ranges, the monostatic return is calculated using returns obtained solely from the ocean floor and may be used to calculate scattering strength of the seabed provided that there is sufficient signal-to-noise. This experiment was performed at 2, 4 and 8 kHz and at grazing angles of 8°, 12° and 15° to increase the accuracy by extending the range over which the monostatic return is close to the center of the beam. These results (not shown) also have substantial azimuthal variability of seabed backscatter near the center of the beam. Note, at steeper angles, away from the center of the beam, scattering strengths are higher in part due to a larger scattering strength and also due to some ringing of the initial pulse in the frame of the WBS.

4 Conclusion

Accurate measurements of the azimuthal variability of the backscattering strength were obtained during DRDC – Atlantic sea trial Q267 in June 2002. Monostatic backscattering strength measurements were performed as a function of azimuth. The monostatic return shows rich structure in the azimuthal dependence of the data as there are many singularities in the seabed structure both inside and outside of the center of the beam. When converted to scattering strength, the monostatic return shows considerable variability as a function of azimuth. The backscattering strength measurements appear to be independent of frequency within the statistical accuracy of the data.

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BISTATIC SCATTERING AT SHALLOW GRAZING ANGLES

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1. INTRODUCTION

In recent years there has been a growing interest in bistatic sonar systems. That is to say, systems for which the source and receiver are not co-located. Measurements of acoustic scatter at bistatic angles is required for performance modeling of these systems. DRDC Atlantic has developed a pair of sea going research systems for measuring bistatic scatter from the seabed in shallow water environments. The first system the Wide Band Sonar (WBS) is a bottom mounted parametric transmitter with a 6 channel superdirective line array receiver. The second system the Underwater Acoustic Target consists of a vertical line array of 8 hydrophones and 2 transmitters [1]. These two systems have been used to make preliminary measurements of low-angle bistatic scattering from the seabed. The bistatic geometry substantially complicates both the collection and the interpretation of these measurements. For these reasons, a computer simulation was developed to compute the pathlengths and the arrival times for the various bistatic arrivals. The simulation is used to help select experimental geometries and to interpret the results in terms of the bistatic arrivals. In this paper the experimental geometry is described and a sample of the data is presented and compared to the results of the numerical simulation.

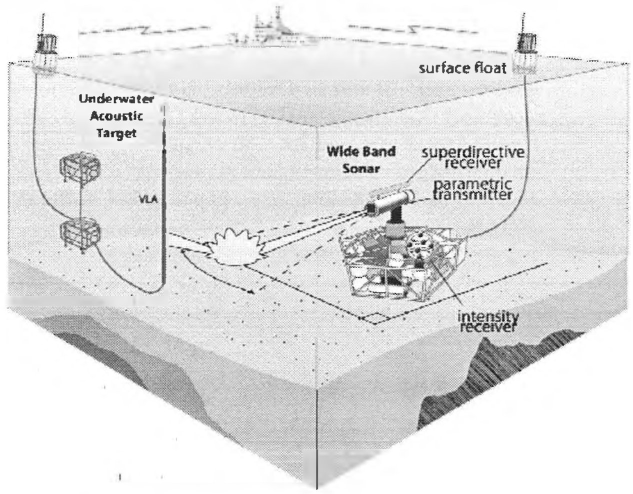

Figure 1: Experimental geometry.

2. RESULTS AND DISCUSSION

Figure 1 contains an illustration of the experimental geometry. The UAT hydrophones were positioned at 4.2 m spacings from approximately 10 m to 40 m above the seabed. The parametric array was 2.7 m above the seabed and was pointed at a grazing angle such that the specular reflection insonified approximately the vertical center of the UAT receive array. This corresponded to grazing angles ranging from 9° 2° depending on the separation of the two systems, which in turn ranged from 160 m up to 900 m. At each azimuth a series of 50 pulses, 2 ms long was transmitted by the WBS at 2, 4 and 8 kHz and the bistatic scatter was recorded on the UAT. The parametric array transmitter was rotated 2° in azimuth and the sequence was repeated. This was done from 15° to +15° relative azimuth.

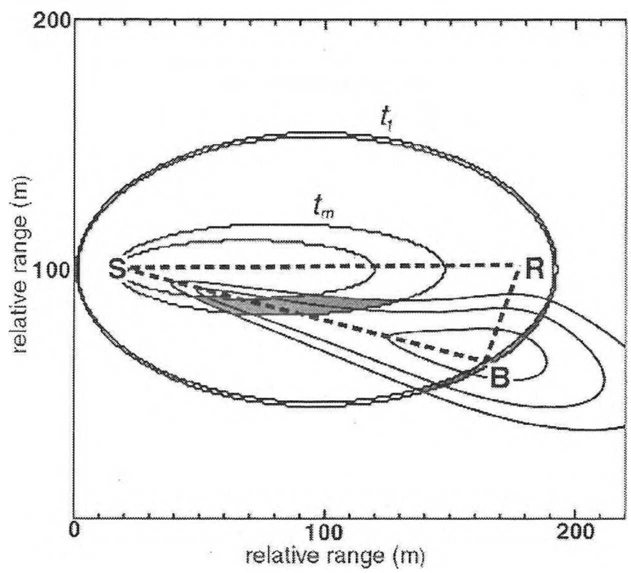

Figure 2: Simulation for a 2 ms pulse and a relative azimuth of 15°.

Figure 2 shows a plan view of the simulation results for a grazing angle of 9°, a relative azimuth of 15° and a separation of 164 m. The normalized contours (plotted in 3 dB increments from 3 dB down to 9 dB) represent the energy received at the hydrophone and include the effect of the parametric array beam pattern and the geometric spreading from source–to scattering point–to receiver. The annulus in the figure labeled t_m corresponds to the time coincident arrivals from the water-seabed interface that occur at the energy maximum of the received signal. It's thickness is proportional to the pulselength and the grazing angle. Note that the sharp focusing of the parametric array and the absence of sidelobes results in a reasonably small, well-defined footprint for the scattering patch rather than the entire annulus. This simplifies the interpretation of the bistatic scattering angles considerably. Note that the annulus is not elliptical and its thickness varies around the circumference because the source and receiver are at different heights above the bottom. For comparison purposes, a second annulus labeled t_1 is shown that corresponds to time coincident arrivals occurring at 20 ms. As the range increases the annulus begins to approximate an ellipse and at long ranges is tends toward a circle. Note that including the effects of the parametric array beam pattern and the geometric spreading, results in the asymmetric shape of the contours and draws the maximum of the bistatic arrival (for example, path SBR for annulus t_1) in toward specular path SR reducing the mean azimuthal angle from 15° to 12°.

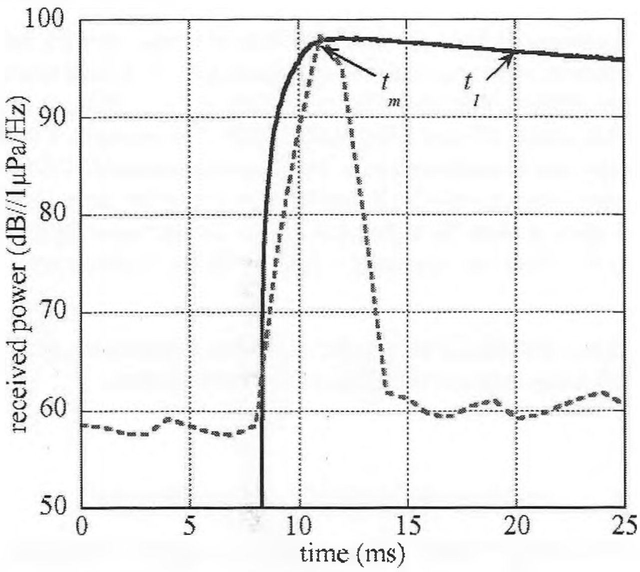

Figure 3: Simulation-data comparison of received energy for a 2 ms pulse duration at 4 kHz, a relative azimuth of 15°, and 9° grazing. Note that the level of the simulation curve has been offset to align it with the peak in the data.

Figure 3 compares the received power at a hydrophone located approximately 23 m above the seabed (dashed line) to the results from the simulation (solid line). The simulation is run assuming that each point on the seabed acts as a perfect reflector from the source to the receiver. This explains the slow energy decay in the simulation. The difference between the simulation and the data can then be used to obtain the scattering strength of the seabed. The times marked t_m and t_1 correspond to the arrivals denoted by the annuli in Figure 2.

Figure 4 shows simulations of the relative energy levels vs. arrival times for azimuthal angles of 0°, 10°, and 24°(dashed lines). The remaining parameters are as for Figures 2 and 3. The delay in the onset of the peak as azimuth increases results from the increased bistatic path length. Comparing the arrival times predicted at various azimuths with the corresponding data will ensure that the specular path from the edge of the source beam is not dominating the received energy. That is to say, if the onset time is not changing with azimuth it implies that the received energy is dominated by contributions along the specular path (recall path SR in Figure 2) rather than along the bistatic path.

Returning to Figure 4, the change in arrival time with angle shown in the simulation is relatively small because the source array is so close to the seabed. Increasing the height of the source to 23 m (the same height as the receiver) increases the time separation. This is shown by the solid lines in the figure.

Comparing the arrival times for the experimental data requires time synchronization between the data streams on the transmit and receive systems. This was achieved during a sea trial in June 2002. Unfortunately, analysis of the data was incomplete at the time of publication of this paper.

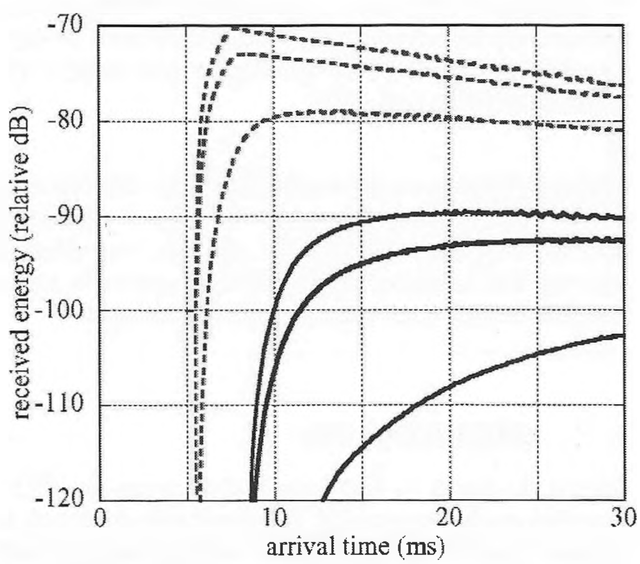

Figure 4: Relative energy vs. arrival times for a range of azimuthal angles for a 2 ms pulse.

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ELIMINATING DIVISION OPERATION IN NLMS ALGORITHM

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1. INTRODUCTION

The normalized least mean square (NLMS) adaptation algorithm is widely used in acoustic and network echo cancellation, noise cancellation, channel equalization, system identification, and so on. In each iteration of a conventional NLMS implementation, a division operation is required to update a variable called the step size. Since a division consumes much more real-time than a multiplication or addition does in a typical digital signal processor (DSP), a significant portion of the precious processing power is spent on the division operations if there is a large number of them present in the algorithm. Furthermore, a conventional NLMS algorithm does not respond to sudden increases in input level promptly enough in order for an echo canceller to meet today's stringent requirements, such as [1].

This paper proposes a patent-pending alternative† where the division operation is avoided so that the amount of computation for NLMS is reduced. In addition, the proposed approach can accelerate the algorithm's response to sudden increases in input level when properly implemented.

2. BACKGROUND

Beyond the scope of this paper, details about the NLMS algorithm can be found in [2]. This paper only deals with the calculation of the step size therein, which is denoted by $\mu(n)$ at sampling interval n. The conventional NLMS finds $\mu(n)$ by using

$$\mu(n) = \beta/\langle x^2(n)\rangle \tag{1}$$

† The author was with Nortel Networks when the present approach was conceived. Nortel Networks retains ownership of intellectual property rights relating to this article and its subject matter.

where β is a positive constant, $x(n)$ is the input sample at n, and <·> is the operator for a weighted time average over a certain number of past samples of the argument. Typically, $\langle x^2(n)\rangle$ is an estimate of the energy in $x(k)$ ($k = n, n\text{-}1, n\text{-}2, \ldots$) over a certain number of most recent samples.

Equation (1) indicates that a division is needed in each sampling interval n in order to calculate $\mu(n)$. It is well-known that division operations are quite expensive, in terms of real-time usage, on most commercial DSPs. For example, it takes only one instruction cycle for Texas Instruments' C54x, a typical commercial DSP family, to do a multiplication, while it takes at least 34 instruction cycles for the same processor to do a basic one-quadrant, 32-bit by 16-bit, division [3].

Thus, minimizing the number of division operations can significantly improve the efficiency of an algorithm.

3. THE PROPOSED APPROACH

The concept behind finding a quotient without performing a division is, in each sampling interval, to compare the numerator with the product between the denominator and a quotient estimate, then to adjust the latter accordingly. The tracking error can be negligibly small if the true quotient varies slowly over time, as is the case with the NLMS algorithm; the numerator in Eq. (1) is a constant and the denominator, being a time-average, is slowly time-varying.

In each sampling interval n, the proposed approach starts with $\mu(n-1)$, an estimate of the step size used by the last sampling interval, compares β with the product of $\mu(n-1)$ and $x^2(n)$, and updates $\mu(n-1)$ accordingly to arrive at $\mu(n)$, step size estimate to be used by the current sampling interval. Being a first-order closed loop feedback system, the approach is illustrated in Figure 1.

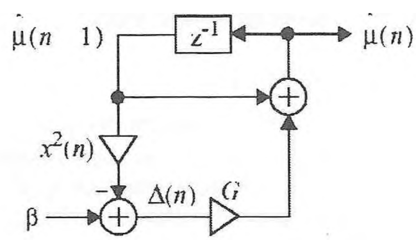

Figure 1. Flow diagram of the proposed approach

The equivalent analytical form is given by

$$\begin{aligned} \hat{\mu}(n) &= \hat{\mu}(n-1) + G\Delta(n) \\ \Delta(n) &= \beta - \hat{\mu}(n-1)\,x^2(n) \end{aligned} \qquad (1)$$

where $\Delta(n)$ in Eq. (1) is the result of the comparison, and G is a positive factor that controls the rate of the adjustment.

To show that Eq. (1) gives an approximation to Eq. (1), take expectations of Eq. (1) while considering the fact that G is small so that $\hat{\mu}(n-1)$ changes slowly. The result is

$$E[\hat{\mu}(n)] = \{1 - GE[x^2(n)]\}E[\hat{\mu}(n-1)] + G\beta \qquad (2)$$

Since

$$GE[x^2(n)] \ll 1 \qquad (3)$$

holds, Eq. (2) converges so that $E[\hat{\mu}(n-1)] \approx E[\hat{\mu}(n)]$. Equation (2) can then be solved as

$$E[\hat{\mu}(n)] \approx \beta/E[x^2(n)] \qquad (4)$$

which, under the assumption that $x(n)$ is ergodic, is equivalent to Eq. (1).

Note that $<x^2(n)>$, as needed in Eq. (1), is not calculated explicitly here. Instead of a time-averaged version of it, only a single sample $x^2(n)$, with a much larger fluctuation, is used in each iteration. In fact since Eq. (3) holds, $\hat{\mu}(n)$ fluctuates much less than $x^2(n)$ does; therefore, as an integration of the differences $\Delta(n)$ over time, it reflects the impact of $<x^2(n)>$ implicitly. This means that the proposed approach saves computation not only by eliminating the division operation, but also by not calculating the time-average $<x^2(n)>$.

In practice, the real-time it takes for a typical DSP, such as the C54x [3], to perform such a "pseudo-division" as Eq. (1) is only about one fifth of what it takes to do a real division.

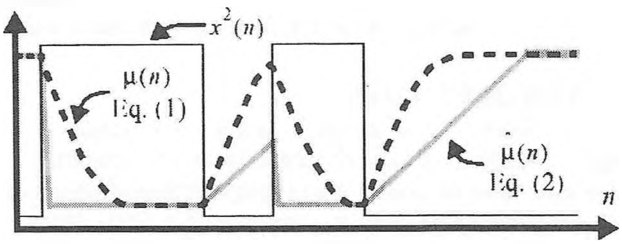

Figure 2. Behaviors of conventional, by Eq. (1), and proposed, by Eq. (1), approaches

Figure 2 illustrates that the proposed approach, given by Eq. (1), responds to sudden increases in input signal level much faster than the conventional approach, by Eq. (1), does. There are test cases required by [1] that incorporate such fluctuations, which can easily cause the conventional NLMS to diverge momentarily because of its large response time. On the contrary, an NLMS algorithm featuring the proposed approach has been proven to survive such fluctuations well. Figure 2 also shows that the proposed approach has a longer ramp-up time when the input signal level drops. This is usually not considered an issue, because it only slows down the convergence when the signal level has dropped, and never causes any concerns for divergence.

1. SUMMARY

A simple and easy to implement way of avoiding the division operation in the widely used NLMS algorithm has been studied. In addition to simplifying the implementation, the proposed approach responds to the input signal dynamics in a manner in favour of avoiding potential system divergence.

The concept in the proposed approach can be applied not only to the NLMS but also to other algorithms where a division is needed and the quotient to be estimated does not change quickly over time.

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THE AGGREGATE BEAMFORMER

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1. INTRODUCTION

Beamforming is used to achieve directional response using a spatially distributed receiver. Antennae have been used for analog electromagnetic beamforming in radio, radar, and microwave communication. Early digital beamformers (such as DIMUS) were developed for the detection of submarines. More advance techniques such as adaptive and synthetic aperture beamforming have now become textbook material.

These conventional beamforming techniques rely on simultaneous (or, at least, systematic) sampling of the individual array elements (channels). Here, we discuss a method of beamforming which collects each sample from a randomly selected array element. The samples collected from all channels this way are then treated together to recover the desired signal; hence the term 'aggregate beamformer'.

Whereas the conventional beamformer cancels the off-beam signals by destructive interference, the aggregate beamformer converts these undesired signals into white noise. The approach gives the same beam response pattern as conventional beamforming but exhibits an additional residual noise. Since the overall sampling rate of the aggregate beamformer exceeds the Nyquist rate for the desired signal, low-pass filtering and decimation is used to reduce the noise. The noise level can be reduced to any desirable level by adjusting the overall sampling rate.

The principle advantages of the aggregate beamformer are its ability to function without stringent anti-alias filters, it scales so that arrays with many sensor elements (hundreds or thousands) can be used without a significant increase in complexity or other component count, it provides improved beam steering resolution, it can be designed using the same principles as conventional beamformers, it has no arithmetic operations (except for the final decimation filter), and the reduced hardware requirements makes solid state integration easier.

2. CONVENTIONAL BEAMFORMING

For simplicity, we compare the aggregate beamformer here only with the conventional delay and sum beamforming (DS), although more sophisticate beamformers are readily implemented. The DS beamformer applies a delay to the signal received at each element and computes the sum of the delayed samples. The beamformer B_{ds} is

$$B_{\mathrm{ds}}(n) = \sum_i X_i(n-d_i) \qquad (1)$$

where i is the sensor index, W_i is a channel weight, $X_i(n) = X_0(n-t_i)$ is the sampled signal, $X_0(n)$ is the source signal as observed at the array centre, t_i is the signal propagation time difference between the array centre and sensor i, and d_i is the (integer) sample delay (approximating t_i).

The sample delay quantization reduces the beamformer performance (directivity) and limits the number of distinct beams that can be formed (steering resolution). Typically, the sampling rate is increased or a digital interpolation filter is used to minimize these problems.

Numerous channels may be multiplexed to a high speed analog to digital converter (ADC) but a separate analog anti-alias filter is needed for each channel (unless separate delta-sigma ADC are used on each channel.)

3. AGGREGATE BEAMFORMING - SIMPLE BROADSIDE CASE

Consider a linear array receiving a far-field signal arriving perpendicular to the array (broadside). The wavefront arrives at all sensors at the same time so the DS beamformer output signal is simply,

$$B_{\mathrm{ds}}(n) = (1/M)\sum_{i=0}^{M-1} X_i(n). \qquad (2)$$

If there is no noise or interference then the X_i are all identical and so one could equally well choose any value for i and define $B(n) = X_i(n)$ as the output! – however, this would not remove undesired signals arriving from other (off-broadside) directions.

Notice, however, that if the sensor index i was changed randomly so that for sample n we choose $i = s(n)$ where $s(n)$ is a random number distributed uniformly over the set of sensor indices, then we get

$$B_{\mathrm{abf}}(n) = X_{s(n)}(n) \qquad (3)$$

which preserves a signal arriving from broadside but introduces random sampling delays that corrupt signals arriving from other directions. We call this the broadside 'aggregate beamformer' (ABF). It is easily implemented in a multiplexed ADC configuration by applying random addresses to the multiplexer.

The corrupted output (for off-broadside signals) have reduced amplitude and added noise. It turns out that the noise is white Gaussian noise and the amplitude is *exactly* the same as that which the DS beamformer would produce. For a signal arriving broadside (on-beam) the total sampling rate of the ABF need only be the Nyquist rate of the signal: this is $1/M$ the total sampling rate of the conventional DS beamformer.

The residual noise of the aggregate beamformer is reduced by increasing the sampling rate by some 'over-sampling factor' beyond the Nyquist rate. Since the residual noise is uniformly distributed over the entire output bandwidth, as in Fig 1, a lowpass filter (followed by decimation) will substantially reduce the noise power without affecting the signal. If the over-sampling factor is M then the residual noise for an off-beam signal will be.

$$\|N_{\text{res}}\| \quad \frac{1}{M}(\|X_0\|+\|B_{\text{ds}}\|) \tag{4}$$

A signal arriving broadside (on-beam) will have no residual noise.

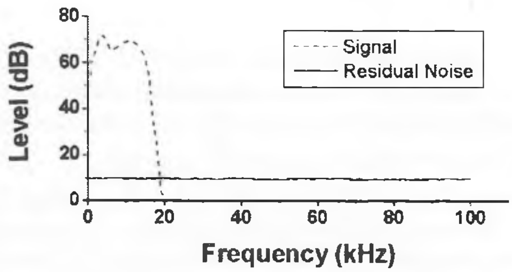

Fig. 1. Spectrum of signal and residual noise. Lowpass filtering reduces the residual noise power

THE GENERAL CASE

The aggregate beamformer can be steered to any beam direction using sample delays d^{θ}_{i} computed as for the conventional beamformer for a specific direction θ. Since the aggregate beamformer has a higher sampling rate the sample delays can be determined with greater accuracy and there is no need for sample interpolation.

Just as for the broadside case, successive samples n are taken from a randomly selected channel but a delay is applied by placing them into the aggregate beamformer output sequence at index $m = n + d^{\theta}_{s(n)}$.

$$B^{\theta}_{\text{abf}}(n + d^{\theta}_{s(n)}) = X_{s(n)}(n). \tag{5}$$

Because of the random time delays $d^{\theta}_{s(n)}$, this process may assign more than one sample n to the same output index m (causing a *collision*) and it may leave some output indices without any data assigned (causing *voids*). Statistical analysis shows that collisions/voids occur about 35% of the time (except for the broadside case). This increases the residual noise level of the aggregate beamformer only slightly.

Handling collisions and voids is an important aspect of implementing the aggregate beamformer but is beyond the scope of this presentation. One approach to handling colllisions is to randomly choose one of the colliding samples to keep. Another approach is to compute the average of colliding samples but this negates one of the advantages of the aggregate beamformer - it requires no arithmetic operations (except for the decimation filtering.) A simple and effective way to handle voids is to replicate the previous data value.

Figure 2 (top) compares the aggregate beamformer to the conventional DS beamformer. The residual noise power (lower frame) is essentially constant off-beam and is substantially reduced at the on-beam direction. (Optimization techniques beyond the scope of this presentation have been used for this example.)

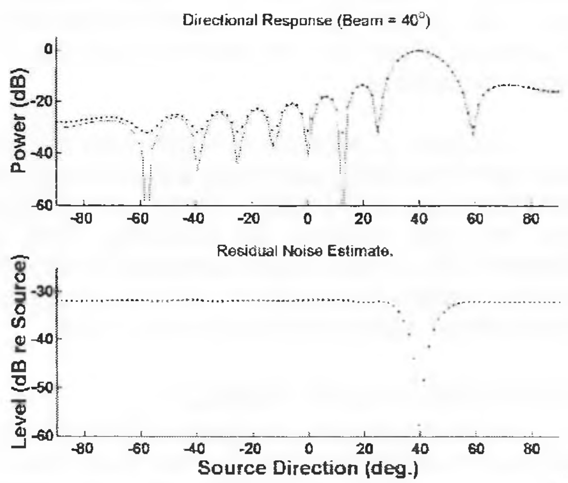

Fig. 2. Directional response (top) and residual noise (bottom) of aggregate beamformer (dot) steered to 40°. A 64-element square array is used with M=256 over sampling factor. The corresponding directional response for a conventional DS beamformer (top, solid) is shown for comparison.

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FREE EDGE CONDITION ON POROUS MATERIALS

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1. INTRODUCTION

The free edge condition is encountered when the surface or contour of an elastic domain interacts freely with the surrounding acoustic domain. For an air-saturated open-cell porous layer, this condition poses some problems since the acoustic impedance of the surrounding domain is of the same order of magnitude than the one of the porous domain. In this case, the fluid loading of the surrounding domain on the porous layer cannot be neglected.

In the mixed Biot *u-p* poroelastic formulation [1], the free edge condition is classically modeled by imposing the acoustic pressure on the free surface of the porous sample to zero [2]. This implementation of the free edge condition does not seem to accurately model reality as the acoustic pressure will vary on the free surface. In this paper, a different implementation of the free edge condition on an air-saturated open-cell porous layer is proposed. It consists in applying an impedance radiation condition on the free surface of the sample. This new implementation of the free edge condition is also derived under the mixed Biot *u-p* poroelastic formulation.

The paper is structured as follow. First, the finite element model used in this research is detailed. Second, the different hypotheses and the finite element implementation of the free edge condition are discussed. Third, the experimental setup used to validate the numerical simulation is presented. Finally, the comparison between the numerical and experimental results are presented and discussed.

2. FINITE ELEMENT MODEL

As stated previously, the poroelastic finite element model used in this paper is based on the mixed Biot *u-p* formulation. To reduce computation time and memory usage and to enhance the convergence rate, both an axisymmetric formulation and high order Lagrange polynomial shape functions are used [3]. Therefore, similarly to the axisymmetric version of the displacement Biot *u-U* formulation [4], a transformation from cartesian to cylindrical coordinates is applied to the original *u-p* formulation, followed by an integration along the θ-axis. This yields the following weak integral equation, $\forall(\delta u_i, \delta p)$:

$$\begin{aligned}
&2\pi\int_{\Omega}\left[\hat{\sigma}_{ij}^{s}(u)\varepsilon_{ij}^{s}(\delta u)-\omega^{2}\tilde{\rho}_{s}u_i\delta u_i\right]rdrdz \\
&+2\pi\int_{\Omega}\left[\frac{\phi^{2}}{\omega^{2}\tilde{\rho}_{22}}p_{,i}\delta p_{,i}-\frac{\phi^{2}}{\tilde{R}}p\delta p\right]rdrdz \\
&-2\pi\int_{\Omega}\tilde{\gamma}\delta\left(p_{,i}u_i\right)rdrd\,z\underbrace{-2\pi\int_{\Sigma}\hat{\sigma}_{ij}(u)n_j\delta u_i\,dS}_{I_{\Sigma}^{S}}, \quad (1) \\
&\underbrace{+2\pi\int_{\Sigma}\left[\tilde{\gamma}u_i n_i-\frac{\phi^{2}}{\omega^{2}\tilde{\rho}_{22}}p_{,i}n_i\right]\delta p dS}_{I_{\Sigma}^{f}}=0
\end{aligned}$$

where Ω and Σ are the volume and surface of the porous aggregate, respectively. dS is an elementary surface on Σ, u is the solid phase displacement field, and p the fluid phase acoustic pressure field of the porous aggregate. I_{Σ}^{S} and I_{Σ}^{f} are respectively the surface integrals for the solid and fluid phases. The other parameters are detailed in references [1,5]. This formulation allows for the modeling of a 3D axisymmetric geometry with a 2D meshing, with (r, z) and (ξ, η) the global and local coordinates of the 2D meshing.

3. FREE EDGE IMPLEMENTATION

The following relations need to be applied on the free surface of the porous sample in order to model the free edge condition:

$$\begin{aligned}
&p = P_{rad} \\
&P_{rad} = -j\omega z_o\left[(1-\phi)u_n+\phi U_n\right] \quad . \quad (2) \\
&\hat{\sigma}_{ij}^{s}(u)n_j^{P} = -(1-\phi)P_{rad}n_i^{P}+\phi\frac{\tilde{Q}}{\tilde{R}}pn_i^{P}
\end{aligned}$$

The first relation indicates that the acoustic pressure on the surface is equal to the radiated pressure in an infinite acoustic medium. A good approximation of this radiated pressure is to consider normal radiation on the free surface, as it is done in the second relation. Finally, the last relation shows that the *in vacuo* stress tensor is the sum of the total stress on the solid phase, $-(1-\phi)P_{rad}n_i^{P}$, and the stress on the fluid phase, $\phi(\tilde{Q}/\tilde{R})pn_i^{P}$.

Substituting the first and third relation of eq. (2) in I_{Σ}^{S} and considering that $\tilde{Q}/\tilde{R} = (1-\phi)/\phi$ yields:

$$I_{\Sigma}^{S} = 0 . \qquad \textbf{(3)}$$

Similarly, substituting the second relation in I_{Σ}^{f} and considering that $\phi(1+\tilde{Q}/\tilde{R}) = 1$ gives:

$$I_{\Sigma}^{f} = \frac{2\pi R}{j\omega z_o} \int_{\Sigma} p\delta p dz . \qquad \textbf{(4)}$$

4. EXPERIMENTAL SETUP

To compare the measurements with the numerical results, disk shape samples need to be used so that the entire setup can be considered as being symmetric along its central axis. Figure 1 illustrates this setup. The porous sample is bonded in between two large rigid plates. There is a hole in the middle of the second plate from where a normally incident plane wave, generated by the standing wave tube, can excite the sample. Also, the circumference of the sample can interact freely with the surrounding acoustic domain. The sample as a diameter of 99 mm and a thickness of 76 mm. The material used for the measurements is a elastic foam whose properties are shown in Table 1.

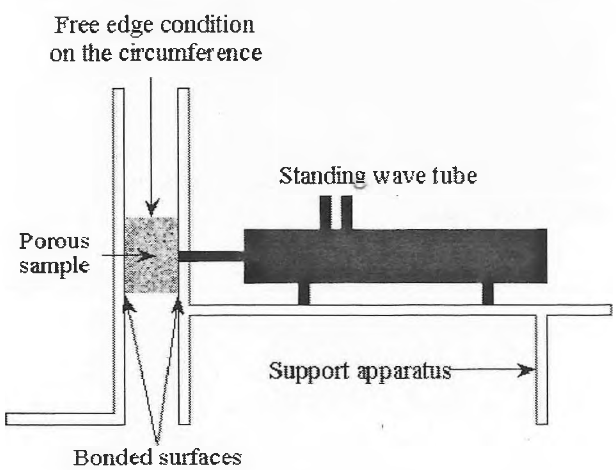

Figure 1. Experimental setup where a disk shape porous sample is excited by a normally incident plane wave. The circumference of the sample can interact freely with the surrounding acoustic domain.

5. RESULTS AND DISCUSSION

In Figure 2, the measurements and the numerical results for the elastic foam are compared. As it can be seen, there is a relatively good agreement between the two. This shows the accuracy and effectiveness of the modeling. Similar results have been obtained for other materials that are relatively isotropic. For certain materials, the agreement is not as good as shown previously, but this could be explained by the presence of a certain anisotropy.

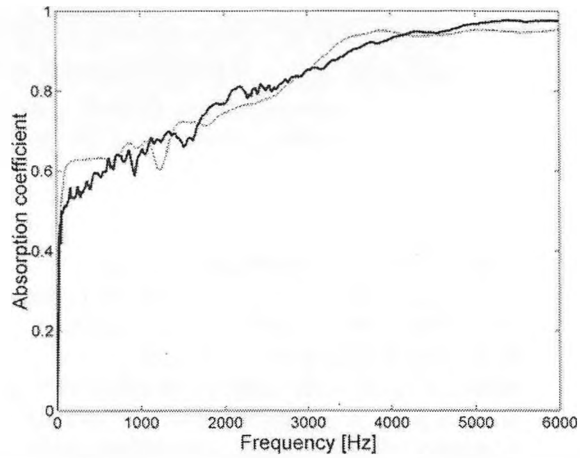

Figure 2. Absorption coefficient vs. frequency. —: measurements with the special setup, - - -: numerical results with the impedance condition on the circumference.

Table 1 - Material properties for the elastic foam

Property	Unit	Value
E	N/m^2	93 348
σ	Ns/m^4	12 569
ρ_1	Kg/m^3	8.9
ν		0.44
ϕ		0.99
α_∞		1
Λ	μm	56
Λ'	μm	319
η_S		0.06

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THE NEW CSA Z94-2 STANDARD HEARING PROTECTION DEVICES – PERFORMANCE, SELECTION, CARE AND USE

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This Standard was published in January 2002 It was prepared by the CSA Technical Committee on Hearing Protection Devices after two years of hard work. It is supposed to be a document balanced between the needs of manufacturers and of users alike. It is expected that health and safety personnel will find a lot of useful information that will help them in their everyday dealings.

The first five sections of the Standard are dedicated to issues common in all similar CSA Standards. Their titles are: Scope, Definitions, Reference Publications, Materials and Requirements.

Section 6, Test Procedures, specifies that tests should be performed as per ANSI Standard S12.6, although test results as per ANSI Standard S3.19 are also accepted. The test procedure to be used is the Method B, that is a Real-ear attenuation at threshold (REAT) procedure, where protectors are fit by the subjects that are persons not familiar with the use of protectors (naïve subjects). Results from the test are reported as octave band data. Those data are used to compute the Single Number Rating (Subject Fit 84 th Percentile), abbreviated $SNR(SF_{84})$ that provides a nominal 84% protection confidence interval (i.e., 84% of the users in a well-run hearing conservation program are expected to receive at least that much protection). The procedure for the calculation of $SNR(SF_{84})$ is in Appendix A of the Standard. Using the $SNR(SF_{84})$ data, a Grade will be assigned to the protector, as per Table 2. If, however, data from measurements using the procedure in ANSI Standard S3.19 are used, then and attenuation class will be assigned as per Table 3 in the Standard.
The Force Measurement of the headband is compulsory for muffs. However, the Physical Performance Tests described in the Standard are optional.
Section 7, Packaging Information is of especial interest to manufacturers.
The following three sections: Section 8, Selection, Care and Use: Overview and General Requirements, Section 9, Selection of Hearing Protection Devices and Section 10, Fit, Care and Use of Hearing Protection Devices are most important for users and health and safety professionals, since they deal with issues fundamental for an efficient use of protectors in the workplace.
This information is complemented with Section 11, Implementation, that is a guide on how to introduce protectors in the workplace in such a way that they are used efficiently.
Sections 8 through 11 include information such as sound attenuation, attenuation at frequency extremes, double protection, overprotection, etc. Their content will be helpful to anyone writing Hearing Protection Program in the workplace.
The last Section 12, Specialized Hearing Protection Devices, is a guide into the non-conventional types of hearing protectors, their characteristics and applications. It includes active protection devices (ANR headsets), protectors with linear attenuation, sound restoration, frequency sensitive, etc.
The Standard, 49 pages long, is available from the Canadian Standard Association whose web site iswww.csa.ca

THE CURRENT STATUS OF INTERNATIONAL STANDARDS FOR SOUND LEVEL METERS

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1. INTRODUCTION

Sound level meters are the most widely-used instruments for acoustical measurements. Most of the well-known International Standards for sound level meters, integrating sound level meters and dose meters [1 to 6] have been in place for over 20 years. In that period, advances in technology have not only transformed the appearance and capabilities of traditional stand-alone sound level meters, but also given rise to new instruments that incorporate personal computers or telecommunications links. Changes in regulatory regimes, the demands of international trade and experience of designing and testing meters to the specifications of the various Standards have also shown up limitations in the old specifications [1 to 6] that must be addressed.

A good description of the changes in the newly published IEC 61672-1 specifications for sound level meters [7] with respect to the current standards can be found in [8]. IEC 61672 will eventually be published in three parts; Part 1, which contains the performance specifications, was published in May 2002. The technical work on Part 2, which specifies the tests required for pattern evaluation (type testing) of new models of sound level meter, is complete and the document will be circulated as a Final Draft International Standard for voting by national committees. Part 3, which specifies the periodic tests (calibration) for sound level meters, is being drafted.

IEC 60651 [1] and IEC 60804 [3] will not be withdrawn until all three parts of IEC 61672 are published. The status of the existing Standard for personal exposure and dose meters [5, 6], is unaffected by the introduction of IEC 61672. The current ANSI S1.4 American National Standard Specifications for sound level meters (1983) (R 2002) and its amendment S1.4A (1985) (R 2002) have been reaffirmed recently [2]. At this stage, the publication of IEC 61672-1 is unlikely to have any real impact on sound level meter usage in North America.

2. CHANGES IN IEC 61672-1

2.1 Performance

The previous Standards for sound level meters classified the performance of instruments into Types 0, 1, 2 and 3 in decreasing order of accuracy. IEC 61672-1 replaces these Type designations with performance classes 1 and 2, whose requirements are broadly equivalent to those for the old Types 1 and 2. However, some notable changes have been made to the specifications to reflect technological advances and to improve international compatibility.

The specifications for directional response now permit meters that are designed to measure random incidence sound (diffuse sound field), as well as those designed to measure sound from one direction in a free field.

Frequency weighting A is mandatory for all sound level meters, frequency weighting C is mandatory for class 1 meters, and frequency weighting B has been eliminated. Although impulse time weighting I is no longer recommended for use, IEC 61672-1 contains an informative Annex that provides recommendations for the specification and testing of time weighting I, should it be implemented in a meter.

The specifications for response to transient signals have been made more stringent with the introduction of specified responses to tone-bursts of durations as short as 0.25 ms, compared to the minimum duration of 5 ms of previous sound level meter specifications.

The minimum level range for meters with digital displays has been increased to 60 dB from 15 dB.

IEC 60651 was amended in 2000 to include specifications for immunity from radio frequency interference, and these amendments have been incorporated into the new Standard.

2.2 Uncertainties

Uncertainties of measurements have been a major concern of international standardization. Normative Annex A in IEC 61672-1 lists the maximum measurement expanded uncertainties, which must be calculated according to the ISO/IEC Guide to expression of uncertainty in measurement (1995). When testing a meter's conformance to the new Standard, the calibration laboratory's measurement uncertainties must now be taken into account and must not exceed the maximum permitted uncertainties for each test that are specified by IEC 61672-1. As a result, the budget for tolerance limits available to manufacturers for the design of a sound level meter is decreased by the maximum permitted uncertainty of the laboratory that tests the meter.

3. IMPLICATIONS

3.1 Measurement capabilities

The design goals for the various frequency weightings specified by IEC 61672-1 remain the same as those of the previous Standards. However, the tolerance limits around these design goals have been re-evaluated. In general the tolerances of IEC 61672-1 have improved. For class 2 instruments, the tolerances for the weightings at 10 kHz are + 5.6 dB and minus infinity. As in the existing standards, this minus infinity tolerance permits the manufacturer to employ a low capability microphone that is unable to response to signals at 10 kHz or higher. For the measurement of impulsive sounds, such an instrument will give relatively large erroneous high frequency level readings. Similarly, for class 1 instruments, the tolerances for the weightings at 16 kHz and 20 kHz are + 3.5 dB and – 17 dB, and + 4 dB and minus infinity, respectively. The tolerances that include minus infinity allow a microphone that is unable to respond to above 16 kHz to be used in a class 1 instrument. The users of all sound level meters should take note on these limitations.

The current ANSI Type 1 and Type 2 specifications [2] have similar limitations for the measurement of impulsive sound as discussed above for class 1 and class 2 instruments. The electrical response of ANSI Type 1 instruments has tolerances for 16 kHz and 20 kHz of + 3 dB and – 7.4 dB, and + 3 dB and – 8.7 dB, respectively. By not having a minus infinity tolerance at 20 kHz, ANSI Type 1 instruments are able to respond to electrical signals at 20 kHz. Also, IEC 61672-1 has eliminated the Type 0 instruments that the ANSI sound level meter standard retains [2]. Type 0 instruments that have tighter tolerances are very useful for precise measurements and calibration laboratories. In North America, the hearing aid industry uses the ANSI sound level meter standard.

3.2 Periodic Calibration

IEC 61672 Part 3 is being drafted with the aim of producing a set of simplified tests for an individual instrument that can be completed by a calibration laboratory with an automated system in less than half a day. By concentrating on areas where existing designs are known to have difficulty in meeting specifications, the new tests are also more likely to detect shortcomings in the performance of an instrument. The drafting experts for the document are faced with two difficult questions: (a) How to determine whether an instrument has been submitted by the manufacturer for pattern evaluation? And (b) if the instrument has not been type tested, how to select the necessary tests to ensure conformance at a reasonable cost? With the above insight, and before the purchase of a new sound level meter, it is sensible for the user to ask whether the instrument has been submitted for type testing. The answer to this question may eliminate the doubt whether the claims stated in the specifications for the instrument are true.

3.3 National Standards

The Canadian Standards Association (CSA) has endorsed the ANSI sound level meter standard [2]. Some Canadian organizations prefer to quote the existing IEC sound level meter standards [1, 3]. The question on the preference for 3 dB or 5 dB exchange rates for sound exposure and noise dosimeters in North America [5, 6] is still to be resolved.

4. CONCLUSIONS

International consensus has led to the publication of a new specification standard for sound level meters [7]. Until all three parts of IEC 61672 are published, the existing standards [1, 3] are still acceptable for current use.

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HUMAN RESPONSE TO VIBRATION AND MECHANICAL SHOCK

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1. INTRODUCTION

Human exposure to vibration, and mechanical shock, is commonly experienced in daily life, for example, in trains, cars, ships and buildings. All of these situations involve motion of the whole body transmitted through a seat, or from the floor in the case of a standing person, where the human response is commonly related to the sensation of motion or the relative motion of body parts. Exposures also occur in occupations involving operation of hand-held power tools, primarily to the hand and arms. The vibration, and shocks, become of consequence when comfort or activities are influenced (e.g., motion sickness), or health is threatened (e.g., back injury, or damage to the blood vessels and nerves of the hand).

Exposures are quantified by measurement procedures described in standards prepared by the International Organization for Standardization (ISO). The metrics to employ in some situations remain the subject of research, and form the subject of a companion paper.[1] The CSA Subcommittee on Human Response to Vibration has participated in, and its members have made important contributions to, the development of international standards in this field for over twenty years. The concepts underlying much of this work are summarized in this paper, and follow the description in a recent publication.[2] Many of these ISO standards are now being proposed for Canadian standards, a process similar to that already underway in the U.S.A.

2. DEFINITIONS AND METRICS[2]

Vibration is a time-varying disturbance of a mechanical, or biological, system from an equilibrium condition for which the long-term average of the motion will tend to zero, and on which may be superimposed either translations or rotations, or both. A mechanical *shock* is a non-periodic disturbance characterized by suddenness and severity with, for the human body, the maximum forces being reached within a few tenths of a second, and a total duration of up to about a second. When considering injury potential, the size and shape of the object in contact with, or impacting, the body is important, as is the posture. In addition, for hand tools, both the compressive (grip) and thrust (feed) forces employed to perform the manual task need to be considered.

The magnitude of vibration is characterized by second, and higher, even-order mean values. For an acceleration that varies with time, t, as $a(t)$, the higher-order mean values are calculated from:

$$a_{RM} = \left[\frac{1}{T} \int_0^T [a(t)]^m \, dt \right]^{1/r} \tag{1}$$

where the integration is performed for a time T, and m and r are numerical constants. By far the most common metric of the magnitude of whole-body or hand-transmitted vibration is the *root mean square* (RMS) acceleration a_{RMS}, which is obtained from eqn. 1 with $m = r = 2$, and is used for the assessment of perception, and discomfort. Other metrics include the *root mean quad* (RMQ) acceleration a_{RMQ}, with $m = r = 4$. The relationship between these metrics depends on the amplitude distribution of the acceleration-time history, and provides insight into the waveform.

In contrast, health disturbances are considered to be related to the exposure, which is constructed from the magnitude of the stimulus, its frequency content and duration:

$$E(a_w, T)_{m,r} = \left[\int_0^T [F(a_w(t))]^m \, dt \right]^{1/r} \tag{2}$$

where $E(a_w,T)_{m,r}$ is the exposure occurring during a time T to a stimulus function that has been frequency weighted in an attempt to equate the hazard at different frequencies, $F(a_w(t))$. In general, $F(a_w(t))$ may be expected to be a non-linear function of the frequency-weighted acceleration, $a_w(t)$. A commonly used function is the so-called *energy-equivalent* vibration exposure for which $F(a_w(t)) = a_w(t)$ and $m = r = 2$. For an exposure continuing throughout a workday, $T \rightarrow T_{(8)} = 28\,800$ s, and eqn. 2 can be written:

$$T_{(8)}^{1/2} \left[\frac{1}{T_{(8)}} \int_0^{T_{(8)}} [F(a_w(t))]^2 \, dt \right]^{1/2} = T_{(8)}^{1/2} a_{w,RMS(8)} \tag{3}$$

where $a_{w,RMS(8)}$ is the 8-hour, energy-equivalent, frequency-weighted, RMS acceleration. A second function, used for exposure to whole-body vibration, is the *vibration dose value*, *VDV*, for which $F(a_w(t)) = a_w(t)$ and $m = r = 4$. The function is thus more influenced by the large amplitudes in a fluctuating vibration than the energy-equivalent exposure.

The output of a single (uniaxial) sensor infrequently characterizes an exposure to vibration. This necessitates the measurement of orthogonal component accelerations (indicated by subscripts X, Y, and Z). The *overall vibration value* is then expressed by the frequency-weighted, RMS, vector acceleration sum, a_{WAS}, using values of the frequency-weighted, RMS, component accelerations, i.e.:

$$a_{WAS} = \left[a_{wX,RMS}^{\ \ 2} + a_{wY,RMS}^{\ \ 2} + a_{wZ,RMS}^{\ \ 2}\right]^{1/2} \quad (4)$$

It should be noted that the frequency weighting employed differs for the X, Y, and Z directions for whole body vibration, but is the same for hand-transmitted vibration (subscript 'h'). Exposures to the latter are described in terms of the magnitude of the 8-hour, energy-equivalent, frequency-weighted, RMS, vector acceleration sum, which is constructed from values of $a_{hX,RMS(8)}$, etc, using eqn. 4.

3. MEASUREMENT ISSUES

Non-contact methods are, in principle, preferred for measuring the motion of soft tissues, but are not considered in current standards. In consequence, measurements are specified at the interface between the skin and a source of vibration, such as a vehicle seat pan or tool handle, and involve the use of specialized mounts for the transducer(s). In some circumstances the vibration of a mechanical structure in contact with the body is recorded using accelerometer(s) rigidly attached to the seat or tool handle. Triaxial accelerometers are available for measurement of component accelerations in three orthogonal directions. The orientation of sensors is prescribed. The use of piezoelectric accelerometers to measure shocks requires the transducer to be mounted on a mechanical low-pass filter, to reduce errors resulting from internal crystalline changes introduced by motion at the sensor's resonance frequency.

4. SUMMARY OF HUMAN RESPONSES

Estimates of exposures and vibration magnitudes necessary for selected human responses are summarized in Table 1. Included in the Table are the metric employed, the source of information, and a representative value for the response, or health effect. The perception of vibration depends on the stimulus frequency and body site. Note that the "whole body" response is specified in terms of a frequency-weighted component acceleration. Adverse response to building vibration (e.g., from underground railway trains) occurs at close to the threshold of perception, and employs a combined frequency weighting for all directions. The discomfort experienced in passenger transportation is largely associated with interference with activities such as reading or drinking, and is related to the magnitude of the stimulus. Vertical vibration at low frequencies (0.1 - 0.5 Hz) may result in motion sickness. The metric employed is an energy-equivalent exposure in which the stimulus function is expressed as a frequency-weighted acceleration in eqn. 2. The value cited is for a 10 % prevalence of vomiting in the general population (men and women). Thresholds for the onset of health effects are given for regular, near-daily exposure, and are expressed in terms of the daily dose. The metrics, however, differ. For hand-transmitted vibration the onset of the hand-arm vibration syndrome (HAVS) is expressed in terms of the 8-hour frequency-weighted acceleration sum, while for whole-body vibration the onset of back injury is estimated in terms of the VDV. The latter may also be specified in terms of an energy-equivalent exposure. The risk of injury from vertical (headward) shocks is described by the response of a non-linear biodynamic model. The model employs a neural network to produce an estimate of the spinal response. This forms the input to an exposure metric that sums the number of shocks using the acceleration peak value, i.e., $F(a_w(t)) = a_{peak}$, with $m = r = 6$ in eqn. 2. An equivalent, daily, static compressive stress is then calculated to assess the risk of injury.

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The definitions in section 2 are from reference 2, and are reproduced here with permission.

Table 1: Estimates of typical human responses to vibration, and mechanical shock, with sources (after Ref. 2)

Human Response	Source	Metric	Typical Value
Perception (mean)			
whole body	ISO 2631-1, 1997	$a_{wZ,RMS}$	0.015 ms^{-2}
fingertips	ISO/FDIS 13091-2, 2002	a_{RMS}	0.0075 ms^{-2} (4 Hz) M - 0.25, F - 0.32 ms^{-2} (125 Hz)
Building vibration	ISO 2631-2, 1989	$a_{wC,RMS}$	0.007 ms^{-2}
Discomfort (transportation)			
not uncomfortable	ISO 2631-1, 1997	a_{WAS}	< 0.315 ms^{-2}
uncomfortable	ISO 2631-1, 1997	a_{WAS}	0.8 – 1.6 ms^{-2}
very uncomfortable	ISO 2631-1, 1997	a_{WAS}	> 2.0 ms^{-2}
Motion sickness (10 %)	ISO 2631-1, 1997	$E(a_{wZ},T)_{2,2}$	30 $ms^{-1.5}$
Health effects (onset)			
hand (HAVS)	ISO 5349-1, 2001	$a_{h,WAS(8)}$	1.0 ms^{-2}
whole body	ISO 2631-1, 1997	VDV	8.5 $ms^{-1.75}$
shocks (vertical)	ISO/DIS 2631-5, 2002	$E(\Sigma a_{peak})_{6,6}$ in spine → static compression	> 0.5 MPa

TEMPORARY CHANGES IN VIBROTACTILE PERCEPTION DURING OPERATION OF IMPACT POWER TOOLS: A PRELIMINARY REPORT

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1. INTRODUCTION

The measurement and evaluation of human exposure to hand-transmitted vibration have been codified in an international standard, where exposures are characterized by daily, eight-hour, energy-averaged, frequency-weighted, RMS component accelerations, $a_{w,RMS(8)}$.[1] The exposure metric assumes: 1) a relationship between the relative hazard presented by vibration at different frequencies, which is introduced by frequency weighting the acceleration-time history, $a_w(t)$, and; 2) temporal summation by means of energy averaging throughout the duration of a work day (specified as eight hours), $T_{(8)}$, as described in Ref. 2, i.e., for each (uniaxial) component:

$$a_{w,RMS(8)} = \left[\frac{1}{T_{(8)}} \int_0^{T_{(8)}} a_w(t)^2 dt \right]^{1/2} \tag{1}$$

The metric is broadly accepted for predicting the chronic vascular disturbances associated with the hand-arm vibration syndrome (HAVS) from exposure to near-continuous vibration, but its applicability to exposures involving repeated mechanical shocks, such as during operation of impact power tools (e.g., pneumatic chippers and hammers) has been repeatedly questioned.[3] While some information on the suitability of the metric can be obtained from the results of epidemiological studies of working populations, the results of studies of acute responses to vibration exposure provide another source of information.

A laboratory experiment has been devised for determining changes in vibrotactile perception threshold (VPT) at the fingertip in response to a mechanical stimulus consisting either of continuous vibration or a repeated, transient, damped sinusoidal shock, with the same carrier frequency. The repetition rate of the transient stimulus is chosen to be characteristic of common impact power tools, and to fall within the range of frequencies at which the VPT is known to be mediated by a single mechanoreceptor population.[4] The stimuli are applied to a handle gripped by a subject with a constant static force. The magnitudes of the stimuli are adjusted to yield equal exposures according to eqn. 1, i.e., equal values of $a_{w,RMS(8)}$.

2. METHOD

2.1 Vibration stimulus

Exponentially decaying sinusoidal waveforms were generated digitally to simulate power tool acceleration-time histories. The algorithm contained the decay rate, the sinusoidal carrier frequency, and the repetition rate of the "impacts". Combinations of these parameters were selected to be comparable with real tools and with the stimuli employed in a previous laboratory experiment.[5] A single waveform decay rate, which corresponds approximately to that observed on bucking bars (a device held behind the work piece during riveting), was employed.

The repetition rates for the waveforms were 4, 16, and 32 per second. The rate of 4 shocks per second is consistent with that produced by pneumatic nailers, and the rate of 16 per second with that produced by riveters, while the rate of 32 shocks per second is in the range produced by scalers and needle guns (typically 30-40 impacts per second). The sinusoidal carrier frequency was 125 Hz, which also served as a non-shock stimulus.

The waveforms were applied to an electro-dynamic vibration exciter (Bruel & Kjaer 4805, with 4811 exciter head), after reducing the low-frequency roll-off in its electromechanical transfer function. The uniaxial stimuli were coupled to the hand through a rigid handle. The hand gripped a tube with an elevated strip mounted along its length, to which was attached a strain gauge bridge for monitoring grip force. An accelerometer was attached to the inside of the tube at its mid point, to record the vibration at a surface in contact with the hand. The frequency-weighted, RMS component acceleration was monitored and was adjusted to provide a constant value of $a_{w,RMS(8)}$, as each exposure was of the same duration.

An example of an acceleration waveform recorded at the inside of the tube during an experiment is shown in Fig. 1. It is evident that the desired stimulus acceleration-time history has been achieved for this the most difficult case, involving a 4 Hz repetition rate. As this waveform was recorded under the palm it may be taken to reflect the stimulus experienced by the skin in contact with the handle.

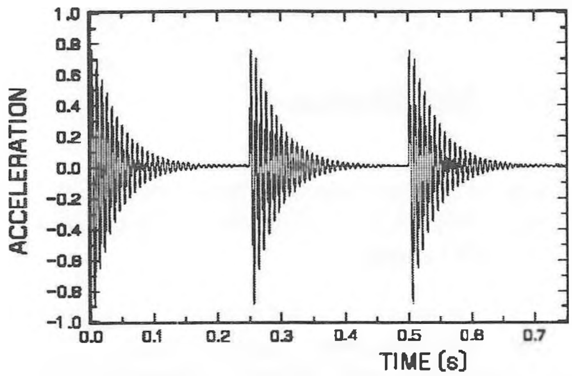

Figure 1: Shock waveform stimulus with 4 per second.

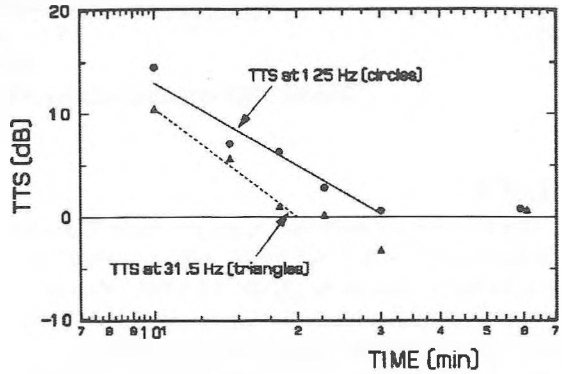

Figure 2: TTS after exposure to continuous vibration for subject #1.

2.2 Determination of vibrotactile thresholds
Vibrotactile thresholds were determined at the fingertip using the tactometer, in a manner analogous to the determination of the threshold of hearing by an audiometer.[6] The tactometer consists of a vibration stimulator and sensor with a probe attached to contact the skin that are lowered onto a fingertip, and a means to support the hand and arm with the palm facing upwards, together with signal conditioning circuits under computer control.

Thresholds were determined sequentially at a fingertip, at three frequencies (4, 31.5, and 125 Hz), commencing as soon as possible after completion of an exposure. Measurements were conducted repeatedly, at predetermined time intervals. The un-adapted threshold was determined before commencing an exposure, and served as the basis for calculating any temporary threshold shift (TTS) occurring as a result of the exposure. A control experiment was performed in which the subject gripped the handle for the duration of the exposures (8 min), but with no vibration stimulus. Fingertip skin temperatures were also recorded.

3. RESULTS AND DISCUSSION

Results are considered here for three subjects and two repetition rates - 4, and 32 Hz, and for continuous vibration. The frequency-weighted, RMS acceleration of each uniaxial stimulus was 2.0 ms^{-2}.

A stimulus-response function for VPTs at 32 and 125 Hz is shown for one subject in Fig. 2, where the responses are to the continuous stimulus at 125 Hz. In this diagram the TTS has been plotted as a function of the time elapsed from determining the initial pre-exposure VPT. The results are the average of three exposures to each stimulus, the order of presentation having been randomised.

It can be seen that in this case the recovery follows a logarithmic time dependence after the termination of exposure (at a time of 9 min), reaching the un-adapted threshold recorded prior to the stimulus (i.e., TTS = 0) after about 20 min for TTS at 125 Hz, and 10 min for TTS at 31.5 Hz. There was no TTS at 4 Hz recorded by this subject. Substantial differences in response were observed between subjects, both in the magnitude of the TTS and in the form and rate of the recovery function. These will be described elsewhere, as well as results from other subjects.

The mean maximum TTS recorded by the three subjects is presented in Table 1, and leads to several observations. Firstly, there appears to be, on average, little or no TTS at 4 Hz in response to any of these stimuli. Secondly, the TTS at 31.5 Hz seems to be, on average, little affected by the nature of the stimuli in contrast to that at 125 Hz, which is substantially less for the slower shock rate. The pattern of these results suggests that the primary influence on TTS is related to the frequency-weighting function used to form $a_w(t)$, rather than to the procedure for summing shocks.

4. REFERENCES

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The authors wish to acknowledge the contribution of Ms. H. Rowland to the processing and analysis of the data. Work supported by NIOSH under research grant RO1 OH04025-02.

Table 1: Mean TTS of three subjects for stimuli with the same energy-averaged, frequency-weighted, RMS acceleration

Stimulus	Mean maximum TTS (dB)		
	4 Hz	31.5 Hz	125 Hz
shock, 4 repetitions per second, 125 Hz carrier	1.7	6.7	8.7
shock, 32 repetitions per second, 125 Hz carrier	-2.7	5.8	14
Continuous, 125 Hz	0	4.7	15

Review of Activity in CSA Standards Activity in Industrial Noise

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ABSTRACT

This paper reviews recent activities in the Canadian Standards Association Industrial Noise Subcommittee, the most active acoustics standards group in Canada. Recent activities have included a new voluntary standard on Noise Emission Declarations, an acoustics section in the recent Office Ergonomics standard and ongoing work in adapting ISO standards on community noise measurement to Canada. In addition, there is a joint working group with ANSI looking at a North American approach to the ISO industrial noise propagation standard. Future opportunities, such as in-plant noise predictions, will be discussed. Recent publications and ongoing activity will be summarised and opportunities for wider involvement of the acoustics community across Canada encouraged.

1. INTRODUCTION

The Industrial Noise Subcommittee is the most varied and active subcommittee reporting to the CSA Acoustics and Noise Control Committee.

Ongoing activities include:

2. GUIDELINES FOR THE DECLARATION OF MACHINERY NOISE EMISSION LEVELS

Guidelines For The Declaration Of Machinery Noise Emission Levels will be a voluntary guide[1] for noise labelling of machinery for use in Canada and compatible with European regulations to allow machinery to be sold into that market.

Labels in this context refer to a statement of sound levels produced by the equipment which would be included with the instruction or maintenance manual. Measurements are made according to ISO standards and include estimates of the likely variability of the measurements.

At the time of writing this standard is out for ballot as CSA Z107.58.

3. ADOPTION OF ISO1996

A working group chaired by Chris Krajewski[2] and including several Ontario consultants is examining using ISO 1996 as a way of updating the way tonal and impulse sounds are handled in community noise. They are currently running round robin tests of the procedures with various sample sounds. Stephen Keith of Health Canada is acting as liaison with the ISO committee.

The first round of tests was reported at the last Canadian Acoustics conference[3] and a third round is ongoing. The standard is written to be compatible with a number of different regulations in Europe. A first draft of an informative annex relating the standard to the Canadian context has been prepared.

4. ISO9613 / CSA Z107.55 NOISE PROPAGATION

This is a joint study group with ANSI looking at adopting or endorsing ISO 9613-2, the ISO standard on prediction of industrial noise in the community. The ISO standard was written by an international group chaired by Joe Piercy of NRC. It would either replace or update CSA Z107.55.

The US group have decided to adopt the standard with minimal changes. Once their version is ready, we will meet separately and come up with a Canadian position. To some extent, this is more important in Canada than in the US because several provinces such as Ontario routinely predict the impact of new and changing industries on the community as part of their approval process.

5. BLASTING NOISE AND VIBRATION

There is an existing standard on Blasting Noise and Vibration measurement in the community which is referred to in Ontario MOE guidelines. It needs updating and we are looking for a working group chair to take this on.

6. OFFICE ACOUSTICS

The Office Ergonomics Standard was recently republished with a completely rewritten acoustics section by a working group from the Industrial Noise Subcommittee. It includes design goals, recommended treatments and was written to be accessible to a lay audience.

7. NOISE EXPOSURE STANDARDS Z107.56

This standard, first published in 1986 has proven to be the most popular acoustics standard in Canada and is referred to in legislation and accepted across the country. Stuart Eaton and Alberto Behar are updating the standard presently.

It will specifically address concerns about spurious impulses due to the microphone cable rubbing on clothing and other concerns recently raised in BC.

8. INDUSTRIAL HALL PREDICTION

This group agreed that a new revision was long overdue but are looking for funds to allow it to proceed.

9. PARTICIPATION

The Industrial Noise Subcommittee and its working groups is an active part of the Canadian Acoustics Standards scene. Those interested in joining are invited to participate by contacting the author.

10. STANDARDS

- Z107.51-M1980 (R1994) Procedure for In-Situ Measurement of Noise from Industrial Equipment
- Z107.52-M1983 (R1994) Recommended Practice for the Prediction of Sound Pressure Levels in Large Rooms Containing Sound Sources
- Z107.53-M1982 (R1994) Procedure for Performing a Survey of Sound Due to Industrial, Institutional, or Commercial Activities
- CAN3-Z107.54-M85 (R1993) Procedure for Measurement of Sound and Vibration Due to Blasting Operations
  Méthode de mesure du niveau sonore et des vibrations émanant des opérations de dynamitage
- CAN/CSA-Z107.55-M86 Recommended Practice for the Prediction of Sound Levels Received at a Distance from an Industrial Plant
  Pratique recommandée pour la prévision des niveaux sonores reçus à une distance donnée d'une usine
- Z107.56-94 Procedures for the Measurement of Occupational Noise Exposure
  Méthode de mesure de l'exposition au bruit en milieux de travail
- Z107.58 Noise Emission Declarations for Machinery – in Ballot

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STANDARDIZATION ACTIVITIES AND CANADIAN PARTICIPATION IN ISO TC43 'ACOUSTICS' AND ISO TC43/SC1 'NOISE'

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1. INTRODUCTION

The foremost aim of international standardization is to facilitate the exchange of goods and services through the elimination of technical barriers to trade. The scope of Technical Committee (TC) 43 of the International Organization for Standardization (ISO) includes methods of measuring acoustical phenomena, and all aspects of their effects on humans and the environment. Specification of acoustic measuring instruments and their calibration procedures are the responsibilities of the International Electrotechnical Commission (IEC) TC 29. ISO TC 43 maintains a close liaison with IEC TC 29.

European Union (EU) directives concerning the safety of machinery, workers and the environment are establishing the priorities of ISO TC 43 and its two subcommittees (SC) for the preparation of joint ISO and European (CEN) standards. ISO TC 43/SC 1 'Noise' has been developing standards for measuring noise produced by diverse sources in diverse acoustical environments. Standards are also being developed for the assessment of effects of sound on humans. ISO TC 43/SC2 'Building Acoustics' is engaged in standardization activities relating to architectural acoustics, building construction and sound propagation in buildings. The standardization activities relating to human hearing are carried out under the main committee TC 43. The major objective of ISO TC 43 and its subcommittees is international harmonization of methods for measuring noise and human hearing assessment. This provides a strong basis for noise reduction measures in products, buildings and the environment as well as for hearing conservation.

2. ACTIVITIES OF WORKING GROUPS

The priorities of ISO TC 43/SC 1 currently include the development of new or revised standards for machinery noise, transportation noise, environmental noise and hearing protectors. EU Directives require noise emission declarations by machinery manufacturers as well as sound power level limits for some outdoor equipment. Standards for the measurement of sound power level and emission sound pressure levels at workstations are being revised to better meet these regulatory requirements. Working Group 28 (WG 28) is currently involved in the revision of ISO 3740 3747, a series of standards on determination of sound power levels of noise sources using sound pressure measurements. A new work item has been proposed by the working group to revise the ISO 11200 to ISO 11205 series of standards on noise emitted by machinery and equipment for the determination of emission sound pressure levels at work stations of industrial machinery. Revisions to the standards within this series are intended to make them consistent with each other and make them user friendly by incorporating the recent developments in measurement methods and instrumentation technology. Measurement uncertainties of the ISO 11200 series standards on emission sound pressure level measurements meet engineering and survey grades of accuracy. Measurement methods described in the 3740 series of standards also include precision grade accuracy. WG 25 has recently completed the development of a three-part series on determination of sound power levels of noise sources using sound intensity measurements. Using this technique, sound power can be determined in any acoustic environment without the requirement of costly special facilities. Three experts from Canada are participating in WG 28 and WG 25.

WG 45 is concerned with the revision of a two part series of standards ISO 1996-1 and ISO 1996-2. Their common title is 'Description, measurement and assessment of environmental noise.' Part 1 is called 'Basic quantities and assessment procedures' and Part 2 is called 'Determination of environmental noise levels.' These standards provide guidance on standardized measurement and calculation procedures and rating methods for environmental noise. This provides a basis for legislative bodies to set limits on allowable noise levels, develop criteria for land use planning and set specifications for mitigation. Canada has one member in the working group.

Working groups 33, 38, 39, and 42 are involved in developing standards on road traffic noise, mostly related to automobiles. The activities of the first three working groups include measurement methods for comparing traffic noise on different road surfaces, sound absorption properties of road surfaces and pavement surface profiling.
WG 42 is concerned with the characterization of exterior vehicle noise under realistic driving conditions. Another activity of this working group is the development of methods for testing muffler performance on the roadside. Although these standards are of sufficient importance to

Canada, it has been challenging to find Canadian experts willing to participate in the working group.

WG 17 is developing standards on hearing protectors. The characterization of hearing protectors in a way that reflects their performance during actual use is crucial to hearing conservation for workers in a noisy environment. The activities of this working group include the development of measurement methods for insertion loss, sound attenuation characteristics, and active noise reduction. There is one Canadian expert serving on the working group.

SC1 is also involved in the development of standards relating to the spatial distribution of sound in workrooms, noise control in offices, silencers and noise in enclosures and cabins.

There are currently three active working groups on 'hearing' under the main committee TC 43. The activities of these working groups relate to definition of normal hearing, measurement of main hearing functions, calibration and proper use of audiometers and acoustical characteristics of hearing aids. WG 1 is concerned with the threshold of hearing, WG 6 is developing a standard on noise emissions from sound sources placed at the ears, and WG 7 is defining loudness scaling by means of categories. Canadian participation is provided through the recent appointment of two new members to the three working groups.

3. OPERATIONAL PROCEDURES

Development of a new standard by ISO TC 43 and its subcommittees is considered after a new work item proposal by a member body at the plenary meeting. For a new work item to be registered, it should have the support of at least five other member bodies that are willing to nominate experts to the working group. At this point a convener is nominated and he/she is required to come up with a working draft within 6 months of the formal registration of the work item. Normally working group meetings are held once or twice a year and, in the interim, most of the drafting is conducted through correspondence by e-mails. Within 18 months of registration, the convener is responsible for coming up with the first committee draft (CD) for comments by the subcommittee's member bodies. The working group normally prepares two or three committee drafts before bringing it to the stage of the draft international standard (DIS). The DIS is to be submitted by the working group for comments and voting by the member bodies within 24 months of registration. The document should receive at least 75% approval from the member bodies to proceed to the next stage. If the DIS is approved, a final draft international standard, (FDIS), is prepared, incorporating the comments and suggestions from the member bodies. The FDIS should be voted on to become an ISO standard within 36 months of registration. Again, this document is required to receive approval by 75% of the votes cast by member bodies to become an ISO standard. The work item will be cancelled if, at any stage, the document is unable to proceed to the next stage. If the target dates are not met, cancellation will occur if the document is not finalized within 7 years.

4. CANADIAN PARTICIPATION

Canada is actively participating in several of the ISO TC 43 and TC 43/SC1 working groups. Health Canada's research programs on the measurement of emission sound pressure level and sound power level are providing input to the development of basic machinery noise emission standards. The National Research Council of Canada through its active research programs in several areas of acoustics has provided substantial input to the development of ISO standards. CSA is adopting and endorsing many ISO standards. A proposed CSA standard 'Noise emission declarations for machinery' is based on several ISO documents on noise emission and sound power standards. The CSA standard provides information to manufacturers, purchasers and importers/exporters on noise data that should be supplied with new machinery.

For several working groups, it has been difficult to find suitable experts who are willing to participate and have access to travel funds. Experts are needed in many areas that include transportation noise, environmental noise and hearing.

NOISE EXPOSURE OF MUSIC TEACHERS

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1. INTRODUCTION

Music teachers are exposed to different noise sources during the course of their activities (teaching band, choir, etc). The size and activities of their classes varies greatly and so are the noise levels they are exposed to. Although students are also exposed to the same or even higher noise levels while playing or singing, the duration of their noise exposure is much shorter than that of the teachers. On a given day, a student may be exposed to high noise levels for only one music class while a teacher can be exposed to high levels for many class periods. Consequently, the risk of hearing loss due to noise exposure in music classrooms is potentially significant among teachers but not among students.

There is abundant literature dealing with the risk of hearing loss among musicians. However there is very little data regarding the exposure of conductors of music ensembles or of music teachers. The only study of teachers' noise exposure that we are aware of was performed in British Columbia (1). A potential reason for the lack of similar studies is the difficulty in determining a "typical" day or week of teachers. The distribution of activities varies greatly from teacher to teacher (due to differences in resources between schools), from week to week and from day to day for a given teacher. Extra-curricular activities such as competitions, school acts, musicals, etc., requiring bands, orchestras or/and choirs increase both the frequency and number of participants at rehearsals.

The study was based on measuring noise exposure levels from single activities (classes) such as rehearsing of bands or teaching of theory. The choice to focus on an activity was made because the "activity" is more easily defined than determining a "typical" day. The physical environment (acoustical characteristics of each room), activity duration, and the number of students involved were recorded. With the knowledge of the average noise exposure for a particular activity and the total duration of activities in an average day or week, one can estimate the noise exposure of a teacher based on the activities they perform.

2. METHOD

The survey was conducted on 18 music volunteer teachers from 15 different schools in an Ontario School Board. There were no special requirements for the teachers, such as a minimum hearing acuity, age, sex or length of service. Participants were explained the objective of the study and the measurement procedure. They were also advised that the study was anonymous and that their names will not be disclosed.

Measurements were performed using Quest Type Q-300 dosimeters. Dosimeters were calibrated using Quest Type QC-10 calibrator, following the procedure recommended by the manufacturer. A B&K Type 2231 Modular Precision Sound Level Meter was also used as a rough check of the results read on the dosimeters.

Each teacher was followed for a day (or as much of a day as possible). A member of the team attached the dosimeter microphone to the teacher's collar and started recording. Members from the team followed the teacher during the measurement period taking notes of activities performed, their duration and of the acoustical characteristics of the environment.

Back in the Laboratory, the information stored in the memory of the dosimeters was extracted using the QuestSuite Professional computer program. The same program was used for setting the dosimeters and for checking their calibration.

Noise Exposure Level (L_{ex}) and Equivalent Noise Level (L_{eq}) for each activity was calculated from recorded dosimeter data. The noise exposure criterion used was a daily 8 hrs exposure limit of 85 dBA and a 3 dB exchange rate.

3. RESULTS

The frequency distribution of the L_{eq} and L_{ex} can be seen in Figure 1. From the figure, it can be seen that while the mode of the L_{eq} is the 88 – 90 dBA range, the mode of the L_{ex} is the 84 – 86 dBA range. The measured L_{eq} exceeded the 85 dBA limit for 14 teachers (78%), was at the limit on one occasion (5%), and under the limit on 3 occasions (17%). The calculated L_{ex} exceeded the 85 dBA limit for 7 teachers (39%), was at the limit for 4 (22%), and under the limit for 7 (39%). The average measurement duration was 4.5 hrs (std.dev. 1.4 hs).

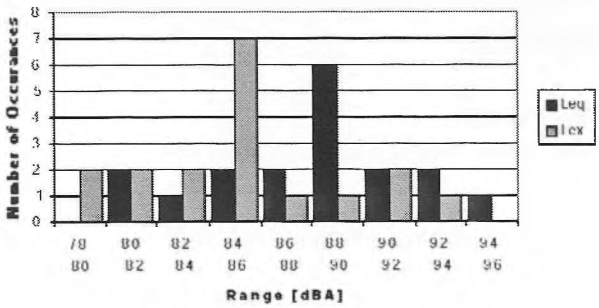

Figure 1. Frequency distribution of L_{eq} and L_{ex}.

Five common activities were considered: Singing, Percussion, Keyboard, Recorder and Band. For each activity the mean, standard deviation and range are stated, where the number of samples allows for those calculations (see Table 1).

Table 1. Leq for each activity in dBA (standard deviation and number of samples given in parentheses).

Activity	Elementary	Secondary	Total
Singing	87.1 (3.8, 14)	88.3 (5.4, 4)	87.3 (4.0, 18)
Percussion	86.9 (3.4, 9)	84 (- , 1)	86.6 (3.3, 10)
Keyboards	84.4 (4.0, 6)	-	84.4 (4.0, 6)
Recorder	88.2 (1.9, 5)	-	88.2 (1.9, 5)
Band	91.7 (3.3, 12)	90.5 (3.6, 19)	90.9 (3.5, 31)

No significant difference between the mean noise exposure levels from elementary or secondary schools was found. However, three of the four singing measurements at the secondary level were of amplified singing (i.e. singing into microphones). The mean Leq for the amplified singing was 90.3 dBA with a standard deviation of 3.1. The overall mean for un-amplified singing is 86.7 dBA with a standard deviation of 4.0.

4. DISCUSSION

The L_{ex} calculated for the majority of the measurements was at or below the 85 dBA level. However, the calculation of Lex assumed that the subject was in a quiet environment outside of the measurement period. This assumption may not be true as extra-curricular activities were not measured and in some cases it was not possible to follow the teachers throughout an entire day. Thus, we can not conclude that the noise exposure of an average music teacher is either safe or unsafe.

However, we can estimate the average L_{eq} for various activities that a music teacher would perform. From this we can calculate a maximum safe exposure time to each activity (the exposure duration for which the L_{ex} will reach 85 dBA). The calculated "safe" exposure durations for each activity can be found in Table 2.

Table 2. "Safe" exposure duration for various activities.

	Hours per Day	Hours per Week
Singing	5.4	27.0
Singing (amplified)	2.4	11.8
Percussion	5.5	27.7
Keyboards	9.4	47.0
Recorder	3.8	19.1
Band	2.1	10.3

It should be noted that the exposure durations calculated in Table 2 assume that the subject is in a quiet environment for the balance of the 8 hrs work day/40 hrs work week.

From Table 2 it is clear that teachers should wear hearing protection when teaching band and amplified singing as the "safe" exposure duration for these activities are only 2.1 and 2.4 hs respectively. Hearing protection may also be needed depending on the schedule of the music teacher.

We recommend that a hearing protection program be instituted where teachers are made aware of the potentially hazardous noise levels, provided with hearing protection (disposable musician earplugs), and educated about the care and use of earplugs. As well, teachers should undergo bi-annual audiometric tests to ensure the hearing protection measures are effective and the teacher's hearing has not been reduced.

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FILTER SELECTION TO ADAPT EARPLUG PERFORMANCES TO SOUND EXPOSURE

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1. INTRODUCTION

For hearing protection to be made effective, the research needs established by NIOSH [1] (National Institute for Occupational Safety and Health) are to find a way for workers to be individually fitted and to offer them increased comfort and the ability to hear speech and warning signals.

To address these individual fit and comfort issues, a new concept has been developed; a re-usable earplug that is custom-fitted using silicon injection- and field tested for attenuation on the worker [2].

The ability to hear speech and warning signals can be partially addressed by adapting the earplug attenuation to the actual noise exposure of the worker [3]. This proposed adaptation is based on a set of acoustic filters that could be placed into the earplug's sound-bore to lead to a protected exposure level between 70 to 85 dBA.

2. INDUSTRIAL NOISE EXPOSURE LEVELS IN CANADA

Based on the data published by Statistics Canada [4] the number of workers in various industrial areas is identified. The corresponding noise exposure is then determined from compilation of published data in the areas of construction , refined petroleum and plastic [6], forestry [7, 8], food, beverages & machinery [9-11], printing and textile [12], transports [13, 14], and other areas [15].

Exposure Levels	Number of Workers	Relative Weight
85 - 90	140,000	6.2%
90 - 95	793,300	35.3%
95 - 100	701,000	31.2%
>100	612,000	27.2%
Total	2,246,300	100.0%

Table 1: Summary of exposure levels and number of workers exposed in Canada's workplaces

3. PROTECTION OUTCOME OF FILTERED EARPLUGS

3.1. *Attenuation Data of Filtered Earplugs*

The attenuation of custom earplugs filtered with a set of 9 elements (Full-Block, 4700 Ω, 3300 Ω, 2200 Ω, 1000 Ω, 680 Ω, 330 Ω and 0 Ω dampers) has been measured per octave bands -from 125 to 8000 Hz - on an Artificial Test Fixture (ATF).

3.2. *Protection Outcome of Filtered Earplugs for Typical Industrial Noises*

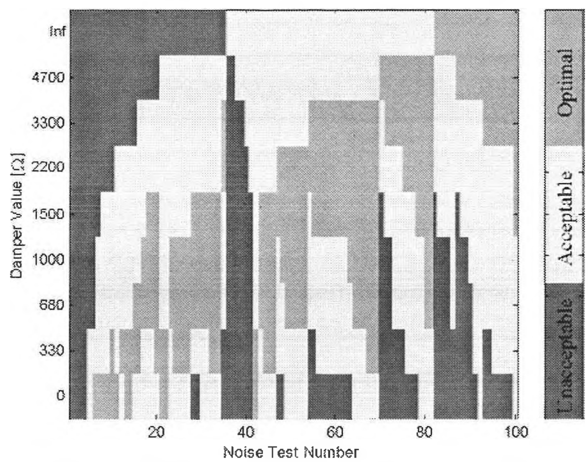

Figure 1: Protection outcome for the different filters on 100 typical industrial noise spectrums ("Unacceptable" are protected sound exposure levels below 70 or above 85 dBA; "Acceptable" are between 70 and 75 dBA or between 80 and 85 dBA; "Optimal" are between 75 and 80 dBA).

From published typical industrial octave-band spectra [16] and the measured filtered earplug attenuations we can compute the protected exposure levels according to the *Octave Band Procedure*; the resulting Protection Outcome –as defined by CSA[3]- is presented in Figure 1.

For 89 of the 100 industrial noises, at least one appropriate filtered earplug could be found that leads to an *optimal* protection (versus 96 for merged *optimal* and *acceptable* protection outcomes).

4. SELECTION OF FILTERED EARPLUG FOR ADEQUATE PROTECTION

	%	85-90	90-95	95-100	>100	Weighted Mean
Damper Value [Ω]	∞	0.0	2.0	40.7	35.0	22.9
	4700	0.0	10.0	59.3	25.0	28.8
	3300	0.0	32.0	59.3	20.0	35.2
	2200	0.0	38.0	33.3	20.0	29.3
	1500	33.3	56.0	29.6	20.0	36.5
	1000	33.3	50.0	25.9	10.0	30.5
	680	33.3	50.0	25.9	10.0	30.5
	330	0.0	38.0	11.1	5.0	18.2
	0	33.3	16.0	7.4	0.0	10.0
Optimal Situation		66.7	100.0	100.0	50.0	**84.2**

Table 2: Percentage of optimal protection for the 9 different filters when the exposure levels are between 85 and 100 dBA and more.

The percentages of optimal protection are calculated in Table 2 for every filtered earplug for noise exposure levels from 85 to 100 dBA and more. The *relative weight* (see Table 1) of each of those noise exposure classes is used to compute the *Weighted Mean* percentage that reflects the usefulness of the corresponding filter to correctly protect workers in Canada's industrial workplaces. The *Optimal Situation* line reflects the percentage of noise cases where at least one filter provides an optimal protection. A global coverage of 84.2% is obtained with the set of 9 filters. Since the protection of some filters overlap, a similar computation could be applied to a subset of filters. For example, it is possible to find among the 512 ($512=9^2$) possible subsets, 2 subsets of only 6 filters that give the same 84.2% *optimal* coverage. When also including *acceptable* protection, the global coverage increases to 94.6% and can be easily obtained with 12 subsets of only 3 filters.

5. CONCLUSION

This study has demonstrated that it was possible to filter a custom earplug to provide an adequate protection in most of Canada's industrial workplaces. The use of a reduced set of simple acoustic filters avoids most of the over-protection situations (uncomfortable and dangerous because speech and warning signals can not be heard) and under-protection (dangerous because of the over-exposure).

6. ACKNOWLEDGEMENTS

The support of SONOMAX, IRSST (*Quebec Occupational Health and Safety Research Institute*) and NSERC (*Natural Sciences and engineering Research Council of Canada*) is gratefully acknowledged.

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TESTING OF ANR EARMUFFS

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1. INTRODUCTION

Although originally patented as early as in 1930s [1], the technique of Active Noise Reduction (ANR) did not find practical application in the hearing protection field for a long period of time. Only in the last 20 years has ANR been introduced in the industry of hearing protectors, primarily due to advances in signal processing technology. An ANR hearing protector has its low frequency attenuation increased by electronic means, by employing a feedback-processing loop, where the intruding signal is re-introduced with a 180-degree phase shift. As a result, the sound pressure level is significantly reduced at those frequencies.

ANR is only effective in the low frequency range, below 500 Hz, because of technical limitation. It is well known that the sound energy that damages the hearing of noise-exposed people is found in the frequency above 500 Hz. Therefore the mere use of ANR does not reduce the risk of hearing loss. However, it does reduce the upward spread of masking effect [2]. By doing so, it increases significantly the speech intelligibility, thus improving oral communication in environments such as airplane and helicopter cabins, armored cars, engine rooms, etc.

Several characteristics of the headsets can be measured. They are the attenuation at different sound levels, intelligibility as perceived by the wearer and the comfort. Although testing of those characteristics is something manufacturers as well as authorities are interested in, there are still no test methods standardized or even recognized.

The objective of the study was to examine the feasibility of using an Acoustic Test Fixture (ATF) for the measurement of insertion loss (IL) of ANR headset. Using the IL results, the attenuation canbe easily calculated.

2. MATERIALS AND METHODS

A total of five headsets were employed in this study: two supra-aural and three circum-aural. **Headset-1**: A supra-aural type headset, used mainly as a comfort device in airplanes. It included a plug connecting to the airplane entertainment center or to sound reproducing equipment. Manufacturer's brochure claimed up to 10 dB cancellation at 300 Hz and a cancellation range of 40 – 1,500 Hz. **Headset-2**: Same characteristics as Headset-1. From the manufacturer's brochure, the cancellation range was 20 – 1,500 Hz and the reduction was 15 dB between 150 and 300 Hz. **Headset-3A and 3B**: Both of the circum-aural type. No technical specifications regarding the ANR performance were published in the manufacturer's brochure. **Headset-4**: A circum-aural type aviation headset. No technical specifications regarding the ANR performance were published in the manufacturer's brochure either.

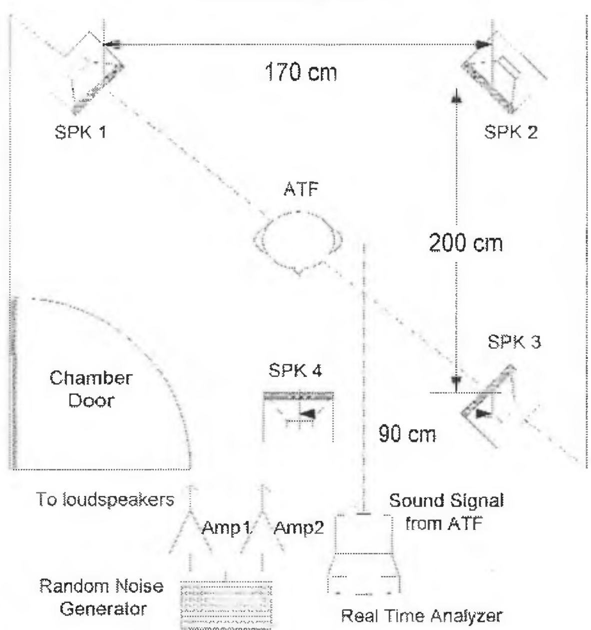

Fig. 1 Measurement set-up

Headsets were mounted on the ATF, a binaural implementation of a mannequin with one instrumented ear [3]. Features of the mannequin include circum-aural areas, pinna and auditory canals fabricated with simulated skin and tissue that retains the correct dynamic mass and textural properties of human flesh. The auditory canal is terminated in Zwislocki type DB100 coupler and Bruel & Kjaer type 4134 microphone, which simulate the acoustical impedance of human ears. The measurements were performed using the instrumented ear.

Tests were carried out in an IAC double wall, double room Audiometric Cabin (measurement set-up shown in Figure 1). A pink noise signal, generated by a General Radio Random Noise Generator Type 1382, was amplified by two Rotel Stereo Integrated Amplifiers type RA-930AX (50W) and fed into the room via four Mirage Speakers Type M-90IS and four horn-loaded piezoelectric loudspeakers Motorola type KSN1016. The signal entering the ear of the ATF was detected by the microphone in the coupler and analyzed by a Bruel & Kjaer type 2144 Real Time Analyzer (RTA). Measurements were performed in 1/3-octave bands at the center frequencies between 20 and 8000 Hz.

Measurements were carried following three steps: a) The Open-Ear Spectrum (OE) was measured with the sound signal on and no headset on the ATF; b) Without changing the sound signal, the Passive-Protected-Ear Spectrum (PP) was measured with the headset donned. The ANR system of the headset was off at this step; c) The Active-Protected-Ear Spectrum (AP) was measured with the ANR system switched on, while keeping other conditions same as in step b). Steps b) and c) were repeated 20 times in each measurement session, without altering the sound signal. The purpose was to examine the changes of the insertion losses resulting from repeatedly donning and doffing the headset. A study of those variations is in progress.

From the spectra in steps a) through c), the following IL was calculated for each one of the 1/3-octave band frequencies. **Passive Insertion Loss** (IL_P): the insertion loss of the headset with the ANR off, IL_P = OE PP (dB); **Active Insertion Loss** (IL_A): the insertion loss due to the effectiveness of ANR only, IL_A PP – AP (dB); **Total Insertion Loss** (IL_T): the total insertion loss of the headset with the ANR on, IL_T OE – AP (dB).

3. EXAMPLE OF RESULTS

Figures 2 4 show insertion losses IL_P, IL_A and IL_T calculated from 20 measurements of Headset 1 [4].

At each frequency point, the maximum and minimum values of the 20 measurements are shown as the upper and the lower bar.

4. DISCUSSION AND CONCLUSION

It was observed that there were significant variations among the calculated values of different ILs at different frequencies. As is expected, however, the high frequency range of the noise is attenuated by passive insertion loss of the headsets. On the other hand, the low frequency is attenuated by active noise reduction. Thus this ANR headset obtained broadband noise attenuation both in low and high frequency band, as is shown in Figure 4.

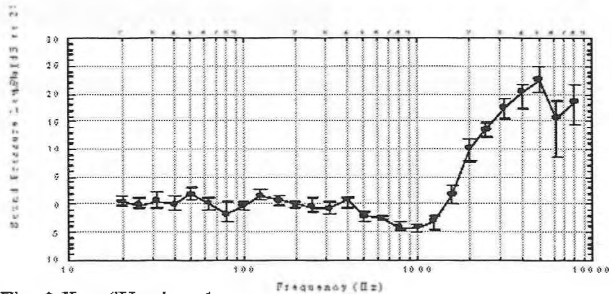

Fig. 2 IL_P of Headset -1

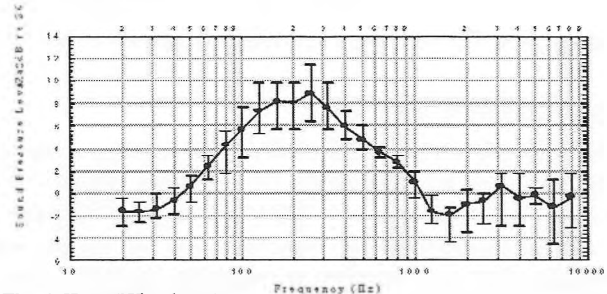

Fig. 3 IL_A of Headset-1

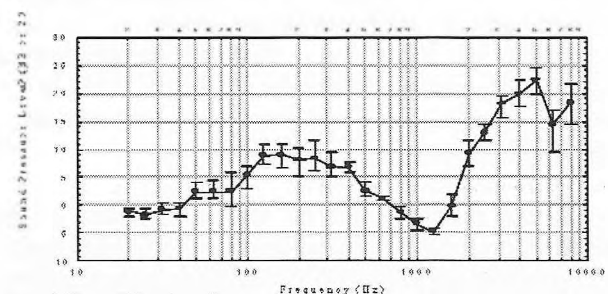

Fig. 4 IL_T of Headset -1

The method allows for a minimum of instrumentation and is relatively simple to implement. It requires an audiometric booth, a signal generator, speakers, and an ATF. The ATF is an essential part of the instrumentation. The background noise inside the booth is not important, since measurements are implemented at a level of approximately 80 dBA. In addition, non-involvement of human subjects reduces the overall cost and makes the tests easy to perform. As an example, the 20 IL tests of one headset were completed within an hour. Therefore, the method becomes especially useful for testing prototypes, since it allows for quick modifications of device for a subsequent re-test. It can be also useful in the case of quality control, because it allows for testing large quantity of headsets in a short time.

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SPEECH ACOUSTICS IN A NOISY ENVIRONMENT

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1. INTRODUCTION

This paper reports on one aspect of the Acoustic Ecology project of the UBC Institute for Hearing Accessibility Research. We examined the effect of a noisy environment on speech communication, with the aim of testing our hypothesis that a noisy room is a hearing-impairing room (Jamieson et al., in prep.). We predicted that the speech characteristics of normal-hearing people in a noisy environment will be similar to those of hearing-impaired people in non-noisy environments. Extensive video, audio, and dosimeter data were evaluated in an examination of discourse productions and shifts of attention.

The speech characteristics of hearing-impaired people include: raised f_0; reduced pitch range; reduced or excessive pitch variation; non-normal pitch contours, lack of terminal rise; lack of normal pitch declination; lack of normal pitch rhythm, i.e., distinction between stressed vs. unstressed units; breathy phonation; vocal tension/harshness; non-normal phoneme duration; inappropriate pausing; nasality; and lowered vowel formants. (See Bench 1992, Chen 1995, Higgins et al. 1994, Itoh & Horaii 1985, Kato 2001, Kotby et al. 1996, McGarr & Whitehead 1992, O'Halpin 2001 and references therein, Pisani 1982, Whitehead & Whitehead 1985, Wirz 1988, Youdelman et al. 1989.)

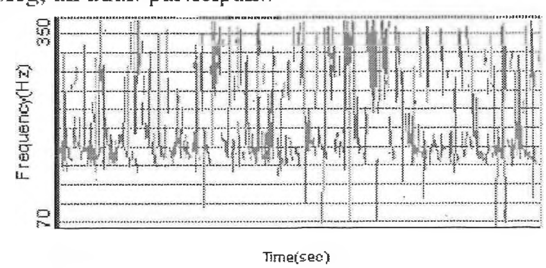
Our research question is: *Does the speech of normal-hearing people in a noisy environment show the characteristics listed above?*

2. METHOD

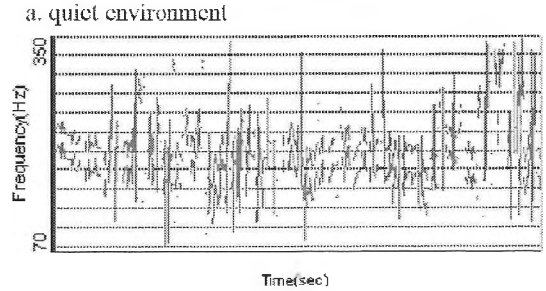
The subjects were normal-hearing adults aged 24 to 57 (n = 12) and children aged 7½ to 13 (n = 24) of both genders. Their speech was recorded in naturalistic noisy environments, a real restaurant setting for the adults and real classrooms (grades 1-3, 5, and 7) for the children. Background noise levels were up to 70 dB (average A-weighted Leq). The noisy environment speech was recorded as received by listeners' ears at binaural ear-level microphone sets, since listeners' ears are the site of any interactionally functional signals. The participants' speech was also recorded with a unidirectional microphone in a soundbooth. The DAT recordings were then loaded onto computer and the speech signals were acoustically analysed using Kay's Multi-Speech 3700®. The acoustic characteristics of the speech in the two environments were compared to identify any of the characteristics of hearing-impaired speech which might be found in the participants' noisy environment speech but not their speech in the soundbooth. Where they are robust, we attribute such characteristics to the noisy environment.

3. SOME RESULTS AND ANTICIPATIONS

Preliminary results indicate that the answer to our research question is 'yes' in several cases. For example, Figure 1 illustrates reduced pitch variation in the noisy environment (sd: 51.05 vs. 65.40 in the quiet environment) for Meg, an adult participant.



a. quiet environment



b. noisy environment

Fig. 1. Pitch contours illustrating reduced pitch variation in noisy environment adult speech.

The full results of our study will be reported. We anticipate that the preliminary results will hold for the full dataset, in support of our hypothesis. This would raise several further issues: 1. to what extent are there similarities between the characteristics for which our findings are 'yes' and the Lombard Effect (Lombard 1911, Junqua 1996), such that the Lombard Effect might be alternatively labelled 'hearing-impaired speech by normal-hearing people'; 2. does the 'hearing-impaired speech by normal-hearing people' effect cast doubt on physical or developmental explanations applied to the hearing impairment (Most & Frank 1994); 3. what potential further support exists for our hypothesis from people's adjustment of their speech to accommodate their listeners' hearing impairment (Imaizumi & Hayashi 1995), if we consider that the normal listeners effectively experience hearing loss in adverse listening conditions; 4. when and why is temporary hearing impairment in a noisy environment interpreted or not interpreted as a signal of cognitive impairment (Pichora-Fuller & Kirson 1994, McKellin 1994); 5. what are the implications of such acoustic ecology issues for environmental design and evaluation.

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NOISE POLLUTION IN A GENERAL HOSPITAL

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1. INTRODUCTION

Noise is a known environmental pollutant and health hazard. Particularly, noise pollution in a general hospital (ex: operating room, recovery room and intensive care unit) is a hazard to surgeons, anesthetists, personnel and patients. It is well documented that noise is stressful, eliciting changes in the autonomic nervous system, impairing mental faculties and producing masking that could affect the staff and the conscious patient alike, leading to decreased work performance and increased anxiety respectively. Moreover, operating room noise can reduce the mental efficiency and short-term memory of anesthesia residents. Therefore noise prevention needs more attention and should be a routine part of patient care.

The design of a modern General Hospital should consider acoustics as one of the main factors to provide comfort to both patients and medical staff. The acoustic environment is determined by the atmosphere of the site (external noise) as well as by the performance of elements as room dimension and destination, construction materials, surfaces and furnishings. Every possible item of sound transmission must be considered from the designer and every possible way of noise impact reduction have to be adopted. The acoustic requirements of doors, walls and ceiling constructions must be calculated and their resulting values can be used to check and correct the presence of unwanted noise. Consideration also needs to be given to the impact of internal services such as plumbing, electrical and mechanical plants and distribution systems like air-conditioning as well as external noise generated by linear sources (road and rail traffic noise) or other point sources in the area.

The paper refers to case studies carried out in some General Hospitals in Tuscany (Italy). In figure 1 the brand new Versilia General Hospital is represented.

2. METHOD

The methodological approach to estimate acoustic atmosphere and acoustic comfort in internal hospital areas have been studied. Noise pollution and sound impact of machines, activities, traffic and other sources have been considered by adapting algorithms and rules provided for in the specific ISO concerning emission and attenuation of noise. Starting from information about territory and structure of buildings, a study of the noise atmosphere has been carried out in a context represented by various different types and compositions of sources and receivers.

The structure of buildings and the related characteristics of sound propagation in the hospital areas and outdoors are analyzed using the ISO methods and models concerning the acoustic properties of buildings.

The chosen method of acoustic analysis considered building and activities as sources of noise as well as noise receivers. Starting from the acoustic climate "ante operam", the study has been performed according to the following program:

- acoustic analysis of inner and external sources;
- measurement and computation of the acoustic impact on the inner and external receivers;
- analysis of acoustic requirement of the building..

Noise impact of the new building on the acoustic atmosphere of the surrounding receivers has been tested and analyzed. Italian City Administrators ask for noise impact prediction, as a necessary preliminary document, to authorize every potentially pollutant activity, as a new general hospital can be.

Measurements of the residual level have been carried out in emplacements homogeneously distributed along the perimeter and in the building area, privileging the directions of propagation from the more meaningful sources towards the closer receivers (see fig. 1).

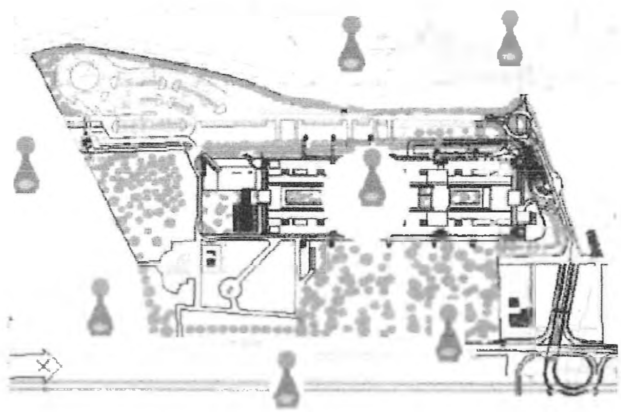

Fig. 1. The new Versilia General Hospital in Tuscany (Italy)

The considered International Standard references are ISO 8297 for the determination of sound power levels of multisource industrial plants for evaluation of sound pressure levels in the environment, ISO 9613-2 for the method of calculation of the attenuation of sound during propagation outdoors and the standard acoustic tests for the conditioning and ventilating systems an plants.

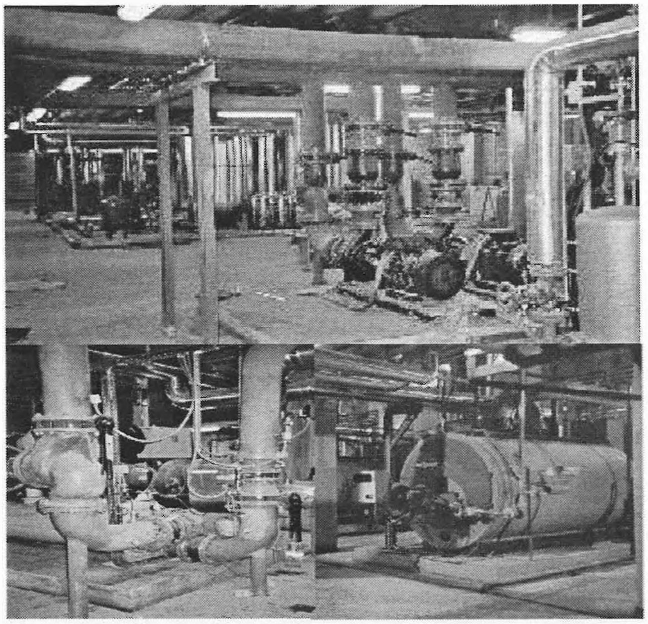

Fig.2. Internal sources

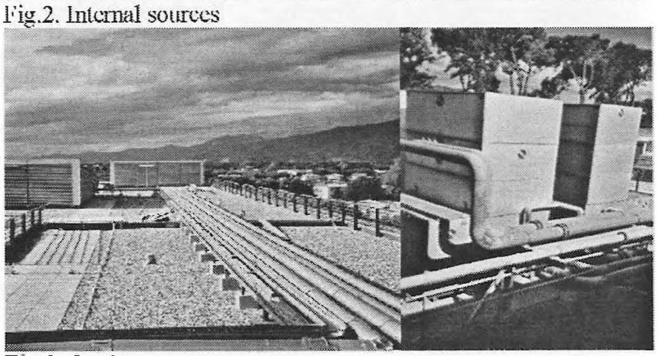

Fig.3. Outdoors sources

In the new Versilia General Hospital the system of sources occupies the building area, the roof and a portion of the surrounding area in a territory that is principally destined for rural use, although a motorway and a high traffic national road stand in proximity of it.

The analysis of the acoustic quality of the building has been performed measuring and calculating all the significant parameters, such as R_w, $L_{n,w}$, $D_{2m,nT,w}$ as defined by the standard ISO, and remembering that the effects of noise on people depend primarily on the duration and timing of the sounds. Each Operating room received special consideration as one of the sensitive areas. Sometimes the flat, water impermeable walls of the modern operating room are outstanding reflecting surfaces for sounds. Noise levels in the operating room can easily approach those from a diesel engine. Loud sounds have been shown to contribute to stress as measured by responses of the pituitary-adrenal axis. In addition to affecting health care workers, the din of the operating room and other acute care areas can be disconcerting to conscious patients. Although noise is known to cause problems other than hearing loss, it appears prudent to limit unnecessary sources in the operating room.

parameter	room 1	room 2	room 3
R_W	41.5	44.5	34.0
$D_{2m,nT,w}$	30.0	21.5	21.5
$L'_{n,w}$	59,0	58,0	58,5

Table 1. Acoustic Quality parameters for three different operating rooms, measured and calculated according to standard ISO rules

3. RESULTS

This method brings to a view of the acoustic performance of a General Hospital, considering the sources and the receivers of noise inside and outside the building.

Various applications of the method have generated and tested a strategy for determining the suitability of an urban or rural area to accommodate a General Hospital.

The study of the systems generating noise pollution, represented by different types and compositions of sources has been performed and the frequency characterization of the emissions has been analyzed. The structure of the buildings and the related characteristics of sound propagation towards the sensitive receivers outdoors have been studied. The acoustic separation and insulation between adjacent rooms or areas have been tested as well as the avoidance of cross-talk via duct work between adjacent rooms. The design of systems and plants had to be changed to minimize air-conditioning and plumbing noise, the design of the internal areas must consider the need of adequate privacy for areas such as bathrooms and the need of acoustic quality in rooms where speech intelligibility is very important, such as consulting rooms or other areas where to ensure patient confidentiality. Finally the design of the perimetrical walls must consider the air borne and structure borne propagation generated by traffic and point sources and transmitted by the structure itself..

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NOISES OFF: WHAT VOCALIZATIONS CAN AND CAN'T TELL US ABOUT ANIMAL WELFARE

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1. INTRODUCTION

In Canada and elsewhere, there is growing public concern about animal welfare. This paper reviews the background to animal welfare research and highlights some welfare studies that have used vocalizations. Emphasis will be placed on the studies' practical significance.

2. BACKGROUND TO ANIMAL WELFARE RESEARCH

Animal welfare is more than what animals feel. The most widely accepted definition is that welfare is an inherent state comprising bodily function (e.g., health), feelings (e.g., fear, pleasure) and the animal's nature (genetic traits manifest in e.g., breed and temperament) (Fraser et al. 1997). Current scientific questions about animal welfare have arisen from developments in our relationship with animals during the last ~50 years. These developments provide the context in which the results of welfare research must be applied, and are outlined below.

Following World War II, farming methods were intensified in order to ensure an abundance of cheap food. Public expectation of the latter continues and results in very small profit margins for farmers, many of whom would not choose the production methods that the market requires. However, having cheap food has increased our disposable income. This increase, coupled with changes in societal fabric, has led to many more dogs and cats being kept as companions. The resulting human-pet bond is intense and has contributed to the present sophistication of companion animal medicine (e.g. diagnosis by MRI; hip replacements). All these developments continue to prompt questions about the effects of our usage on animals. The discipline of animal welfare science seeks to address these questions.

Animal welfare science applies physiology, ethology, neuroscience and veterinary medicine to questions about what happens to animals when we treat them in different ways. Animal welfare is complex, therefore there is not yet a comprehensive way to assess it. Instead, scientists tend to focus on specific questions (e.g., Are variations in piglet vocalizations related to degree of need?) and to use either physiological or behavioural measures of outcome. Both types of measure are objectively quantifiable, but carry the contradiction of being used to make judgements about the animal's total experience - how it is faring - which is partly subjective and therefore not accessible to measurement within a Cartesian framework (Wemelsfelder 1997). However, despite philosophical criticisms, traditional behavioural measures can tell us something of how animals are faring e.g. their needs and aversions. Vocal behaviour has been used in this way, as outlined in the next section.

3. USE OF VOCALIZATIONS IN ANIMAL WELFARE RESEARCH

Vocalization offers promise as an endpoint in animal welfare research. Its role has been reviewed by Watts & Stookey (2000) who note that the multidimensional nature of vocalizations (frequency, duration, rate) and their specific role in communication, offer advantages over less specific indices that can increase in response to pleasant or unpleasant events and are therefore difficult to interpret (e.g.,heart rate). The authors identify several categories of vocal response that may assist in welfare research, including pain-related vocalization, need-related vocalization and social vocalization.

3.1 Pain-related vocalization

Many species vocalize when in pain. The calls indicate clearly that the caller is distressed, therefore they are useful in welfare research. Pain-related vocalization has been used to study the welfare of animals undergoing (i) procedures needed to safeguard welfare under management systems that would otherwise lead to reduced welfare (e.g., debeaking of poultry chicks to prevent cannibalism), (ii) procedures that protect the animals' long-term welfare (e.g., docking lambs' tails to prevent "fly strike"), and (iii) procedures that are conducted for human benefit (e.g., castration of piglets to avoid "boar taint" in meat). The use of vocalization to examine the welfare of (male) piglets at castration is reviewed below.

Piglets are castrated at ~1-2 weeks of age. The procedure is surgical and takes ~75 seconds. Economic and practical considerations preclude use of anaesthesia or analgesia (e.g., anaesthesia would interfere with piglets' return to the litter and to suckling). A series of randomized, controlled studies with appropriate statistics, used the rate and frequency of vocalization to identify those aspects of castration that are most distressing to piglets. Weary et al (1998) compared, in litter-mate pairs, piglets that were surgically castrated with those that were sham-castrated. The authors found that (i) the frequency of individual calls showed a bimodal distribution with peaks at 100-600Hz and 3000-4000Hz, and a trough at ~1000Hz, (ii) piglets undergoing castration made significantly more calls over 1000 Hz than did shams, and (iii) different methods of restraint produced different rates of high-frequency calling, but restraint did not affect the pain caused by castration. Using a similar design, Taylor and Weary (2000) investigated which part of the surgery caused the most pain. High-frequency calling was increased by skin incision and was highest when the spermatic cords were pulled and cut. The method of cutting the cord did not make a significant difference to the rate of calling. Further work examined the effect of age (Taylor et al 2001). Piglets castrated at 3, 10 and 17 days of age were compared with sham-castrated littermates. Older piglets in both groups produced more high-frequency calls of longer duration than did younger piglets. However, age did not affect the rate of high-frequency calling during castration which was three times the rate during sham-castration. Duration of calls was not affected by castration. Taylor et al (2001) concluded that, ideally, male pigs should be left intact and slaughtered when younger in order to avoid "boar taint". Until this is practised, they recommended development of non-surgical castration. The case illustrates that science alone cannot indicate how animals should be treated. That question also depends on our values (Fraser 1995), because animals are not the only moral stakeholders in issues of welfare concern. At present, the value placed on preventing male piglets from suffering is overriden by that placed on our having cheap food. Thus, pigs are slaughtered when farmers can get maximum return on their inputs, but this is also when meat from intact male animals would be unpalatable.

3.2 Need-related vocalization

The concept of "honest signalling" has been applied to questions of whether animals' distress calls are reliably related to degree of need (Watts & Stookey 2000; Weary & Fraser 1995). Weary & Fraser (1995) found that the rate of calling in piglets seemed to reflect the degree of hunger and of thermal discomfort i.e., the need for food and warmth.

3.3 Social vocalization

An important type of social vocalization is the calling between mother and offspring. Early separation causes both to vocalize and show distress, but is a commercial necessity in some farm species. Dairy cows are usually separated from their calves 24 hours after birth. This is thought to be less distressing than separation at 4 days when milking of the cow resumes. Work by Weary and Chua (2000) confirmed this: calves and cows vocalized more when separated 4 days after birth, than at either 24 hous or 6 hours. However, the older calves had less diarrhoea. Thus, choice of separation time requires judgement between distress and health, on both welfare and economic grounds.

4. CONCLUDING REMARKS

This paper highlights some important studies that have used vocalization to address questions about animal welfare. Other such research includes the effect on dogs in shelters of the noise from barking, vocalization as a sign of anxiety in dogs left alone and of distress in farm animals at slaughter, and stereotyped vocalization as a sign of extreme frustration. Vocalizations could also be used to examine when animals are experiencing pleasure. Research in these areas will help to inform the debate about animal welfare.

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Utilization of species specific vocalizations to improve productivity and welfare in food producing animals

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1. INTRODUCTION

The need to efficiently produce abundant, safe and healthy food of animal origin often conflicts with ever increasing societal concerns about animal welfare. Traditional management methods may increase productivity but create a detrimental effect on animal welfare. The use of species specific vocalizations, be they maternal, from the progeny or in combination, appears as a clear opportunity to enhance productive processes without compromising animal welfare.

1.1 Objectives

Three species; poultry, turkeys and swine, have been used to evaluate the effectiveness of using maternal, progeny or a combination of vocalizations to enhance different aspects of production. In the avian species the objective was to initiate ingestive behaviour as early as possible and to maintain higher levels of food consumption by reproducing maternal calls. In pigs the objective was to increase frequency of feeding by using a combination of sow and piglet nursing sounds.

2. METHOD

2.1 Turkeys

Recordings of contempt turkey hens vocalizations were reproduced in the incubator for five minutes every hour during the last 5 days of incubation. Recordings of broody turkey hens vocalizations were reproduced trough speakers placed in the feeder of newly hatched poults for two minutes at 20 minutes interval. These were compared to groups of animals raised in similar housing without the sound stimulation. Performance in growth and mortality were recorded every three days for the first three weeks of age.

2.2 Poultry

Recording of broody hen were played back for two minutes every 20 minutes from within the feeder or from a speaker in a lateral wall of the pens. Growth performance was monitored for the first two weeks of age.

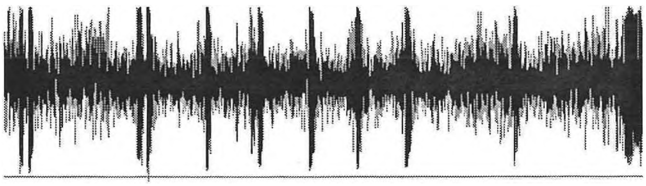
2.3 Swine

Sounds were recorded in a farrowing room (including sow feeding calls, piglets feeding grunts and room background noise). These recordings were played back to newly farrowed sows for 3 minutes at either 37, 47 or 57 minutes during the first three weeks of life.

3. RESULTS

3.1 Turkeys

The vocalizations of turkey hens are characterized by repeated cycles (4 sec long) of a sequence of clocking sounds at 0.3-0.4 sec intervals. (Fig.1) The majority of the spectrum is based in frequencies of 1-5.4KHz and almost nothing below 400Hz.



Fi g. 1. Pattern of one cycle of vocalizations of a broody turkey hen.

There was no beneficial effect on exposing eggs to contempt vocalizations at the end of the incubation period. Exposing poults to broody feeding calls elicited feeding response within the first 2 hours after placing in the pens. The increased feeding activity observed in sound exposed animals translated in an increase in body weight of 16, 19 and 12% on days 3, 15 and 21 respectively. This weight increase did not translate in significant changes in body composition .

3.2 Poultry

The vocalizations of a broody hens were repeated cycles of (about 2.5 sec long) a sequence of 8 to 10 short clocking repeated at increasing frequency during each cycle (Fig2.)

Exposing newly hatched chicks to broody hen vocalizations resulted in approach to the source of sound within the first two minutes of stimulus. Birds in pens with the speaker in the feeder initiated ingestive activity within the first cycle of stimulus while birds in pens with the speaker in a wall spend the first 18 hours approaching the speaker in the wall upon

Fi g.2. Pattern of one cycle of vocalizations of a broody hen.

stimulation and did not initiated any significant ingestive activity. As a result of the treatment birds stimulated from the feeder were heavier than those stimulated from the wall and these, in turn, heavier than the controls. (Fig 3.)

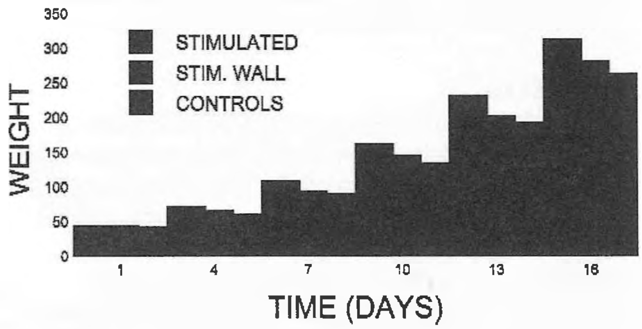


Fig 3. Weight change in chicks exposed to vocalization stimulus over the first two weeks of life

3.3 Swine

The vocalization of sows consist of a sequence of very low frequency tones grunts lasting about 0.1 sec every 0.6-0.8 sec. The sequence of nursing vocalizations last between 2-3.5 min., of which one minute is shown in Fig. 4.

Fig. 4. Sequence of sow vocalizations recorded over one minute.

There was an overall significant treatment effect on the weight of the piglets when the feeding calls were used every 47 and 57 minutes (Fig .5) however, when the frequency of feeding calls was increased to 37 minutes, the response was similar to that observed in the animals of the control group.

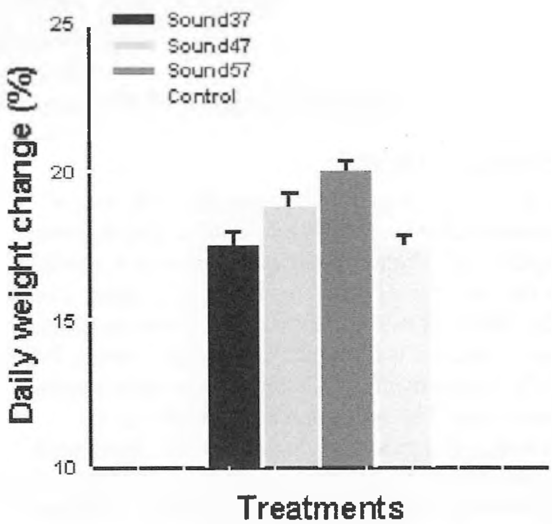


Fig. 5. Performance of piglets stimulated with sound nursing calls at different frequencies.

At a frequency of 47 minutes the piglets in the sound stimulated group performed significantly better than those in the control group. The sound exposed animals grew an average of 2.29% faster than the controls. This value is not only statistically significant but it can be economically significant. At a frequency of 57 minutes there were no differences between the sound treated animals and the concurrent controls

The results to date appear to indicate that the use of feeding calls can enhance growth rate in lactating pigs as long as the frequency of vocalizations is adequate. The behavioural data should confirm if this is due to an actual increase in the frequency of suckling. In preliminary trials it was observed that the response of the piglets to maternal calls was more noticeable when the sound was reproduced by speakers capable of delivering very low frequency. A trial is presently being conducted to determine the frequencies within the vocalization which convey the signal to the newborn animals

4. CONCLUSIONS

It is clear that several food producing species can be manipulated to enhance productivity via auditory signals. These appear to be much more welfare friendly than other traditional methods to increase productivity.

GAP DETECTION WITHIN AND ACROSS AUDITORY CHANNELS IN YOUNGER AND OLDER LISTENERS

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1. INTRODUCTION

A common way to investigate the properties of auditory temporal processing is by means of gap detection. In the standard paradigm the subject is required to decide, which of the two stimuli presented in a trial contains a silent gap. In the target stimulus, a leading and a trailing marker, which can be similar or dissimilar in spectral content, bound the gap. The comparison tone comprises the same markers but without a gap. The sound can be a broadband or bandpass noise, or a pure tone. Several studies have used pure tones as stimuli to investigate gap detection thresholds of young adults. Results show that gap detection thresholds are quite small (2-5 ms) when leading and trailing markers are identical in center frequency. Moreover, gap detection thresholds appear to be relatively independent of center frequency when both markers are similar in spectral content (Moore & Glasberg, 1988). When frequency disparity between leading and trailing marker increases, the task becomes more difficult and the shortest detectable gap can be lengthened to 10 – 50 ms (Boehnke & Phillips, 1999; Lister, Koehnke, & Besing, 2000). Hence, the spectral characteristics of the markers bounding the gap exert great influence on the magnitude of the gap threshold. These results suggest that there is something fundamentally different about perceptual processes involved in within and across frequency gap detection tasks. Several researchers suggested an auditory channel theory to explain the discrepancies in threshold (Formby, Gerber, Sherlock, & Magder, 1998; Phillips, Taylor, Hall, Carr, & Mossop, 1997). According to this theory, identical markers constitute a simple discontinuity detection task within a given auditory channel. In this case, the information can be carried by any or all of the afferent nerve fibers serving that channel. On the other hand, when the markers differ in spectral content, each marker activates different populations of peripheral neurons and higher-order processes are required to compare the relative timing across the different perceptual channels subserving the leading and the trailing markers. These cross-channel timing operations exhibit poorer temporal acuity than within-channel gap detection.

A few studies have investigated the relationship between age and gap threshold. In general, these studies found an elevated gap threshold for older listeners in within channel conditions (Schneider, Pichora-Fuller, Kowalchuk, & Lamb, 1994; Snell, 1997). We are not aware of any study that has examined older listeners' gap detection threshold in the across frequency case. Because audiometric thresholds and gap detection thresholds are uncorrelated, at least in the within channel case, these age effects cannot be attributed to age-related hearing losses (Moore & Glasberg, 1988; Schneider & Hamstra, 1999). Thus, age rather than audiometric loss appears to be the main reason for the significant age effects.

In the present study young and old adults were tested in within and between channel gap detection tasks using pure tones of two different center frequencies.

2. METHOD

Twenty four young (age range 19 to 25 years of age) and 24 older adults (65 years and older) participated in the study. Stimuli were generated digitally with a sampling rate of 20 kHz and converted to analog form using a 16-bit Tucker Davis System (TDS) digital-to-analog converter. The amplitude envelopes for the leading and trailing markers were constructed by summing a series of Gaussian envelopes (standard deviation = 1 ms) whose centers were spaced 1 ms apart to form a flat top envelope with ovigal rise and decay times (Schneider & Hamstra, 1999). The end of the leading marker was defined as the time at which the envelope (expressed in units of power) of the leading marker declined to 0.9 of its peak value. The beginning of the lagging marker was defined as the point in time where the envelope of the lagging marker reached 0.9 of its peak power. Gap duration was defined as the time difference between these two points. Because the rise-fall time of the envelopes (time difference between the .1 and .9 power points) was 2.45ms, tone were only reduced in amplitude rather than being completely turned off at the shorter durations. Two different kinds of comparison tones were used. In the adjusted condition the number of Gaussian envelopes added to construct the standard equaled the combined sum of Gaussians in both markers plus the omitted or reduced Gaussians in the gap. In the fixed condition the comparison tone had a fixed length equaling the sum of the two marker durations. This way, the comparison tone had the same rise and decay times as the leading and trailing markers, but without the gap. One and two kHz tones, aligned in cosine phase with the peaks of the Gaussians, were multiplied by the amplitude envelopes to obtain target and comparison stimuli. Before multiplication the envelopes were normalized so that after the multiplication the total energy in each stimulus was equal to the total amount of energy in the stimulus defined by multiplying two widely separated Gaussian envelopes with the appropriate tone. Thus all stimuli were identical with respect to total energy. The tones, after being multiplied by the marker envelopes were presented at 90 dB SPL to the left ear over TDH-49 earphones in a single-wall sound-attenuating booth. The duration of leading and trailing markers were kept constant within each block but varied in between blocks between 10 and 20ms. A 2IFC method was used to determine the gap detection threshold in each condition. A three-down / one-up adaptive tracking procedure (after Levitt, 1971) was used to determine the 79.4 % point on the psychometric function. Starting gap durations were 32.4 ms in the within frequency and 62.4 ms in the across frequency condition. Marker durations of 10 and 20 ms were used. The within channel conditions used leading and trailing markers of identical frequency (1 or 2 kHz, respectively); the across channel conditions used markers of different frequencies (1 and 2 kHz or 2 and 1 kHz for leading and trailing marker, respectively). Subjects were tested four times under each condition and the four thresholds were averaged to compute a final threshold estimate.

3. Results and Discussion

In agreement with other hearing studies (Moore, Peters, & Glasberg, 1992; Schneider et al.,1994; Snell, 1997) the present study found larger and more variable gap thresholds for older normal hearing listeners in within-channel comparisons. As Schneider & Hamstra (1999) pointed out, the age-related differences in threshold may have been especially pronounced due to the rather short marker durations and may vanish eventually for marker durations over 200 ms. No such age-difference was detected in the across-channel conditions. Gap thresholds did not vary systematically with marker duration in the within or between channel task.

The aging auditory system is characterized by accumulating deficits especially in peripheral processes. Within channel gap detection tasks are assumed to engage peripheral temporal processing mechanisms. The age differences found in this task emphasize once again the gradual decline of the peripheral auditory system with age. Across frequency tasks engage more central processes and thus reveal the developmental course of central mechanisms. In accordance with other recent studies the lack of an age difference in the between channel case is interpreted as showing that older adults appear to be able to compensate for sensory deficits on a more central level and minimize age-related sensory losses (Schneider, Daneman, Murphy, & Kwong See, 2001). Hence, it seems to be most likely that the age-specific impairment for older listeners occurs on a peripheral processing level.

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AGE-RELATED CHANGES IN CAR-DRIVING

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1. INTRODUCTION

Auditory warnings pervade all aspects of car-driving activity. Ninety-one percent of exposure to music occurs during transportation (Sloboda, O'Neill and Ivaldi, 2001). Usually warning sounds call for immediate attention in order to prevent serious mishap (Stanton and Edworthy, 1999). Moreover, it has been found that music tempo consistently affects perceived car-speed estimates. Thus, the tempo of background music affects the frequency of virtual- (Brodsky, 2002) and possibly, real-traffic violations. Without any doubt, car-driving performance depends on ability of a driver to hear warning sounds such as car-horns and emergency siren.

It is known that hearing changes with age with a reduction in the ability to hear high frequency sound components. Thus, when it comes to a decision based on hearing, the elderly will rely on less information than younger adults (Botwinick, 1985). Moreover, cognitive decisions during very complex tasks such as car driving depends on the ability to integrate inputs across many modalities. Thus, car-driving performance depends on the ability to process and integrate information from the various sensory systems and incorporate this information into appropriate decisions. The decision depends on driver's ability to attend to auditory warning sounds (Ison, Virag, and Allen, 2002).

Canada's population is quickly "graying"; a greater percentage of population is becoming more than 65 years old. It is expected that by 2045 year Canada will experience a dramatic increase of older drivers (above 65 years old) relatively to that in 1998 (Caird, 2001).

The present study attempted to explore driving performance as a function of age, type of warning sound, and listening condition. It was hypothesized that:

4. The elderly need higher intensity warning sounds than younger adults in order to prevent serious mishap.
5. The presence of background music (easy listening) should increase the detection threshold of a warning sound in both age groups of drivers.

2. METHOD

2.1. Participants

Twenty-eight older adults (64-85 years old) and twenty-four young, untrained students (21-32 years old) served as participants. All participants had normal hearing (15 dB or better for audiometric frequencies from 500 Hz to 8kHz).

Older participants were paid $10.00 Cdn per hour for their participation. Younger participants obtained a bonus credit towards a chosen course for their participation.

2.2 Stimuli

Stimuli included two spectrally different auditory warning sounds (targets) presented simultaneously with "easy-listening" music and road noise or with the road noise alone. Auditory warning signals had duration of 600 ms (car-horn) or 2500 ms (emergency siren) and intensity level (initially) of 80 dB SPL. Intensity level of the warning sounds subsequently followed an adaptive procedure (3 correct down and 1 incorrect up) to determine the threshold value. The intensity level of noise road alone and road noise mixed with music was always presented to participants at 67 dB SPL Music included three motives from "Cat Tantzu", "Chaos Opera" and "Abyss Project". The intensity level along the duration of three musical motives and road noise was maintained (as much as possible) constant.

2.3 Procedure

Participants were tested individually in an anechoic chamber. Sound was presented binaurally via headphones of flat frequency characteristics. At the very beginning of the each experiment, participants were presented with a practice session until full understanding of the task was achieved. Emergency signals (car horn and emergency siren) were initially provided to the participants at the maximal level of 80 dB SPL by Tucker-Davis Instruments' hardware and software. During the experiment, delivery of the intensity level of stimuli was controlled by Tucker-Davis Instruments' hardware and software as well as customized software.

Participants' answers were collected by a PC computer HP VECTRA.

An experimental session began with a single-track adaptive procedure to determine the 79.4% detection threshold of a target sound in road noise alone. Next, a single-track adaptive procedure to determine the 79.4% detection threshold of a target sound in road noise mixed with the easy listening music was administered. Participants were able to pause and relax anytime during the experiment.

3. RESULTS

Results obtained during experiments were first examined separately, and subsequent analyses of relationships between the experiments were then performed.

Figure 1 shows the results of the study for each age group and listening conditions. In order to obtain these results, a 2 (age group) x 4 (sound listening conditions) repeated measures ANCOVA, using the audiometric averages as a covariate was performed on the threshold values. There was a significant main effect for the age group, $F(1,45)=12.31$ at $p=0.001$ and for the condition, $F(3,135)=100.75$ at $p<0.001$. However, there was no significant interaction between the age group and the target type, $F<1.0$.

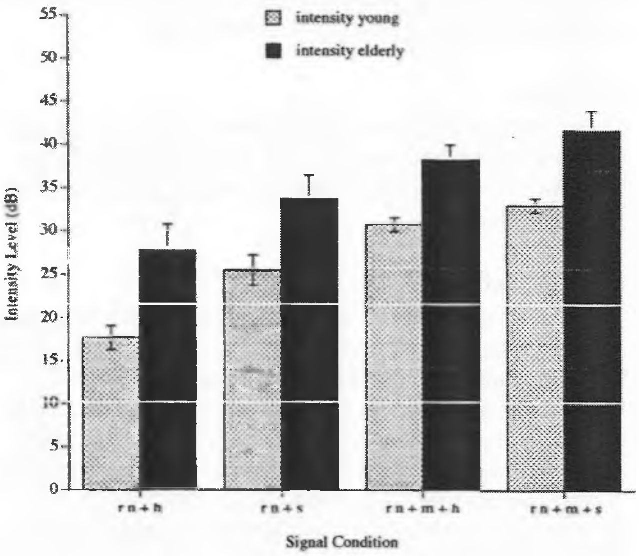

Figure 1. Means and standard deviations of warning sound detection thresholds for both age groups and type of stimuli and four listening conditions.

Thresholds for the various target sounds and conditions revealed the same pattern for both age groups. Elderly drivers demonstrated the large variability (see Figure 1.). Indeed, some of the elderly drivers performed equal to the young listeners, and some of the young listeners had thresholds that were as high as those of some the elderly drivers.

4. DISCUSSION

The hypotheses of the study were supported: Older adults required a higher intensity level of warning signals in all conditions compared to younger drivers. This finding supports other studies showing that the elderly have poorer hearing than young listeners.

Moreover, thresholds for 2 types of signals revealed the same pattern for both age groups and thresholds were higher when listening to music and road noise compared to road noise alone. The same pattern of auditory warning sounds' thresholds processed by both age groups of listeners supports results Ison et al. (2002).

Differences in detection thresholds between car-horns and emergency sirens could be due to the similarity of acoustical aspects of the emergency siren to music and thus, could cause difficulties in detection of the emergency siren when listening to music. Spectral characteristics (noise-type) of car-horn were significantly different from musical motives.

In conclusion, by using changing distracters over time this study suggest that informational masking has a greater impact on the elderly when performing tasks such as driving. These findings of the study indicate the importance of being able to clearly hear the outside environment.

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Age Related Aspects of the Processing of Emotional Language and Detection of a Vehicular Warning Sound: a Preliminary Investigation.

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1. INTRODUCTION

Keeping a driver's mind 'on the road' means more than keeping a driver's hands 'on the steering wheel'. According to the National Highway Traffic Association, there are four basic kinds of distractions, which can occur during driving. These include: visual distractions such as reading a street sign or checking your mirrors; auditory distractions such as corn horns, kids arguing; biomechanical distractions such as manipulating a knob on the radio or CD player or the buttons on a cell phone; and cognitive distractions which involve the mental attention you pay to driving or to something else.

The results of this study represent preliminary findings of a larger investigation involving driving skills, distractions, and age. The general purpose of this initial investigation was to assess a more illusive distraction in driving than those listed above, that of emotion. Human emotion involves an interplay between emotions, physiology, and psychology (Murray & Arnott, 1993). As an initial step this study investigated threshold measurements for a vehicular warning sound in the presence of a single word and a narrative text.

2. METHOD

Experiment 1

2.1 Participants

Eight university undergraduate students (mean age 23.4 years) served as participants in this first experiment. All participants held a valid Alberta driver's license had been driving for at least 3 years. All participants had clinically normal audiograms for the octave frequencies of .5 kHz to 8 kHz.

2.2 Stimuli

Stimuli consisted of 2 single syllable English words (bomb and ball). These words were chosen based on the rating of 8 independent judges who judged a list of words in terms of their emotional or an affectively neutral content. The target stimulus was a car horn of 450 ms. Each of the words was 600 ms in duration.

2.3 Procedure

A two interval forced choice paradigm was used to determine the threshold of the car horn. The participant's task was to indicate via a button push as to whether the car horn occurred in the first or second interval. Each interval consisted of either the word 'bomb' or the word 'ball' and the car horn was always present in one interval. The computer program randomly determined the order of presentation of the words. During the presentation of the stimuli, low level recorded road noise was continuously played during the trials. An adaptive procedure, which determined the attenuation level of intensity, required for threshold of the siren. Maximum intensity level was 90 dB SPL. Thresholds were determined from the last 8 reversals of the attenuation levels for each of 3 presentations.

3. Results

There were no differences in the attenuation levels for either the word 'bomb' or the word 'ball'. The mean attenuation level for 'bomb' was 78 dB and the mean attenuation level for 'ball' was 80 dB.

4. Discussion

The results of this experiment indicate that even if listeners perceived the words as having a different emotional content, that this was insufficient to have any impact on the threshold of a car horn. Greasley, Sherrard and Waterman (2000) have shown that valency ratings for the emotional aspect of words depends upon presenting them in a context. A second experiment was conducted to determine if the thresholds would be affected if the target stimulus was embedded in a full narrative of text.

1.2 Experiment 2

The results of experiment one indicated that a single word without context is insufficient to produce any effect. Experiment 2 was conducted to determine the impact of a narrative text on the threshold for a vehicular warning sound.

2.2 Method

2.2.2 Participants

7 listeners participated in this study. The youngest group of listeners (n = 4, mean age = 26.5 years) were recruited from the University of Calgary undergraduate population. The older group of listeners (n = 3, mean age = 57.7 years) were volunteers from the outside community. All participants had clinically normal hearing for the octave frequencies of .5 kHz to 6 kHz though the older group of listeners had higher thresholds relative to the younger listeners.

2.2.3 Stimuli

Two passages of text chosen to represent 'emotional content' and 'neutral content' were a passage on the life of Dimitry Shostakovitch (emotional) and a passage by Thoreau (neutral). These were recorded by a male speaker, who spoke both passages in a neutral tone of voice.

Three separate listening conditions were tested: two where the listener attended to the narrative, and one where the listener merely had to listen for the presence of the target stimulus. These are referred to as: neutral attend, emotion attend, and emotion not attend. To ensure that listeners would focus on the content of the narrative, they were informed that they would be asked some simple questions about the content of the passage at the conclusion of the experiment.

2.2.4 Procedure

The method is the same as described in experiment one except for the presence of a continuous narrative of text, which played for a possible total duration of 30 minutes. In this experiment, participants were instructed to attend to 2 lights, which flashed above each of 2 buttons on a response box. First the light above the button on the left flashed followed by the button on the right. The listener's task was to depress the button corresponding to the light flash that occurred with the car horn. The target stimulus was always present in either the first or the second interval though the computer program randomly determined which interval.

3.2 Results

Table 1 shows the mean attenuation level for each listening condition for each age group. There appears to be no effect for the 'content' of the passage (neutral attend versus emotion attend). Average thresholds were lower for the attended condition than for the non-attended conditions (greater attenuation levels mean lower thresholds). Additionally, there is no age effect for the non-attended listening condition.

Table 1. Mean attenuation levels and standard deviations (in parentheses) for each age group of listeners for all 3 listening conditions.

	Neutral attend	Emotion attend	Emotion Not attend
Young	73.78 (5.8)	71.6 (6.1)	68.19 (9.78)
Older	68.9 (.98)	61.4 (3.4)	67.64 (2.46)

4.2 Discussion

The lack of an effect for the content could possibly be explained by the fact that both passages were read in a neutral tone of voice. The important factor appears to be whether or not a listener is required to attend to the information. Furthermore, the judgement of what is emotional content is a highly individual process (Vakoch & Wurm, 1997). Further studies are examining the role of the intonation contour; the content of the information, and whether the listener is required to interact with the information.

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Age related changes in glide discrimination and its relation to the ERB

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1. INTRODUCTION

Various psychophysical phenomenon concerning the detection of changes in signal features can be understood in the general framework of an excitation-pattern (EP) model (Moore & Sek, 1994). The EP of a tone is its representation in a bank of tonotopically organized auditory filters and is a reflection of the activity in several overlapping filters (Demany & Clement, 1997). Moore and Sek (1994) have advanced the argument that if discrimination is below the phase locking capabilities of auditory fibers, then thresholds should be based on 'place' cues and be a constant proportion of the ERB. If it is assumed that in the process of glide discrimination, a listener compares signals by comparing the differences between their start and endpoints, the accuracy with which this change can be measured would be determined by the resolution of the location of these points along the basilar membrane. This in turn would be a function of the slope of the excitation pattern, which in turn is determined by the bandwidth of the auditory filters centered at and just below the test frequency (Moore & Sek, 1998). The purpose of the present research was to investigate whether frequency glide discrimination could be explained in terms of an EP model.

2. METHOD

2.1 Participants

143 persons ranging in age from 20-75 years of age were selected for participation. The final group of listeners represented five decades as follows: 20-34 years, n = 30; 35-44 years, n = 30; 45-54 years, n = 28; 55-64 years, n = 25; and 65-75 years, n = 30. The two youngest age groups were recruited from the University of Calgary and the outside community. Other age groups were volunteers from the outside community. Each participant had pure tone thresholds no greater than 20 dB HL for the frequencies of .5kkHz to 8kHz and normal middle ear status.

2.2 signals

Glide signals were created at 2 different frequency regions: one centered slightly above 1kHz (low frequency region) with a center frequency (CF) of the maximal total glide excursion of 1030 Hz; the second series was slightly above 2500 Hz (high frequency region) with a CF of the maximal glide excursion of 2685 Hz. Two different patterns of glide trajectory: upward or downward, and end frequency conditions: diverging or converging, were constructed. Background noise was presented for only the converging upward and diverging downward series of signals at each frequency region for a total of twelve discrimination conditions. The following abbreviations will be used to describe the signals: DU (diverging up); DD (diverging down); DD-N (diverging down in background noise); CD (converging down); CU (converging up); and, CU-N (converging up in background noise. For all series the slowest changing signal in the series was designated as the 'standard' and was used as the comparison stimulus in all pairings.

For each frequency continuum, a series of 17 gliding signals were created. At the low frequency region, signals spanned a maximal excursion of 900 Hz to 1110 Hz. At the higher frequency region the maximal glide excursion was 2685 Hz. Signals were constructed such that the CF of each glide was roved over a range of frequencies. At the low frequency region, A CF spanned a range of 80 Hz from 1005 Hz to 1085 Hz. At the high frequency region, A CF spanned a range of 240 Hz. Discrimination thresholds were determined under two listening conditions: quiet and in the presence of speech spectrum noise. The noise was low pass filtered at 1kHz and had a 10dB per octave roll off to simulate the long term spectral characteristics of speech and was presented ipsilaterally at a spectrum level of 35 dB. Stimuli were delivered monaurally to the right ear through Kros Pro/4x headphones at a presentation level of 60 dB SPL. The noise was played continuously during the background noise condition.

2.3 Procedure

A two-alternative forced-choice (2AFC) paradigm was used to determine the smallest discriminable differences that individuals could determine. Each trial consisted of two stimuli with an inter-stimulus interval (ISI) of 500 ms and listeners responded 'same' or 'different' via pushing one of two buttons on a board. Each continuum of signals at both frequency regions was presented in blocks of 370 trials. Each trial consisted of the standard stimulus and one of the other 16 signals in the series with the standard being either the first or the second member of the pair an equal number of times. Each series of signals were tested within one block though the order of experimental trials within each block was randomized for each participant.

3. RESULTS

Percentage correct data were then transformed into probit scores. Thresholds were measured from the 70% correct position on the psychometric function. The width of the ERB for the center frequencies of both frequency region were computed based upon Moore and Glasberg's (1990) equation: ERB = 24.7(4.37F +1). The ERB for the CF at the low frequency region is 133 Hz, and the ERB for the high frequency region is 314.5 Hz. Threshold values were expressed in Hz/ ERB for each signal condition at each frequency region. As the extent of the signal was variable within each signal condition (to keep duration and either the endpoint or the starting frequency constant), signals are not a constant proportion of the ERB. However, the standard signals in the low frequency region spanned .38 ERB, and the standard in the high frequency region spanned .48 ERB. Table 1 shows the threshold values expressed on a scale of auditory filters (ERBs).

Table 1. ΔF values expressed on a scale of auditory filters (ERBs) for each age group (top row).

	20-34	35-44	45-54	55-64	65-75
	L H	L H	L H	L H	L H
du	.26 .37	.27 .38	.32 .42	.33 .37	.38 .54
cu	.30 .36	.32 .37	.32 .48	.35 .53	.42 .82
cun	.33 .42	.41 .43	.43 .52	.42 .62	.47 .81
dd	.25 .32	.25 .32	.35 .38	.37 .34	.34 .47
ddn	.28 .35	.32 .37	.31 .29	.38 .38	.37 .50
cd	.44 .51	.45 .50	.47 .54	.49 .54	.43 .69

4. DISCUSSION

When thresholds were expressed as Hz/ERB the proportions ranged from fairly constant at .3 for the 20-44 year olds across conditions and frequency regions. However, values for the 45-74 year olds ranged from .3 to .5 within each signal type for each group. These values are higher than the values of Moore and Sek (1998) and Madden and Fire (1997) for their 50 ms signals with a transition span of .5 ERBs at 2 kHz for the youngest groups of listeners. The proportions are considerably higher for the eldest group of listeners. The increase in the proportions in this study could be a reflection of experience. In the studies by Moore and Sek (1998) and Madden and Fire (1997) listeners were practiced for 10-15 hours before data collection. Therefore, their values probably represent asymptotic performance of discrimination.

In summary, these data do not support models of glide discrimination which predict an enhancement for an upward sweep, nor do they support models of cochlear dispersion cues based on an increment in frequency targets. The most parsimonious explanation is that listeners in this study were relying on terminal frequency pitch cues for discrimination. However, the ΔF/F values are quite large. Weber fraction equivalents of the values in Table 1 range from .3 to .6 which are considerably larger than ΔF/F values for equivalent pure tones (Wier, Jesteadt, and Green,1977) or for glide detection (Dooley & Moore, 1988).

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AGE-RELATED CHANGES IN INFORMATIONAL MASKING

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1. INTRODUCTION

In spite of noisy backgrounds, people can listen to an important sound and reject irrelevant, distracting information. In order to separate relevant from irrelevant information during simultaneously-presented sounds, people have to focus their attention on the target and ignore the distracter. Various theories or models attempt to explain auditory information processing when attention is engaged. All of them refer to informational masking (e.g. Allan and Whitman, 1995; Slawinski and Scharf, 1998; Ison, Virag, Allen, and Hammond, 2002).

The present experiments measured the effect of a masker's presence (distracter) and its uncertainty on detection of a target sound. Results of previous work performed by Neff, Dethlefs and Jestead (1995) have shown that when the frequency content of a multitone masker (distracter) changes, detection of the target sound deteriorates markedly. Previous work by Slawinski and Scharf (1998) focused on tonal stimuli and suggested that low intensity, uncertain, randomized distracters (four) were able to increase a target's threshold in a similar way for older and younger participants.

The present study augmented the previous study by Slawinski and Scharf (1998), while incorporating a few changes:

a. Two distracters were presented only instead of four distracters.
b. Speech syllables were presented, in addition to tonal stimuli.

In order to explore age-related differences in processing important sounds (targets), two experiments were designed. The two experiments differed in the type of target sound used, distracters, as well as background noise. Each experiment was designed to find the detection threshold of the target sound (syllable-ga or pure tone-1kHz). Results of both experiments were obtained using an adaptive procedure. It was hypothesized that:

1. Target detection thresholds will be a function of age.
2. Participants will show different change in threshold as a function of age depending on a type of presented stimuli.

2. METHOD

2.1. Participants

Twenty-eight older adults (64-90 years old) and twenty-five young, untrained students (21-32 years old) served as participants. All participants had normal hearing (15 dB or better for audiometric frequencies from 500 Hz to 8kHz) and were free from any neurological problems.

Older participants were paid $10.00 Cdn. per hour for their participation. Younger participants obtained a bonus credit towards a chosen course for their participation.

2.2. Stimuli

All stimuli were synthesized. The speech-like stimuli (syllables) were synthesized using software KLSYN 88a implemented on a PowerMac computer. Intonation contour of speech-like stimuli syllables imitated naturally spoken syllables. However, the intonation contour was the same for all type of syllables.

The pure-tones were generated on-line by the Tucker Davis Technology modules and software implemented on a Vectra PC computer. The targets' intensity changed according to an adaptive procedure, while the distracters' intensity was maintained constant during both experiments.

2.3 Procedure

Participants were tested individually in an anechoic chamber. Sound was presented binaurally via headphones of flat frequency characteristics. At the very beginning of the each experiment, participants were presented with practice sessions until full understanding of the task was achieved. Target sounds were initially provided to the participants at the maximal level of 80 dB SPL by the Tucker-Davis Instruments' hardware and software. During the experiments, delivery of the intensity level of stimuli was controlled by the Tucker-Davis Instruments' hardware and software as well as a customized software.

An experimental session began with a single-track adaptive procedure to determine the 79.4% threshold in background noise alone, separately for the targets and each of the distracters. Next, a two-track interleaved adaptive procedure was used to determine the 79.4% thresholds for targets in the presence of each of the two corresponding distracters. On a given trial, each distracter was a different distracter in each interval. Once the threshold had been determined with a given distracter, that distracter was no longer presented with the target's sound but could be presented in the other, noise-only interval.

Background noise varied depending on the experiment. The background noise was a cafeteria noise during presentation of speech-like stimuli and a band-pass white noise (300 Hz to 1800 Hz) for pure-tone stimuli. Background noise was presented at 60 dB SPL.

Participants' answers were collected by a PC computer HP VECTRA.

Participants were able to pause and relax anytime during the experiments.

3. RESULTS

Results of both experiments are presented together. Detection thresholds of target sounds of the elderly were compared to those of young participants.

When differences in hearing were not taken into account, the effect of age was significant in detecting targets in the presence of the corresponding distracter. This difference disappeared when hearing was taken into account $F(1,51)=.41$ at $p=0.525$.

The MANCOVA, using the audiometric averages as a covariate was performed on the detection threshold values. Younger and older adults showed similar performance patterns (means and SD for finding the syllable and tonal target). However, both age groups demonstrated higher detection thresholds when speech sounds were presented compared to that of tonal sounds ($F(1,51)=27.97$ at $p=0.000$.

Thresholds were also higher for detection of the syllable [ga] in a presence of the syllable [da] than in presence of syllable [ba]. The detection of tonal stimulus was poorer in presence of high frequency distracter (2 kHZ) than in presence of low frequency distracter (500 Hz).

4. DISCUSSION

Our hypotheses were partially supported by the findings of this study. Older listeners required more intense targets in order to detect it compared to young participants. Secondly, both age groups followed the same pattern of performance.

Thus, it seems that older people are probably making more errors when detecting the syllable [ga] than young adults. Moreover, the syllable [ga] acoustically is more similar to the syllable [da], thus, it was not surprising that all participants had more problems to detect the target [ga] in presence of [da] distracter.

The results of the study by Slawinski and Scharf (1998) that showed that detection of a tonal target is more influenced by the presence of a higher frequency distracter than the target itself, were also supported. However, currently it is difficult to provide an explanation for this effect.

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Absolute identification of azimuthal sound location and judgment of auditory circular direction.

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1. INTRODUCTION

As reported at the last CAA-ACA annual meeting, a program of research on auditory circular direction has revealed that the proximity between audio sources determines whether a sound appears to move from one audio speaker to another in a clockwise (CW) or counterclockwise (CCW) direction. More specifically, the directed vector takes the shortest distance between the two sound sources. A similar finding in the pitch domain has been observed by Shepard, 1964. In addition, as is typical of localization data, front-back confusion is often prevalent, as early described by Toole (1970). Thus, if listeners tend to hear sounds from only the front or back hemisphere, tones from sources in the ignored hemisphere will be re-located to the mirror-imaged position in the preferred hemisphere. This conception was consistent with three proposed models of directional judgments. The Circular Model assumed that listeners could hear sources in 360 degrees. The Front Model assumed that listeners heard sources located in the front hemisphere. The Back Model assumed that the listeners heard sources located in the back hemisphere. The trajectory data were fit best by one of the models, however, the best-fit could only be determined post priori. The aim of the present study was to predict the best-fit model a priori on the basis of the ability to locate individual sources in an absolute identification paradigm.

To the best of the authors' knowledge, the present study is the first to relate absolute identification results to judgments of trajectories (CW/CCW directional judgement) created by separate audio sources. Listeners were tested in absolute identification of the location of 12 circumcranial sources and subsequently carried out the task of direction judgment of all possible pairs of the 12 sources. To determine the stability of the individual location judgments, the absolute identification task was then repeated. For each listener, errors in absolute identification provided evidence of their degree of front-back confusion. Error patterns should help predict the applicability of the previously developed Circular, Front, and Back theoretical models of the trajectory data (Cohen et al., 2001). Results of the analysis would confirm one of two possible outcomes. The first is that the absolute identification error pattern for each listener will provide an a priori basis for predicting which of the models is most appropriate for the directional data of that listener. The second is that the two different tasks, absolute identification of individual sources versus the task of direction judgement of a pair, engage different spatial auditory processes.

2. METHOD

Subjects. There were 16 male and 16 female subjects ranging from 18 to 42 years of age. One half of each of the gender groups had more than 4 years experience playing a musical instrument. Hearing level, tested with Digital Recordings AUDIO-CD™ was within normal limits in the range 1 - 4 kHz.

2.1 Apparatus.

In a single-walled sound-attenuated room (Eckel), 12 small Koss speakers (12 x 8 x 8 cm) were spaced at intervals of 30-degrees around an azimuthal circumference of the largest circle (diameter 119 cm) that could be accommodated by the room. The speakers were 1.5 m off the floor, roughly at ear level for an individual seated in the centre of the circle. A multiplexing switch directed an audio signal to one of the 12 speakers. The signal was a complex tone composed of 10 octaves of 22.5 Hz with an envelope that approximated a Gaussian function. Each signal was 250 ms in duration.

2.2 Procedure.

Listeners were tested individually, seated centrally within the circumference of the 12-speaker array, facing a corner of the room subtending speaker 1 and interacted with a computer screen using a cordless mouse.

<u>Part 1. Absolute Identification.</u> Listeners were presented with 11 blocks of 12 trials. Each block consisted of presentation of the tone from each of the 12 speaker locations in a random order. The listener was required to indicate from which location the sound was emitted by moving the cursor to the analogous position represented on the computer screen. Testing time was approximately 10 min.

<u>Part 2. CW/ CCW Direction Judgments.</u> Each listener was presented with a block of (12 x 11) 132 pairs of successive tones, such that all possible successive pairs of the 12 speakers were represented. The intertone interval within a trial was 450 ms. On each trial, the listener judged the direction of the sound around his or her head represented as CW or CCW on a computer screen. The trial block took about 10 min. There were 3 successive blocks in a session, such that each pair was represented 3 times for a total of 396 trials in approximately 30 min.

<u>Part 3. Absolute Identification</u>. Part 3 repeated the procedure of Part 1 for 10 blocks of trials.

3. RESULTS

3.1 Absolute Identification.

Fig. 1 illustrates the mean per cent correct for each tone for all listeners for Parts 1 and 3. Performance is poorest for the speakers behind the head and performance improves on the 2nd block. Overall improvement (mean % correct on the first set was .52 and on the second set .65) and selective improvement on the back speakers was also noted. The mean percent correct for each speaker for the pre- and post- test were entered into a repeated measures ANOVA having one factor of speaker position and one factor of time (pre/post). The effect of location was significant, F (11, 330) 42.66, p < .0001, with a strong quadratic trend, F (1, 30) 83.77, p < . 0001.

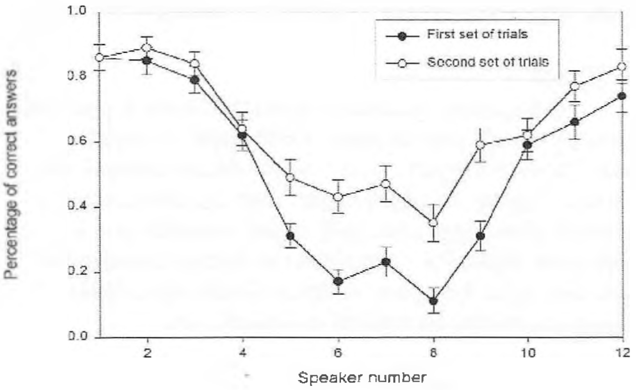

Figure 1. Mean % correct identification as a function of location and session (speaker 1 is directly ahead of listener).

There was a significant interaction between repetition and location, F (11,330) = 5.94, p < .0001. There were no significant effects of musical training or sex of listener, but mean performance for males (.63) exceeded that of females (54) (p < .07).

3.2 Part 2. Direction as a function of CCW rotation.

As in previous work (Cohen et al., 2001), three models were applied to each of the 3 sets of 132 trials per listeners and to the total 396 trials per listener.

The best-fit models were distributed among Circular and Front with a tendency for increased applicability of the Circular Model with time in the task (applicability of the Cirucular model was 13 of 32 listeners for Block 1, and 16 for Block 3). The size of the correlation increased with block, indicating a stabilizing of the listener's hearing strategy.

3.3. Prediction of Best-fit Trajectory Models From AJ data.

It was reasoned that those listeners who did well on the front speakers but poorly on the back speakers could be the same listeners whose trajectory data were best fit by the Front model. Similarly, those who did well on the front and back speakers in the absolute identification (AJ = absolute judgment) task would be those for whom the Circular Model was the best fit. To test this hypothesis, predictors of the best-fit trajectory model were developed from the AJ performance data as follows: performance on 12 speakers, speaker 1, speaker 7 and then all groups of speakers symmetrical round these two points: the 3 front, 5 front, and 7 front and the 3, 5, and 7 back. Only the back speakers proved to have high absolute correlations with the model values, positive correlations with the Circular model and negative correlations with the Front model. Predictions were derived separately for the two blocks of absolute judgment data. Higher correlations arose from the second set of data, and the highest correlations were for the 3rd set of trajectory data. The first set of AJ data correlated most strongly with the first set of trajectory data (Block 1).

Correlation of 2 of the AJ predictors with Circular Model Fits				
	Set 1 (pre)		Set 2 (post)	
Predictor	Back 5	Back 7	Back 5	Back 7
Block 1	**.55**	.52	.63	.58
Block 2	.44	.44	.73	.67
Block 3	.40	.43	**.83**	.80
All 3	.47	.48	.80	.75

Multiple recognition of the separate predictors on the Circular and Front correlations raised the predictability but not dramatically. The superior applicability (yes or no) of Circular over the Front model was predictable for 87.5 % of the listeners (28 out of 32).

4. DISCUSSION

Earlier, perceived direction of auditory motion has been shown to be influenced by proximity of sound sources. The present study adds to the picture by verifying that tendencies to hear sounds in one contribute to these direction judgments, in particular the applicability of a circular or front or back model. In terms of the initial question, it is clear that auditory-location identification data and auditory circular direction judgments tap similar underlying processes, at least for most listeners.

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TIMBRAL SEGREGATION OF SIGNALS AND THE AUDITORY ATTENTIONAL BLINK

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1. INTRODUCTION

This study serves as a continuation of previous inquiries by Slawinski and Goddard (1999: Goddard and Slawinski, 2001) into the nature of the Auditory Attentional Blink (AAB). AAB effects are seen experimentally when dynamic sound streams are presented to participants who are instructed to attend and respond only to those sounds within the stream that meet some criterion (e.g., those that are louder than the others). A "blink" is said to occur when two to-be-attended (TBA) signals are presented in rapid succession and are reported as a single signal. The effect does not occur due to energy masking, but rather due to some shortcoming in sensory or cognitive processing. Supposedly, one signal's processing is not completed due to the processing of the other signal. The former is referred to as the Probe (P), and the latter as the Target (T). It is unknown precisely what causes this effect, what physical parameters are necessary to produce the effect, and whether or not the effect is similar in its causes to the Attentional Blink in vision, about which far more is known.

A typical blink task will involve the presentation of several streams of stimuli, some of which contain P alone, and some of which contain T and P. Increased failure to report P in the presence of T defines the blink effect. In visual research, this is typically observed when the onsets of T and P are about 450ms apart (e.g., Raymond, Shapiro & Arnell, 1992). The criterion that is used to segregate TBA signals from the rest of the stream may be varied between blink tasks, and the results of this variation are as yet unexplored. Previous studies have typically employed intensity as the criterion, yielding a clear AAB effect (e.g., Slawinski and Goddard, 2001). The current study employs spectral characteristics as the criterion, and is aimed at further clarifying parameters necessary for producing the AAB in hopes of learning more about possible causes for the effect. Specific aims of the study will be to determine what kind of a blink effect (if any) this streaming task will produce, to explore possible differences between TBA-signal identification and mere detection tasks, and to look for learning effects in the AAB phenomenon. Learning effects are expected, since the blink does not seem to significantly affect our perception of speech– a rapidly changing signal and a likely candidate for the AAB effect. Learning effects may help explain this.

2. METHOD

2.1. Participants

29 volunteers participated in the study. All were screened and met criteria for normal peripheral hearing (at a level sufficient to discriminate between sounds being used in the study). Participant age ranged from 19 to 51 years. Data from seven participants were not included in analyses, mainly due to their apparent misunderstanding of the task.

2.2. Stimuli

Participants listened to several streams of pure-tone distractor signals with complex TBA signals embedded among them. All signals were comparable in intensity and envelope. Signals that participants were asked to attend to consisted of an organ and a bell sound sampled from a synthesizer, while filler signals in the streams consisted of tones varying in frequency between 200Hz and 2.5kHz. Streams consisted of a total of 16 sounds each.

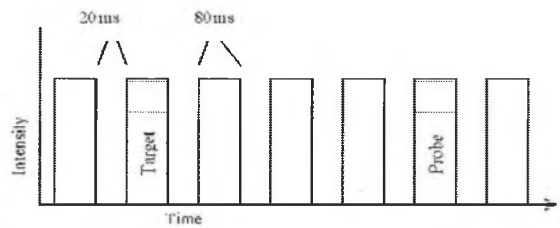

Figure 1. Diagram of a portion of one experimental signal stream. Inter-stimulus intervals and signal durations are shown. Variations in timbre are represented by variations in colour.

2.3. Procedure

Example streams were played for participants until they demonstrated that they understood the task and that they could accurately discern targets from the filler sounds.

Each participant listened to 80 streams during their trial. Half of the participants were asked to simply indicate how many targets (zero, one, or two) they had heard in each stream, while the other half were asked to identify the sounds they heard (the sounds were given labels before these participants' trials, which were taught to the participants). 40 streams in each trial contained only P as a control measure (a measure of the likelihood of a participant's reporting P in the absence of T), while the other 40 contained both T and P (a measure of the effects of the presence of T on the report of P). Delay between the onset of T and the onset of P in experimental streams (those containing both P and T) was also varied between 100 and 500ms in 100ms steps, a variable known as Stimulus Onset Asynchrony (SOA). Streams of varying SOA and condition (experimental/ control) were played in random order. As a final independent variable, each participant's performance on streams from the first half of each trial (20 experimental and 20 control streams) was compared with performance in the second half of each trial (20 experimental and 20 control streams) to assess learning effects.

3. RESULTS

A mixed-model ANOVA revealed significant main effects for SOA and Condition (control versus experimental), as well as a significant learning effect. Significant Learning by Condition and Condition by SOA interactions are shown graphically below.

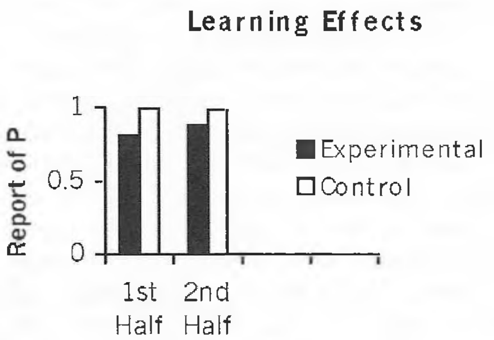

Fig. 2. Significant learning effect. X axis shows 1st v/s 2nd half of trials performed by each participant, Y axis shows the likelihood out of one of reporting P when presented with (black bar) and without (white bar) T. Performance in experimental trials improved significantly for the second half of trials performed. SOA of individual streams was balanced between the two halves of each trial.

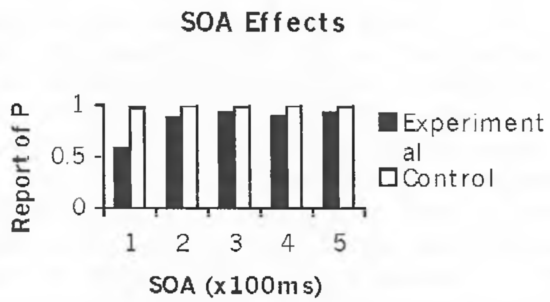

Figure 3. Significant blink effects were seen at SOAs 1, 2 and 5. Y axis shows the likelihood out of one of reporting P when presented with and without T.

4. DISCUSSION

As is shown in Fig. 2, a blink effect was observed at SOAs 1, 2, and 5. The lack of statistically significant effects at SOAs 3 and 4 was most likely due to lack of statistical power and increased variance within these SOAs. These results suggest that the AAB is robust across streaming tasks, although further manipulation of streaming tasks may be necessary to determine this with more certainty.

The lack of an effect of task demands (identify versus detect) suggests that the root of the AAB is at some step in signal processing common to both processes, as they were carried out in this experiment specifically. It is possible, however, that this result is merely an artifact of participants carrying processing of signals through to an identification in spite of the fact that they had not been taught labels for the TBA signals in the "detect" condition- i.e., participants may have attached their own labels to these sounds and attempted to discriminate them from each other even in this condition. Further research is required to determine which common processing step is responsible for the non-effect, and to determine whether or not it was also an artifact of a lack of statistical power in this study.

Learning effects (i.e., amelioration of the blink after repeated exposure to stimuli) observed in this experiment provide an interesting new basis for study– the AAB is evidently not an absolute limitation of the system, rather it is caused by limitations that are resolved as stimuli become more familiar. One candidate for such a process may be streaming itself; auditory scene analysis seems prone to learning effects (Bregman, 1990), and it has been demonstrated in previous Attentional Blink research that easier streaming tasks (i.e., those involving more exaggerated differences between TBA and not-TBA signals) yield weaker blink effects (Raymond et al. 1995).

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BETTER ANALYSIS FOR AUTOMATIC SPEECH RECOGNITION

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1. INTRODUCTION

Automatic speech recognition (ASR) of speech appears at first glance to be a simple task. Commercial systems often claim to do ASR reliably, but they usually require high-quality voice input and often impose many restrictions on what is said and how speakers talk. In addition, recognition of speech in poor acoustic conditions is often unreliable. Another consideration is that of computer resources. When ASR is done at a central computer, where system speed and memory is less of a concern, this latter issue may not be so important, but when ASR occurs in portable devices with limited power and memory, minimization of resources becomes important. We discuss in this paper an efficient ASR analysis method, which applies to adverse acoustical conditions.

2. BACKGROUND

A major problem for most ASR systems is robustness: they often are insufficiently general or are over-trained when furnished with small training sets (as typically happens in many practical cases). An ideal robust ASR system should be able to properly decode speech from all speakers of a language (e.g., English), in any reasonable environment, and with different microphones and transmission channels. In practice, instead, environmental noise (from natural sources or from machines) and communication distortions in transmission channels (e.g., static, fading) both tend to degrade ASR performance, often severely. Human listeners, by contrast, usually can adapt rapidly and successfully to most such difficulties. This large difference between machine and human speech recognition performance strongly suggests that major flaws exist in current ASR schemes. In particular, much of what the scientific community knows about human speech production and perception has yet to be properly integrated into practical computational ASR.

The speech signal must be regularly converted to a representative, small set of parameters or features, in order to efficiently and reliably intrepret the audio signal (input to ASR) in terms of phonemes and words. The most common analysis method for today's ASR is the mel-frequency cepstral coefficient (MFCC) approach [1]. In a first step, either an FFT (fast Fourier transform) or LPC (linear predictive coding) spectrum is obtained using each speech frame as successive input. Then, for each frame, the logarithm of the amplitude spectrum is taken (converting to the decibel scale). Thirdly, a set of about 20 triangular filters, spaced according to the perceptual mel (or bark) scale, weights this result, yielding a simple set of 20 output energies. Finally an inverse FFT using the 20 energies as input is performed [2]. The low-order coefficients (e.g., 10-16 in number) of this last step provide the spectral vector for ASR use.

Among the advantages of this standard approach are the following: 1) an automatic and efficient method, which needs no controversial (i.e., error-risking) decisions, 2) actual ASR results that appear to be better than with some other methods extensively examined in the past (e.g., basic LPC, or a filter bank), and 3) an elegant mathematical interpretation of the MFCCs as somehow decorrelated (because the inverse FFT uses orthogonal sinusoidal basis functions). Despite their considerable popularity, however, the MFCCs are suboptimal:

1) The fourth step of the MFCC calculation (the inverse FFT - effectively low-order cosine weightings of the log spectral weighted energies) is motivated almost entirely on mathematical grounds, rather than communicational or scientific reasoning, which has led to representing spectral information for speech in a very convoluted way. The first output coefficient (C0, which uses a zero-frequency cosine weight) is simple energy, hence easy to interpret and utilize (if desired). The second (C1, using a cosine whose spectral period comprises the full frequency range) is thus a spectral parameter which indicates the global energy balance between low and high frequencies (the initial, positive half of the cosine weights the lower half of the frequency range positively, and vice versa for the upper range). Thus, the first two coefficients are useful and subject to easy interpretation. However, all the other MFCCs are very difficult to relate to major aspects of speech production or perception. For ASR purposes, it is not essential that the parameters used be physically interpretable, but the MFCCs must be used altogether, in order to exploit the fact that they contain increasingly finer spectral detail (as the order increases). Although most individual MFCCs have little clear meaning (in terms of acoustics, the vocal tract shape, or phonemes), used together they can discriminate different sounds. Unfortunately, their lack of direct or simple correspondence to speech production and perception means that they are very vulnerable to degradation when speech occurs under non-ideal acoustic conditions, such as in noise or with speakers having foreign accents.

2) It has often been posited that the MFCCs are uncorrelated in some sense, owing to the orthogonal basis functions used in the inverse FFT [2]. It is quite evident, however, that the MFCCs contain much overlapping spectral information, which causes the covariance matrices of their joint probability densities, as used in ASR, to be far from diagonal. This in turn leads to poor modeling assumptions in many ASR

applications which do indeed often assume diagonal matrices (to cut computational costs), or to significantly increased computation to handle large general matrices [2] (for the minority of cases that use full-covariance matrices). As model order for MFCCs increases, of course, these matrices grow in size proportionately. Requiring 10-16 parameters (or more) becomes increasingly expensive in storage, computation, and training. The correlation of MFCCs is easily seen in the example that both C0 and C1 share large positive values for vowels and are negative for fricative phonemes.

3) Different speakers (especially those with different accents) exhibit varying spectral patterns when uttering (what listeners interpret to be) the same phoneme. Many of the variations often have simple interpretations acoustically and linguistically; hence adaptation to such variation should not theoretically present such a large problem. In practice, the ASR field spends much research on adaptation issues, both those due to varying speakers and to varying transmission channels. If we employ spectral parameters that have a ready physical interpretation, it is feasible to model accent and channel variation simply. However, the lack of such ability to interpret the MFCCs usually forces ASR to employ "brute-force" methods, e.g., simple averaging of distributions to handle different speakers and channels. Such merging of data models often leads to increased variances and hence to lowered discriminability against other, incorrect phoneme models. For these and other reasons, the MFCCs should not be considered as the ideal speech analysis tool, despite their recent popularity.

3. ALTERNATIVE SPECTRAL MEASURES

In the early stages of serious ASR work, formant frequencies were considered the primary objectives of speech analysis. Expert system approaches to ASR abounded then, and formants were widely accepted as the obvious targets for speech analysis. Unfortunately, the automatic formant estimation methods of the 1970s (needed for "automatic" speech recognition) failed to achieve sufficient accuracy. Formants were difficult to follow reliably, as they often approached each other close enough to be viewed as a merging (in spectral displays, such as the FFT) and the formants varied widely in amplitude as a function of time.

We do not propose yet another attempt at formal formant trackers, for two reasons: 1) the continued difficulty of formant tracking, and 2) formants (as such) are not required for ASR. In our opinion, it was an error for ASR researchers to insist on a strict formant tracker as a separate module for ASR. Indeed, robust spectral measures better than MFCCs are quite feasible based on spectral peaks similar to formants, and this is where we propose to raise ASR accuracy. When faced with increasingly noisy speech (as is found in many practical ASR conditions), the peaks of such speech spectra are the most robust (i.e., the last aspects to be lost as noise grows). More robust ASR is thus possible by directly exploiting peaks, rather than approaches that deteriorate quickly in noise (e.g., MFCCs or LPC).

Trying to consistently track all the formants was a mistaken and unnecessary task for ASR. Instead, identifying the major spectral peaks in an utterance and their gross temporal dynamics are what appears to be crucial for ASR. In other words, coarse detail about spectral peaks is important; precise tracking of formants is not. We do not need formants identified as F1, F2 and F3 for all speech frames. Instead, we propose a spectral-peak-based analysis measure which can be simultaneously robust, informative, and efficient. Such a measure needs as few as six coefficients to represent the main spectral peaks (three center frequencies and their coarsely-measured bandwidths), and thus is clearly more efficient for ASR than either LPC or the MFCCs.

For noisy telephone digit strings, our method can achieve good recognition rates, without requiring the complexity of full mel-cepstral evaluation and avoiding the large search calculations of a full HMM approach. As noise levels are increased, the weaker portions of the telphone-band spectrum are increasingly obscured, but sufficient information remains concerning the spectral peak positions of the lower formants to allow digit discrimination, even in significant noise. Mistakes confusing 5 and 9 are common when the noise obscures most of the consonant energy in those digits, although the coarticulatory effects of the consonants (labial in 5 and alveolar in 9) permit some discrimination even when the consonants are fully obscured. Allowing a comparison focussed on critical frames at the ends of the vowel (rather than a uniform frame-based method) permits better utilization of the speech energy in the presence of noise. More details of the results will be presented at the conference.

4. CONCLUSION

A case can be made that the current HMM-MFCC approach to ASR has sufficient flaws as to need eventual replacement. Certainly the persistence of high error rates for many tasks that humans find easy argues that incremental improvements may well not be enough to render current ASR suitable for widespread applications. The ASR of the future must be both knowledge- and stochastic-driven.

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TONES THAT REPRESENT CONTINUATIVE INTONATION IN ACADIAN FRENCH

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1. INTRODUCTION

Descriptions of French intonation state that continuity or non-finality in declarative sentences is generally realized with a rising tone. Phonological analyses (e.g. Di Cristo 1998; Jun & Fougeron 1995, 2000; Post 2000) based on these descriptions agree that a tone is associated with a stressed syllable and that an utterance is organized into different prosodic levels which are related hierarchically. There is however disagreement on the number of different levels as well as on the representation of the tones. It is also noteworthy that these phonetic and acoustic descriptions are based mainly on corpora elicited in laboratory settings using tasks such as reading and repetition.

Following the research cited above, I adopt an autosegmental-metrical framework (see Ladd 1996) and assume that French melodies consist of sequences of High (H) and Low (L) tones organized into at least two intonational units. The accentual phrase (AP) is a lower-level tonal unit that is the domain of primary and secondary stress; it has a final rise that delimits processable chunks of speech and that is often represented as /LH*/. The intonation phrase (IP) is a higher-level tonal unit that ends in a boundary tone that marks a major continuation rise (/H%/) or a major final fall (/L%/).

This paper examines tonal variation related to continuations at both AP and IP levels. It provides a focus on spontaneous speech, a style which complements most data types used in the laboratory based corpora. A second feature of this study is that it examines Acadian French, a variety spoken in Canada's Atlantic region which differs in significant ways from varieties such as Parisian French. In addition to phrase-final stress found in many varieties of French, Acadian French also has a penultimate stress. Certain vowels in open penultimate syllables lengthen and carry a pitch accent. For example, *anglais*, 'English', is pronounced [ANglais] or [anGLAIS]; *on lisait*, 'we used to read', is [on LIsait] or [on liSAIT]. This penultimate stress interacts with phrase-final intonation structure. Cross-varietal differences in the intonation of phrase-final syllabes have been attested in recent research on varieties of British English (Grabe et al 2000) and underscore the importance of this dimension to the study of intonational phonology. The theme underlying the present research is to arrive at an inventory of phonological contrasts and their pitch accent realizations in Acadian French.

2. METHOD

Subjects were 12 native speakers of a Nova Scotia variety of Acadian French; six were male, six female; three age groups were represented. Data were stories told spontaneously in the context of recorded sociolinguistic interviews. Approximately three minutes of speech per subject were digitized for analysis. Three listeners (all native speakers of this variety) carried out auditory analyses to identify prominent syllables and domain edges in this corpus; one of their main tasks was to isolate AP and IP boundaries. Two additional listeners determined grammatical structures in the data. Auditory and acoustic analyses (F0 tracks obtained with the XWAVES software package) were used to analyze and transcribe the intonational structures of the utterances.

3. RESULTS AND DISCUSSION

A total of 1124 units (out of about 1500 APs and IPs) were identified as showing continuity (as opposed to hesitation, finality and interrogation); 685 were APs and 439 IPs. Six patterns of surface realizations of tones in phrase-final positions were observed. Table 1 reports the relative frequencies of each pattern in the two phrasal contexts.

phonetic pattern	description	AP	IP
a. [L H]	low rise	15%	14%
b. [H]	high plateau	15%	15%
c. [L]	low plateau	43%	37%
d. [HL]	fall	10%	14%
e. [LHL]	rise-fall	10%	12%
f. [H L]	plateau-fall	7%	9%

Table 1. Relative frequencies of six types of surface tone realization by phrasal context.

The most striking result is the degree of difference between spontanous speech data and other styles (often used in studies).
a. [L H]: the low rising tone is fairly frequent, as predicted by most phonological analyses. In reading tasks (Jun & Fougeron 1995), this tone is very frequent (almost 90% of units observed).
b. [H]: the high tone is more frequent in a spontaneous setting than in reading (15% vs 1%).

c. [L]: the large number of low plateau tones is common in spontaneous speech. These tended to occur in contexts where listeners identified no phrase-final or phrase-penultimate stress.
d. [HL]: the falling tone is more common in spontaneous settings although it does occur in reading (Post 2000).
e. [LHL] and f. [H L]: these contours occur in both penultimate- and final-stressed syllables. Other varieties have the rise-fall pattern with phrase-final stress (Post 2000).

Figure 1 is an F0 track of the utterance: *(À l'école) on lisait anglais*, "(At school) we used to read English". The continuative rise-fall [LHL] tone on *lisait* has penultimate stress and is located at an AP boundary. *anglais* has final stress, is at an IP boundary and carries a [L] tone that indicates finality.

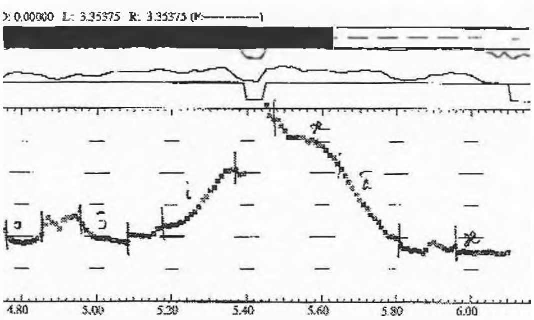

Figure 1. F0 track of *(À l'école) on lisait anglais*. Beginning and end of each vowel is indicated.

One of the relevant issues for a ToBI-type analysis of Acadian French is the representation of [L] tones at the end of APs where stress is penultimate. Current analysis suggests that AP-final tone is [H*] with no boundary tone. It is unclear how a fall can be generated from a peak in the accented syllable. One possible explanation may involve differences in timing needed to reach pitch targets.

As shown in Table 1, both AP and IP contexts have approximately the same frequencies of each type of surface tone realization. This suggests that, at least in continuative contexts, evidence from tones does not support two distinct levels of representation. Nevertheless, it appears that timing may be the cue for this distinction. In Acadian French (Cichocki 1996) and in Parisian French (Jun & Fougeron 2000), lengthening of final syllables is greater in IPs than in APs. The fact that native listeners were able to distinguish auditorily between two levels may reflect this cue.

4. CONCLUSION

Continuative intonation in Acadian French is realized by at least six types of surface tones, and these appear at both AP and IP levels. This high number of types is likely due to the context of spontaneous speech as opposed to read speech. A more elaborate analysis of discourse structure may also be revealing. Nevertheless, more systematic comparisons among styles are needed to develop a model of intonation. The presence of a penultimate pitch accent in Acadian French invites a more detailed phonological analysis of the intonation contours.

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PITCH-BASED ACOUSTIC FEATURE ANALYSIS FOR THE DISCRIMINATION OF SPEECH AND MONOPHONIC SINGING

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1. INTRODUCTION

A system capable of discriminating between human speech and human monophonic (*solo a capella*) singing would be a useful tool for several classes of applications including query-by-humming and content-based annotation and search of multimedia databases, as well as speech and music therapy and music education. Differences between speech and song have been investigated by List [1], and previous work by Zhang [2] contains a section on discriminating between speech and song (both with background music) in the context of a general audio classification scheme, using three features to make this classification: harmonic ripple, harmonic segment duration, and fundamental frequency above 300 Hz. Eric Scheirer and Malcom Slaney [3] have done work on discriminating speech and music using a set of 13 features, some of which are applicable to the speaking versus singing task described in this paper.

To investigate the perceptual differences between talking and singing, human subjects were exposed to a corpus of singing and talking sounds, and asked first to classify each sound on a scale between speaking and singing, and then to indicate the characteristics of the sounds that lead to their judgements. The subject responses indicated that pitch is a primary factor in making this judgement. Subjects indicated many pitch-based subfeatures including vibrato (similar to Zhang's harmonic ripple), excessively low or high pitch, adherence to a musical scale, and smoothness of pitch. Other features not directly related to pitch include rhythm, rhyme, context and expectation. As will be seen later in this paper, some of these features can also be investigated using pitch as a base feature.

2. METHOD

To develop a classification engine for the talking versus singing discrimination task, feature extractors are constructed based on perceptual characteristics indicated in the human subject trials. The full classification engine takes as input a sound file, extracts base features from this file, extracts subfeatures from these base features, combines these features using dimensionality reduction, and then presents a classification based on the nearest representative class in the experience of the classifier. This paper presents the extraction models for features based on the fundamental frequency (f_0), the physical counterpart of pitch.

2.1 f_0 Extraction

For this work, several current f_0 extractors were considered based on periodicity detection algorithms such as cepstrum and autocorrelation. The method used in this work is an autocorrelation-based algorithm [4] with several domain-specific improvements. f_0 measurements were extracted from each sound file at 15ms intervals. Example f_0 tracks are presented in Figures 1 and 2. These f_0 tracks are used as a base for the subfeatures described in Section 2.2.

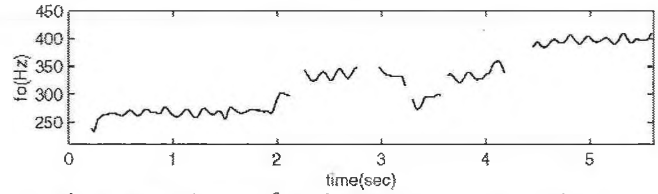

Fig. 1. Example sung f_0 track: "Row, row, row your boat."

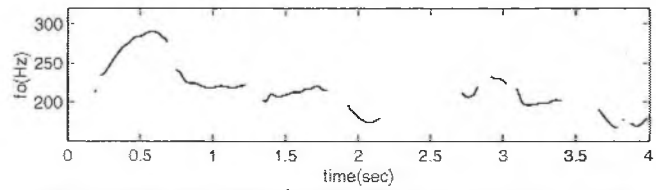

Fig. 2. Example spoken f_0 track: "Row, row, row your boat."

2.2 Subfeatures of f_0

The characteristics indicated in the human subject trials described in Section 1 inspire a set of f_0 subfeatures. Most of these subfeatures are direct derivations from the f_0 track, while some use additional information.

<u>Vibrato</u>

The most obvious difference between the f_0 tracks in Figures 1 and 2 is the presence of a ripple in the sung utterance. This phenomenon is caused by the singer modulating his or her vocal pitch [5] using a modulation frequency near 4 - 8 Hz. Vibrato is not present in all sung utterances, but it is present in very few spoken utterances. Vibrato indicators are extracted from the f_0 track using two standard periodicity detectors: autocorrelation and cepstrum.

<u>f_0 Statistics</u>

Many subjects commented on how sung utterances were higher or lower or more consistent in pitch. First order statistics are extracted from the f_0 track to measure these perceptual differences. Four statistical features are extracted: max, min, mean (μ) and standard deviation (σ). The slope of the f_0 track is also expected to be informative, indicating the rate of change of the pitch. The same statistical measures are used for the f_0 track slope.

Correlation between Syllables

Phrase repetition is a common characteristic of sung utterances. Often, a segment of pitch will be repeated with the same or different words, and this phenomenon is uncommon in spoken utterances. The correlation (R) between syllables is one way to identify the presence of this phenomenon. The f_0 track is separated into syllables at non-pitched segments and segments of high pitch slope. The set of f_0 segments ($f_0(k)$) is then correlated to obtain a measure of the similarity between syllables in the utterance.

3. RESULTS

Each feature described in Section 2 is implemented and tested on a corpus of singing and talking samples. The feature models are generated using the following procedure: Each feature extractor is presented to the set of singing files and the set of speaking files, and a statistical profile is determined based on the estimated likelihood of each feature value. These estimated probability density functions (*pdf*s) are then combined to generate the final feature model: where the *pdf* of the speech files is greater than the *pdf* of the song files, the feature values are evidence for speech, and vice versa. If the difference between the *pdf*s is below a threshold, the feature value is neutral.

Two feature models are presented here as examples. The "class" plot indicates the classification at each feature value. 1 = speaking, 5 = singing, 3 = no evidence.

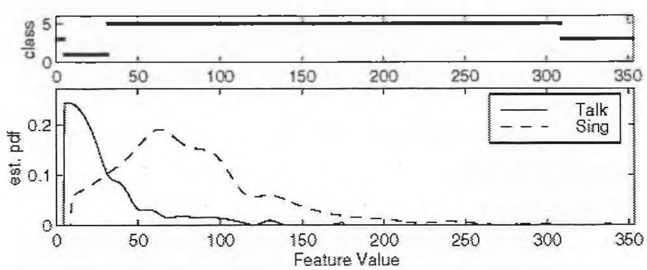

Fig. 3. Feature model of autocorrelation-based vibrato.

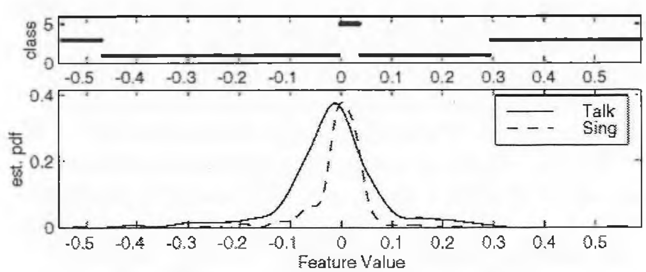

Fig. 4. Feature model of mean pitch slope.

Each feature model is evaluated based on the number of corpus files correctly classified. The correct rate for each feature is presented in Table 1.

Table 1. Percent correct for f_0-based features.

Feature	Rate	Feature	Rate
Vib AC	69.78%	max(f_0')	55.93%
Vib Cep	72.12%	min(f_0')	63.94%
max(f_0)	65.28%	μ (f_0')	65.28%
min(f_0)	54.09%	σ (f_0')	47.41%
μ (f_0)	68.78%	R ($f_0(k)$)	78.63%
σ (f_0)	55.93%		

4. DISCUSSION

Because some files do not exhibit the phenomena tested for in each feature, some individual features perform poorly. The next stage in this work is to develop multi-feature models. Dimensionality reduction is expected to be successful - because the features presented here are all based on a single superfeature, it is likely that some of these features are measuring the same underlying phenomenon.

A pitch-based feature that would seem useful in the speech/song task is pitch continuity. A song (it seems) is a series of discrete pitches, and it should be a simple task to recognize this in the pitch track. The difficulty with this is twofold: the human oratory system is not good at generating a stationary pitch, and the human auditory system is very good at recognizing non-stationary pitch tracks as consistent notes, using mechanisms which are not currently understood. Even identifying the intended target pitch of an utterance *known* to be song remains a difficult task.

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AUTHOR NOTES

The work was conducted while Mr. Gerhard was a student at Simon Fraser University.

Acquisition of Musical Vernacular in Children, Pre-Adolescents, and Young Adults

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1. INTRODUCTION

It has been proposed that children are biologically set to learn the languages to which they are exposed during early childhood (Chomsky, 1964; Jackendoff, 1994; Johnson & Newport, 1989; Lenneberg, 1967). Several psychologists and music theorists have pointed to similarities between language and music (Gardner, 1983; Lerdahl & Jackendoff, 1983; Sloboda, 1985). Gordon (1987) has suggested that musicality is fixed by the age of 8 or 9 years. Little research has focussed on the mental ability to acquire musical knowledge. To this end, a research program has been underway to test the notion that, like language grammar, a musical grammar is acquired early in life.

In earlier research, young and older adults were asked to rate familiarity of musical excerpts representative of music popular since 1900 (Cohen et al., 1995). Seniors were more familiar with music popular in the early decades, whereas, young adults were more familiar with music from recent decades. In a subsequent surprise recognition test of those same excerpts, for each age group, recognition memory was superior for more familiar selections. These findings were consistent with the view that the rules of a music 'vernacular' are acquired early in life and music in unfamiliar styles, that violates the original vernacular, is less readily encoded and retained in memory (Cohen, 2001). However, because children were not included in this study, the time frame for acquisition was unclear.

Snow (1993) proposes that the ability to acquire a second language is dependent on the social significance of the language. Therefore, acquisition of the rules of music may occur during adolescence when the social significance of music intensifies. The present study aimed to determine if the period for acquisition of the rules of musical vernacular coincides with (a) the early critical period for language acquisition, or (b) the time during adolescence when popular music is so socially significant.

2. METHOD

2.1 Materials

Six audio tapes, each containing 40 excerpts of popular music from the 10 decades of the 20th century, were updated from a previous study (Cohen et al., 1995). All selections were vocal arrangements ranging from 11 to 23 sec. Each decade was represented by eight excerpts. Each tape consisted of 4 blocks of 10 selections with a selection from each decade in each block. Four "presentation" tapes were used to measure preference and familiarity and two "test" tapes were used to measure recognition. One presentation tape and one test tape were used during each testing session. Each test tape contained 20 selections from the presentation tape and 20 foils that were not included in the presentation tape. There were 2 new and 2 old excerpts representing each decade.

2.2 Participants

Subjects included 17students in grades 1and 2 (6 males, mean age = 7.16 years, SD = .64), 47 students in grades 5 and 6 (28 males , mean age 11.64 years, SD .64) and 31 university students (7 males, mean age = 21.42 years, SD 2.16).

2.3 Procedure

All participants were first exposed to a presentation tape. On a 7-point scale, the children and some of the pre-adolescents rated the excerpts for preference; the remaining pre-adolescents and the young adults rated familiarity. All participants secondly received a surprise recognition task utilizing a test tape. The subjects rated on a 7-point scale how sure they were that the excerpt had been previously presented.

3. RESULTS

3.1 Preference

Mean preference of the excerpts increased with increasing decade, and for all 10 decades but the 90's mean ratings of the youngest group consistently exceeded those of the pre-adolescent group. Results of a mixed-model analysis of variance (ANOVA) with 1 between-groups factor (age group) and 1 within-groups factor (decade) indicated: (1) significantly higher mean preference for the youngest subjects (M = 4.57) than for the pre-adolescents (M = 3.08), F (1, 48) 40.05, $p < .001$, (2) a significant effect of decade, F (9, 432) = 49.52, $p < .001$, and (3) a significant interaction of decade and age group, F (9,432) = 5.30, $p <$ .001, attributable to a linear trend, F (1,48) 10.11, $p <$ .003. The flatter function for the young children suggests that they are more accepting of many styles of music than the pre-adolescent group. However, preference consistently increased with decade of popularity.

3.2 Familiarity

The familiarity ratings of the young adults and pre-adolescents increased with decade and were higher for the young adults. Familiarity ratings were entered into a mixed-model ANOVA. Young adults had significantly higher familiarity ratings ($M = 4.62$) than the pre-adolescents ($M=3.02$), $F(1,43) = 1306.21$, $p < .001$. Familiarity significantly increased with decade, $F(9,387) = 121.77$, $p < .001$, and the interaction between age group and decade was significant, $F(9, 387) = 6.90$, $p < .001$, attributable to a linear trend, $F(1,43) = 15.64$, $p < .001$, consistent with the greater differentiation by decade of the older subjects.

3.3 Recognition

Mean recognition scores increased with age. Mean recognition scores were entered into a mixed-model ANOVA. Results indicated a significant effect of (1) age group, $F(2,92) = 4051.90$, $p < .001$, (2) decade $F(9, 92) = 4.29$, $p < .001$, and (3) interaction of decade and age group, $F(18, 92) = 3.08$, $p < .001$. Bonferroni multiple comparisons indicated that mean rating scores for all groups were significantly different from each other.

For the foil scores, Bonferroni comparisons indicated that the mean scores of the young adults were significantly different from the two younger groups but the two younger groups were not significantly different from each other. To obtain a bias free measure of discrimination between songs actually heard and foils, the mean recognition and mean foil scores by decade were entered into a d' analysis. The mean d' scores collapsed across decades indicated that discriminability increased with age (Figure 1). A correlation between recognition d' and decade was significant only for young adult's recognition, $r = .81$ ($df = 8$), $p < .01$. For the youngest and the pre-adolescent students memory for excerpts of recent styles was not privileged as it was for young adults.

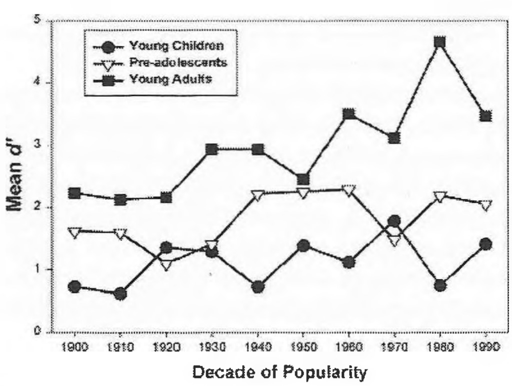

Figure 1. Mean d' for recognition as a function of age group and decade of popularity of the excerpt

4. DISCUSSION

The results that young children show no difference in immediate recognition of music of different styles and are more accepting of different styles of music than pre-adolescents supports the notion of an early critical period for the acquisition of music grammar. The superior memory of young adults for music in recent, familiar styles suggests that acquisition of the rules of the musical vernacular also coincides with the time during adolescence when popular music is socially significant. The results are consistent with the notion of an early period during which music of any style can be retained, as well as, a second period of refinement of sensitivity to a particular style. Such retention forms the basis for developing mental rules representing the musical vernacular to be used throughout life for encoding music, and as the basis of judgments of musical familiarity and preference. Therefore, exposure to many styles of music during the early sensitive period may provide a broad basis for the appreciation of multiple musical idioms throughout life.

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AUTHOR NOTE

Betty Bailey is a doctoral candidate in the Department of Music, University of Sheffield, Sheffield, UK, S10 5BR. The research is based on her honours thesis entitled *Similarities between the acquisition of musical structure and language grammar*, Department. of Psychology, University of Prince Edward Island, 1999.

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CONFIGURAL PROCESSING IN MELODY RECOGNITION

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1. INTRODUCTION

In the object perception literature, configuration refers to cases in which an object "contains several elements whose spatial relations to each other are perceived as structured and systematic, often giving rise to emergent properties" (Palmer, 1999, p. 707). Most studies of configural processing have been conducted with stimuli in the visual domain; we sought to investigate configural processing in the auditory domain. We chose to study melody recognition because melodies are considered the quintessential Gestalt object, according to Gestalt theory. We hypothesize that a configural processing occurs during melody perception and recognition. By "configural", we mean the processing of an abstract representation of a melody as relying on a specific arrangement of its constituent elements.

Preliminary empirical evidence for configural processing in melodies comes from the detrimental effects of manipulations that disrupt the holistic structure of the melodies, but leave the individual features intact, such as the scrambled melodies employed by Levitin, Menon, and Schmidt (2002). Listeners were to identify well-known melodies that had been divided into small pieces, then scrambled. Listeners reported that the scrambled melodies retained tonality, pitch, timbre, rhythmic, and instrumental information, but they could not recognize the melodies. The authors argued that it is the configural aspects – relations between elements in a melody – that are important for melody recognition.

In the present study, we weakened the influence of featural information of familiar melodies while preserving the configural properties (actual arrangement of the individual elements) of the melodies. Specifically, we created stimuli that minimized local cues to the identity of the melodies by disrupting the pitch sense. This was accomplished by replacing the tones of well-known melodies with bandpass filtered white noise bursts that degraded the pitch quality to such a degree that in the "high noise" conditions, absolute and relative pitch identifications were severely disrupted, while the overall configuration of the melodies remained intact.

We first tested pitch perception by presenting single degraded tones in isolation. Second, we presented pairs of degraded tones. Finally, we presented melodies that consisted of the degraded pitches and tested pitch perception of tones and their overall melody recognition. We refer to these three experimental conditions below as the "contexts" for the pitch identification tasks.

2. METHOD

2.1. Materials & Apparatus

All the stimuli employed were made of filtered noises, with bandpass filters centered symmetrically in log frequency around a centre frequency (F_0). The bandpass filters varied in their bandwidth (or Q: 0.5, 1, 2, 3, 6, 9, 12 semitone-wide).

Single-tone condition included stimuli that were single, filtered noises whose centre frequencies ranged from A4 to G#5 for a total of 12 frequencies. Six different bandwidths were randomly assigned to the 12 different centre frequencies of the filtered noises. Each tone was two seconds in duration. Double-tone or "interval" condition was composed of 12 pairs of filtered noises with the same Q per pair. Six different bandwidths were randomly assigned to the 12 pairs. The centre frequency of the second noise of the each pair ranged from A4 to G#5. The centre frequency of the first noise was randomly assigned from 12 musical intervals (ranging from minor 2nd to unison). The centre frequency of each second noise was randomly assigned to one of the 12 intervals, and the first tone was calculated using the randomly assigned interval size. Each tone was two seconds in duration. For the Melody condition, twenty well-known melodies were chosen based on melodies that are not identifiable by rhythmic cues alone. Each melody was assigned to one of five bandwidths. The centre frequency was assigned such that the last tone of each melody ranged from A4 to G#5. The average duration of the melodies was 14.53 seconds.

2.2. Procedure

Participants were 16 North American adults with at least ten years of music training. They were asked to identify a tone that could range from A4 to G#5 on a piano keyboard. They were allowed to hear the stimuli as many times as necessary, and they could also hum or whistle as an aid to matching the pitch of each tone. For the Single-tone condition, listeners were instructed to identify each single tone, but for the other two conditions, they were to identify the last tone of each pair of tones or each melody. In the Melody condition, participants also provided keywords or a title for each melody.

3. RESULTS

Figure 1 illustrates the average percentages of pitches that were correctly identified for each experimental condition across the different bandwidths (BW). A 3 x7 (Context x BW) repeated measures ANOVA indicated that there was a main effect of bandwidth ($F(6, 90) = 128.94$, $p < .001$), showing that the percentage of correct pitch identification differed significantly across different bandwidths. A main effect of context was not significant ($F(2, 30) = 1.70$, $p = .20$), indicating that there was no difference between contexts in pitch identification accuracy. Also, there was no interaction effect between contexts and bandwidths ($F(12, 180) = 2.02$, $p = .30$).

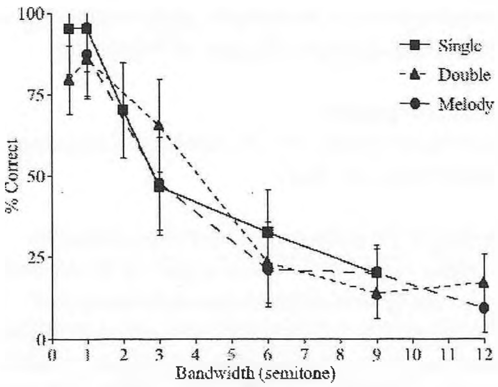

Fig. 1. Percentage of accuracy pitch identification.

We also compared melody recognition and pitch identification for the melody context (Figure 2). There was a significant main effect of task ($F(1, 15) = 137.67$, $p < .001$), indicating that listeners recognized melodies far better than they identified pitches. The main effect of bandwidth was also significant ($F(4, 60) = 34.95$, $p < .001$), indicating that listeners' performance differed across different bandwidths. There was a significant interaction between context and bandwidth ($F(4, 60) = 9.78$, $p < .001$). As shown in Figure 2, the significant interaction suggests that listeners' ability to identify pitches of the melodies declined with decreasing bandwidth while their melody recognition ability did not worsen.

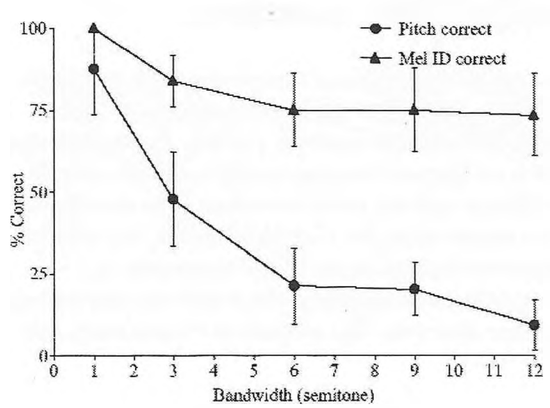

Fig. 2. Percentage of accurate pitch identification and melody recognition.

In order to determine whether listeners had access to contour information, which might have formed the basis for their melody identification judgments, we analyzed their pitch direction (a pitch going up or down) judgments in the Double-tone context. We calculated the percentage of correctly identified pitch directions using only the pitches that were incorrectly identified. The overall percentage of correctly identified pitch directions was 91.5% (N = 201), suggesting that listeners generally had a good sense of pitch direction when two tones were presented to them.

4. DISCUSSION

We sought to investigate whether melody recognition was possible on the basis of configural information. Feature-by-feature identification was disrupted since listeners' pitch perception weakened as the bandwidth of the filter increased. We also presented intervals and melodies consisting of filtered noises in order to provide richer contexts. Perception of degraded pitches was not enhanced by adding another tone or a sequence of tones. Most importantly, listeners were able to recognize melodies although they were not able to identify individual pitches that made up the melodies. Their response accuracy was still very high even when the bandwidth of the individual pitches became very wide, thereby disrupting their sense of pitch.

Listeners had a good sense of pitch direction, but this does not imply that melody recognition was based solely contour processing. In previous studies (Dowling & Fujitani, 1970; White, 1960), melody recognition based on contour information was rated much lower than that in our study (59 and 60% repectively). Also, there is evidence (Levitin et al., 2002) that a region in the brain is responsible for the arrangement of musical elements. Taken together, our results suggest that melody recognition does not need to rely on absolute pitch information; even the relative pitch information does not need to be grounded in real and clear pitch elements. Listeners could perceive relations between elements that have no clear identify and use these to access a stored memory trace.

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THE EFFECTS OF THE FAMILIARITY OF THE LANGUAGE OF BACKGROUND MUSIC LYRICS ON A WORD MEMORY TASK

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1. INTRODUCTION

Do lyrics in a language familiar to the listener distract the listener more than lyrics in an unfamiliar language? Huron (1999) defines lyric listening as a particular mode of listening in which the listener "may pay special attention to 'catching' the lyrics and attending to their meaning." This type of listening is only possible when the lyrics are in a language "understood by the listener." In turn, it is then expected that lyrics in a language unfamiliar to the listener can be more easily disregarded.

Salame and Baddeley (1989) compared the effects on memory in college students of unattended vocal and instrumental music, speech and amplitude-modulated noise. Speech and music with lyrics disrupted subjects' short-term recall abilities. Salame and Baddeley (1990) also showed that unattended speech can disrupt immediate phonological memory. In a previous study of Salame and Baddeley (1987) subjects classified pairs of visually presented consonants classified on the basis of either case or rhyme while continuous spoken Arabic played in the background. The Arabic, unfamiliar to the subjects, had no major effect on the subjects' performance.

Several other studies regarding music with lyrics and recall show negative effects of music on recall as opposed to conditions with no lyrics or no music at all (Balch, Bowman & Mohler 1992; Balch & Lewis 1996; Eich 1995; Sousou 1997). In addition, several studies on unattended speech also conclude that the speech decreases recall scores (Logie & Baddeley 1987; Martin, Wogalteret & Forlano 1989; Salame & Baddeley 1982, 1986, 1987, 1990). All these studies reinforce the fact that extraneous music and speech have an impact on memory and recall for other primary material presented at the same time. Music alone and speech alone affect memory, and music combined with lyrics (speech) also has a similar negative effect.

Based on the conclusions from the studies previously mentioned, it can be hypothesized that listening to lyrics in a language familiar to the listener will be more distracting than listening to lyrics in a language unfamiliar to the listener. While in a learning setting, the familiar language should cause disruption in the learning period and thus impair recall scores. In contrast, the unfamiliar language should have a smaller disruptive effect or no disruptive effect during the learning period and thus should not impair recall scores to the same degree as the familiar language.

2. METHOD

2.1 Participants

The subjects were 20 students of the University of Prince Edward Island between the ages of 18 and 24.

2.2 Apparatus and Materials

The lists containing two 20 word pairs consisted of two one-syllable nouns per pair.

The compact disc (CD) contained a rendition of Renato Carosone's classic swing style "Americano" in Italian and the Brian Setzer Orchestra's neo-swing style version of "Americano" in English. "Americano" was selected because it met the criteria of being recorded in the subjects' native language, English, and in another language with which the subjects were unfamiliar, in this case Italian. While a number of songs have been recorded in both English and French, none of these songs satisfied the familiarity criterion since most Canadians have some knowledge of French. Both songs had been edited to balance sound quality. The CD was played on a Panasonic RXES-25 Boombox stereo at a the same level for all listeners.

2.3 Procedure

Each subject received a condition with the English song and with the Italian song. Random assignment determined which 10 subjects heard the Italian version first and the English version second, and which 10 subjects heard the English version first and the Italian version second. Random assignment also determined which subjects received word list A first or word list B first.

Subjects completed the tasks in a quiet room. Each subject was told that the purpose of the experiment was to study memory capabilities while music is playing. The subject was handed the word list and the song was played. The subject was asked to stop looking at the word lists once the song had ended. One minute after the song had stopped, the subject was requested to recall as many of the word pairs as possible. The subject was given a word list with only one of the words from each pair. The subject attempted recall and write down the missing words. No music was played during the recall session.

The second condition preceded the same as the first, except with a different version of the song being played and a different list of word pairs. After the recall period, the subject was debriefed. It took approximately 20 min to complete the experiment per subject: 3 min for the first condition, 1 min to *relax*, 2 min for recall, 5 min between conditions, 3 min for the second condition, 1 min to *relax*, 2 min for recall, and a few additional minutes for instructions and debriefing.

The subjects were not informed of the actual intent of studying the difference between the recall scores while the different lyric languages are played. This was done in order to avoid an expectancy effect. Because the subjects would not be given all the details of the experiment in the beginning, the subjects were asked to sign a second consent form after the debriefing, allowing their data to be analyzed.

3. RESULTS

Of the 20 subjects, there were only 4 whose memory for English words was higher than that for Italian words, and 2 subjects whose scores were equal in both languages. The other 14 subjects had scores higher in Italian than English consistent with the hypothesis that lyrics in an unfamiliar language will be less detrimental to memory than lyrics in a familiar language.

The alpha level was set at .05 for the statistical report. The mean score for the English condition was 8.35 (standard deviation 4.67). The mean score for the Italian condition was 9.85 (standard deviation of 5.7). For the English condition, only 41.75% of the words were recalled while the Italian condition showed 49.25% of the words wererecalled; a 7.5% difference. These differences were in the expected direction but a t test showed no significance of the difference between the groups [$t(18) = 0.9$, $p > .05$]

4. DISCUSSION

While the results were not statistically significant, the trend in the means is consistent with the hyothesis that the more familiar language--English–for the lyrics would be more distracting. Only 6 of the 20 subjects had higher recall under the English lyrics, leaving 14 subjects whose scored higher under Italian lyrics. It must be noted that there is a confound in this study. The musical pieces, Carosone's "Americano," were recorded by two different artists in different styles and production settings. It is possible that if the recordings were by the same artist, and there were additional control, the effect would be stronger.
Based on Huron's definition of lyric listening, a difference in recall was expected between the presence of lyric language familiar to the listener and lyric language unfamiliar to the listener. Returning to this definition of lyric listening, that listeners "may pay special attention to 'catching' the lyrics and attending to their meaning" when the lyrics are in a language "understood by the listener" (Huron 1999), it could be hypothesized that greater interference of lyrics would arise with the familiarity of the language of the lyrics. The majority of the subjects ion the present study were more distracted by lyrics in a language that was familiar, and had a lower recall score than listening to lyrics in a language that was unfamiliar.

Lower scores in the English condition coincide with Baddeley's postulate of the articulatory loop. Memorizing the word pairs would entail subvocalizing and holding the words in articulatory loop. The English lyrics may have also entered the articulatory loop, interfering with the retention of the word pairs. When trying to recall the word pairs , the information would be inaccessible if forced out of the loop by the English lyrics. Because the subjects' ability to understand Italian is very limited, the words and their meanings would not be understood as word units by the subject, and thus would not enter the articulatory loop.

It is here suggested that the unfamiliar language does not simply disappear into thin air, but rather, is mentally converted from a language *into another instrument*, thus making the song an instrumental, and the lyrics *ignored* with the rest of the music. The whole song is simply *tuned out* to a level where is it not distracting, somewhat like the hum from a computer, a heater, or any other instrument of the soundtrack to daily life.

Although the original hypothesis, that English lyrics would be more distracting than Italian, was not statistically supported, the trends in the data warrant further exploration.

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For full references see: http://www.geocities.com/blacksuitband/References.html

NOTE. The study arose through an independent experiment for a class project and is being followed up in an honours these under the direction of Dr. Annabel Cohen

Targeting Timbral Recognition Abilities of Musical Participants: When Musicians Hear What Isn't There

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1. Introduction

The area of timbral studies in music perception is under-represented. Although part of the reason for this is owing to the difficulty of measuring timbre relative to ones perception, (Handel and Erickson, 2001), the focus of this study is geared toward simultaneous presentation of timbres and the difficulties listeners encounter with their perceptions, which often result in the misidentification of the instruments generating the timbres in question (Bregman, 1990). In spite of the fact that identifying a single timbre of any instrument (given reasonable range, onset, duration constraints and critical bandwidth) falls well within the purview of competent perceptual skill levels of musicians and non-musicians (Bregman, 1990), it is the discernment of instruments in combination that create perceptual problems for listeners, and that remains an important issue requiring investigation. Here, in this introductory study, we hypothesize that participants will have difficulty discerning the identity of four orchestral instruments presented from a passage taken from the first movement of Igor Stravinsky's *Ebony Concerto*.

2. Methods

A passage of 1300 ms in length consisting of 17 combined tones over 8 beats of measure 43 of Stravinsky's *Ebony Concerto* was played in three conditions to 22 participants. Condition 1 consisted of the measure in question, condition 2 provided 10 s of the musical material preceding measure 43 (approximately measure 37 ff.,) and condition three consisted of condition 2 material with 3000 ms of musical material after the targeted stimulus. Condition 3 was created so as to permit the participants to contextualize the stimulus window of condition 1, by hearing what follows the target stimulus in the piece itself. Trial Version 1: Participants (N=13) were presented with the targeted stimulus (bar 43) first, so that they knew what the target would be in advance. They were told that there would be three conditions, but that the order of the conditions would be randomized and that they could have up to ten voluntary iterations of each condition (such as condition order 2-1-3, to name one example). Trial Version 2: In this version of the experiment, nine (N=9) participants were played the three conditions in any order over eight compulsory iterations. The recording chosen was that of *Ensemble Intercontemporain*, directed by Pierre Boulez. Participants were also told the orchestration of the piece in advance of the trials, and that they were expected to choose instruments from the list provided.

3. Results

None of the participants were able to correctly name all four instruments (harp, tom-tom, clarinet, trombone). All participants named piano as an instrument even though no piano is found in the target stimulus passage. 45% of respondents (N=10) scored three correctly (the so-called "three-group"), and 45% scored two correctly (the so-called "two-group"). The remaining two respondents scored only 1 correctly. The three-group and the two-group misidentified trumpet equally, (four times each) but the two-group misidentified more instruments more often than the three-group overall, including french horn three times to one time in the three-group.

Overall, there were 2 opportunities for participants to score higher when moving from one block of iterations to the next, and when multiplied by 13 participants, this created an aggregate total of 26 potential differences for scores between blocks. Of these 26 potential differences, only 8 blocks registered improvement in accuracy (32.5%), indicating a low percentage overall for learning effects across conditions consistent in this case with a mild saturation effect. In the second trial version, 5 participants (N=9) answered with at least two correct identifications, 4 of whom accurately increased their correct scores. One participant answered initially with one correct answer and increased the score subsequently, and 3 answered with 0 but increased their score by the second iteration (N=2) or third iteration. It appeared that participants who were satisfied with their answers felt that what they were hearing was reflected accurately with what they were perceiving. No participants listened to fewer than three iterations per condition in Trial Version 1.

4. Discussion

Given the hypothesis as stated at the outset, it was not surprising that no one named all four instruments correctly. The two main groups, the two-group and three-group, showed one significant difference in the types of errors they made, namely that the three-group identified the clarinet more successfully. Both groups almost equally failed to identify the trombone. The most significant response however was the unanimous belief that piano was part of the target stimulus. Regardless of condition-order presentation, or number of iterations, 'piano' was responded by all participants. Such a widespread confusion was not anticipated. A variety of possible answers present themselves. The fundamental tone of the final 300 ms of condition 2 is G (98 Hz). The fundamental tone of both harp and tom-tom at 360 ms is also 98 Hz and thus is the same as the lowest tone of the final four eighth-note grouping of the piano in measure 42. The similarity of pitches from measures 42 to 43 is not surprising given Stravinsky's propensity to use pitch-class sets.

The close comparison of both fundamental frequency and super-imposed tones, with the exception of the C natural in measure 42, and the change of f natural to F sharp in bar 43 in harp, creates qualitatively very similar spectrographic conditions in the frequency domain, thereby making it possible that listeners could perceive a melodic and harmonic continuity from piano to harp. Furthermore, the spectral energy of both piano in measure 42, and harp/tom-tom in measure 43 appear similar enough in the upper harmonics to cause confusion.

One other possible explanation for the confusion may involve onset times. Onset forms the critical perceptual determinant of instrumental timbres and it is known that the deletion of the first few milliseconds of onset of a piano will result in a timbre closely approximating a harp (Sundberg, 1991). While the reverse certainly is not true, it raises the question of whether the combination of the harp/tom-tom timbre results in a peculiar perception, namely, the additive qualities of one percussion family instrument (tom-tom) plus one plucked-string family instrument (harp) equaling roughly the timbral quality of an instrument that is often regarded to be from both families (piano). Given this possibility, it could be that the harp, struck with the nail, (as indicated in Stravinsky's score) plus the tom-tom played with a felt stick could combine to create a characteristic onset quality and thus, a similar spectral quality that explains how participants could confuse the harp/tom-tom combined timbre with piano.

The task was also implicitly asking whether each participant could identify the combined timbre at the 975 ms point (beat 8) of the target stimulus, and whether they could deduce that the two timbres were clarinet and trombone. Participants reported that the combination of clarinet and trombone seemed to be played perfectly in tune, which is unusual when both instruments are orchestrated very high relative to their respective ranges. The clarinet plays the second harmonic of the trombone's fundamental, a challenge for performers in tuning the two notes doubled at the octave, and for participants in discerning the qualitative difference between the two instruments. The placement of both clarinet and trombone in a high acoustical range together likely creates vibrational complications for the performers resulting in fractional intonational differences (similar to when two french horns play together in their upper register). In this case, it may be surmised that intonational , and thereby, timbral differences are driven by the acoustical properties of the oscillators controlling pulsating airflow, namely the clarinet reed and the brass player's lips, resulting in perceptual problems for participants in identifying the two instruments.

5. Conclusions

Perception of combined tones is a difficult task that requires multiple iterations before discernment of individual instruments may be accomplished. Misidentified timbres tended to run along instrumental family lines (trumpet or french horn for trombone; piano for harp, or piano and harp for harp's differing timbres). However, the preliminary data suggest that errors may move along lines of instrumental familiarity, i.e. participants make errors by identifying the sounds of instruments they think they know best. These examples further illustrate the on-going experimental challenges involved in testing timbral perception.

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ARE MUSICIANS LISTENING TO SOUNDS IN A DIFFERENT WAY THAN NON-MUSICIANS?

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1.0 INTRODUCTION

People are subjected to many sounds entering the auditory system. Sounds may convey important information (targets) or may serve to distract a listener. Distracting sounds carrying irrelevant information can occur simultaneously or in a sequence with a target sound. In the first case, the distracter is mostly spectrally masking the target and in the second case, a temporal-like masking can occur - of which one phenomenon is the so-called Attentional Blink (AB). In order to explore both types of masking, two experiments were performed which tested musicians and compared their performance to that of non-musicians.

2.0 EXPERIMENT 1

Attention is deployed across time to regulate the ongoing flow of information arriving at the senses. Experimentally, the temporal nature of attention can be studied with rapid serial presentation techniques (RSVP, RAP). In these procedures, a series of stimuli (letters, pictures, tones, etc) are presented in rapid succession at the same location., and participants must identify 1 or 2 pre-specified targets. Numerous studies (e.g., 2, 3, 8) using these procedures have found that having to attend to the first target impairs the ability to identify a second target (probe) for ~ 500 ms - a phenomenon known as the *attentional blink* (AB).

The AB is influenced by a number of factors including stimulus complexity, task difficulty, and learning strategies (5, 6, 12). Furthermore, aging and attention deficits (e.g., ADHD) have been found to increase the magnitude of the AB (4, 9). Recently, in a preliminary study we (2) demonstrated that the magnitude of the auditory AB was *attenuated* among the congenitally blind, a finding that partly provided the impetus and rationale for the present study. Here, we examined whether the auditory AB would be attenuated in a group of musicians. Specifically, we hypothesized that because musicians have extensive experience with tones, the magnitude of the auditory AB should be attenuated, but not for the visual AB.

2.1 Method

<u>Participants:</u> Participants (19-39 yrs) were 17 university students who participated in the study for course credit. Musicians (n=8) had at least 7 years formal musical training.

<u>Procedure and Stimuli:</u> After training, participants were presented with 168 Rapid Auditory Presentation (RAP) streams (11 tones/sec) consisting of 25 equally loud tones ranging from 1000 to 2490 Hz. All tones were 85 ms in duration, separated by a silent 5 ms interstimulus interval. Targets to be named were 1500 (low) and 2500 (high) Hz tones increased in intensity by approximately 10 dB SPL above stream items. In the visual task, participants were presented with 168 Rapid Serial Visual Presentation streams (11 lines/sec) consisting of 25 sequentially-presented lines at orientations of degrees ranging from 30 to 150. Lines were 15 ms in duration, separated by a blank interval of 75 ms. Targets to be named were thicker lines of 45 (right), 90 (vertical), and 135 (left) degrees. In both tasks, 2 targets were present on 1/2 of the trials, balanced across SOAs of 90, 180, 270, 360, 450, 540, and 630 ms.

2.2 Results

Musicians had an attenuated auditory AB compared to non-musicians and the magnitude of the auditory AB was significantly reduced for this group (p's $< .05$). Unexpectedly, musicians also had an attenuated *visual AB*, except for the briefest interval of 90 ms (ABs not shown). The magnitude of the visual AB was also significantly reduced for musicians (p's $< .05$). Auditory and visual AB magnitudes are shown in Figure 1.

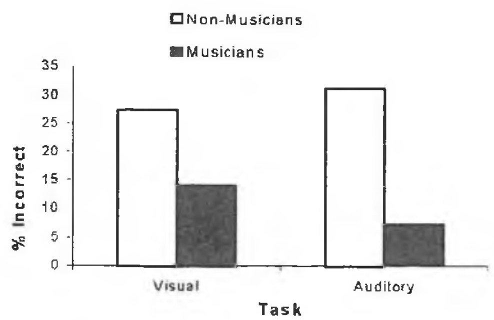

Figure 1. Visual and Auditory AB Magnitudes in Musicians and Non-musicians

2.3 Discussion

The hypothesis that musicians would demonstrate an attenuated auditory AB was supported. These preliminary data suggest that the auditory AB (i.e. auditory attention) is influenced by stimulus familiarity/learning. The demonstration of an attenuated visual AB was not anticipated, but suggests the presence of cross-modal enhancement, consistent with emerging findings from other investigations (11). Thus, there is more than just stimulus familiarity influencing the AB in musicians and, although the mechanisms responsible for these enhancements are not clear at present, enhanced attentional abilities may be one potential mechanism.

3.0 EXPERIMENT 2

This experiment was inspired by several lines of previous research. One study (1) showed that the detection of a pure-tone in noise was poorer in the presence of a second tone burst whose frequency and level varied randomly. Another study (10) demonstrated that the presence of a low intensity distracter increased detection threshold of target independently of listeners' age. Finally, it has also been shown that when the frequency content of a multitone masker (distracter) changes, detection of the target deteriorates (7). In the current study, we examined whether musicians would be better able to attend to a target sound compared to non-musicians in the presence of simultaneous distracters.

3.1 Method

Participants: Non-musicians were 24 students (21-32 yrs) who participated in the study for course credit. Musicians were 20 members 19-50 yrs) of the Calgary Philharmonic Orchestra, recruited from the Department of Music at the U of C, and had at least 7 years formal musical training. All participants had normal hearing (15 dB HL or better for all audiometric frequencies).

Stimuli: The target was a 1-kHz, 200-ms tone burst. The four tonal distracters were 350 ms in duration and 6 dB SL. In addition a continuous band-pass white noise (300Hz-1800Hz) was presented at a total intensity level of 60 dB SPL.

Procedure: Participants were tested individually in an anechoic chamber. A four-track interleaved adaptive procedure (3-up, 1-down) was used to determine the 79.4% threshold for the 1 kHz target in the presence of each of the four, randomly selected distracters.

3.2 Results

When detecting a 1 kHz target (important) sound in the presence of four uncertain tonal distracters (irrelevant sounds), musicians who played "soft" instruments (violin, viola, piccolo, flute, soprano, tenor) were able to detect the target sound significantly better than non-musicians and musicians who were playing "bold" instruments (tuba, double bass and band's players). These results are presented in Figure 2.

3.3 Discussion

Musicians playing low intensity instruments relative to other instruments, had lower thresholds for the detection of targets in the presence of simultaneous distracters. We speculate that these musicians may have developed a strategy for focusing on important sounds and are thus better able to reject irrelevant stimuli.

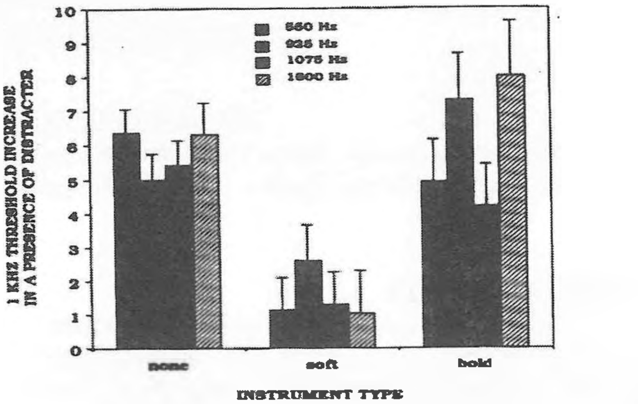

Figure 2. Means and standard deviations for music exposure groups (none, soft, and bold). See text for terms/explanation.

4.0 GENERAL DISCUSSION

Each of these experiments demonstrated that "soft-instrument" musicians were better able to focus on relevant targets and ignore distracters; consequently, it appears that they are less susceptible to the effects of masking. Both simultaneous and temporal masking may be part of a more general phenomenon known as informational masking, which are mediated by learning and attentional processes. If so, then musician's enhanced attentional abilities appear to generalize to other tasks, both within a sensory modality and across modalities.

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Acoustic Parameters as Cues to Judgments of Happy and Sad Emotions in Music

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1. INTRODUCTION

Though music can be appreciated for its technical merit and precision, it is no doubt the emotional content that is the main attraction for listeners (Panksepp, 1995). Clynes (1986) believed that emotions have biologically specified dynamic forms present in a gesture, music, dance, etc. If so, then the dynamic forms of emotion in music must be identifiable from the available acoustic parameters.

Acoustic parameters may be defined in terms of musical properties. Two parameters that have been studied extensively are tempo, or beats per minute (bpm) and mode, or key. Generally, a fast tempo and major mode are associated with 'happy' music, and a slow tempo and minor mode with 'sad' music (Dalla Bella et al., 2001; Hevner, 1935, 1937; Peretz, Gagnon, & Bouchard, 1998). However, other parameters, such as vibrato (frequency modulation) and articulation (related to tone onset rise time) (Gabrielsson & Juslin, 1996), have also been found to affect judgments of happy and sad emotion in music.

Our strategy was to select short musical segments, half composed with fast tempo and in the major mode, and half with slow tempo and in the minor mode. On the basis of past research, it was expected that the former would elicit judgments of happy and the latter judgments of sad. Differences in tempo and mode were then selectively removed so that tempo and/or mode could not be used as cues to discriminate emotional content. Reduced discrimination between the two types of segments would therefore implicate the role of tempo and mode. As well, the segments were later analysed for other potential correlates to judgments of emotion.

2. METHOD

2.1 Participants

The first of two groups included 20 (17 women, 3 men) third-year university undergraduates with a mean age of 23.0 yrs (range 21-30 yrs). The second group consisted of 42 (23 girls, 19 boys) high school students with a mean age of 16.7 yrs (range 15-18 yrs). Participants in both groups represented a wide range of music training--from zero to 12 years.

2.2 Stimuli

Twenty pieces of Western classical music in MIDI format were chosen so that 10 of the pieces had a relatively fast tempo (mean = 151 bpm) and were in the major mode and the other 10 had a relatively slow tempo (mean = 60 bpm) and were in the minor mode. Segments of three durations--0.5 s, 1 s, and 2 s--were taken from each piece, each beginning at a common point within the piece. Segments were presented under each of four cue conditions: (1) original tempo and mode; (2) equalized tempo (90 bpm) but original mode; (3) all in major mode but original tempo; (4) equalized tempo with all in major mode. Condition 2, 3, and 4 were constructed with sound editing software. Playback of stimuli was restricted in timbre and intensity. All segments were realized by a Yamaha S 100 XG piano timbre and key velocity was held to a narrow range.

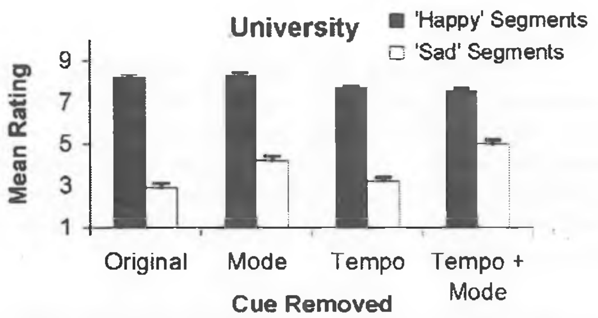
2.3 Design and Procedure

There were 240 trials--20 randomly ordered segments within each of 12 blocks. Each block represented one of the factorial combinations of three segment durations and four cue conditions. Participants rated each segment on a 10-point scale, where '1 = very sad', '5 = slightly sad', '6 = slightly happy' and '10 = very happy'. Participants were tested in groups.

3. RESULTS

For both the original and edited segments, segments from compositions with fast tempo in the major mode were rated as significantly more 'happy' than segments from compositions with slow tempo in the minor mode. The difference decreased as the original tempo and mode cues were removed, as exemplified in Figure 1 for the 2 s segments. Differences were highly significant ($p < .001$) even at the shortest (0.5 s) duration, and increased as segment duration increased. Thus, the data implicate tempo, mode, and duration as influencing judgments of happy and sad emotions in music.

However, the finding that ratings for the two types of segments differed significantly when both tempo and mode cues were removed suggests the presence of other cues influencing judgment. Mean ratings for each segment were entered into a regression analysis with predictors note density (number of notes per s) and average pitch height along with tempo and mode. Results are shown in Table 1. All four predictors contributed significantly to the regression. From the beta weights, note density and mode were determined to be the most important cues in predicting emotion ratings.



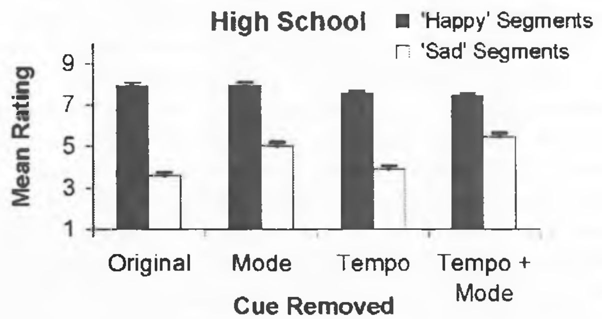


Figure 1. Mean emotion ratings for 2.0 s 'happy' and 'sad' segments in the four conditions for University (top) and High School students (bottom). The orientation of the rating scale is from sad '1' to happy '10'.

4. DISCUSSION

Both groups were easily able to use ratings of happiness/sadness to distinguish properties of musical segments with as little as 0.5 s exposure. When tempo and mode cues were removed, judgments became less distinct. Replicating earlier studies, the present study found that tempo and mode are clearly implicated in judgments of musical emotion.

The regression further clarified the involvement of tempo and mode but also showed that note density is as important as mode, and average pitch height of a segment may be more important than tempo in judgments of emotion. More notes per second and a higher average pitch height both lead to higher happiness ratings. Similarity of the two groups in ratings and in the regression equations suggests that, by the age of 15 years, people respond in an adult way.

In summary, tempo and mode were verified as important cues to judgments of happy and sad emotion in music. The addition of note density and average pitch height as perhaps equally important cues to judgments of emotion has implications for further research on emotions in music. Interpretation of data, taking into account only tempo and mode, may be problematic because identification of emotion can take place through other, uncontrolled cues available in music (Lantz, Kilgour, Nicholson, & Cuddy, in press).

Table 1. Results of multiple regression of acoustic cues on emotion ratings for both university and high school students.

	University			High School		
Cue	beta	t	p	beta	t	p
Note Density	.459	12.31	<.001**	.399	11.13	<.001**
Mode	.352	9.80	<.001**	.415	11.99	<.001**
Pitch Height	.215	6.28	<.001**	.228	6.94	<.001**
Tempo	.179	2.64	<.001**	.186	5.53	<.001**
Variance Accounted For		76%**			78%**	

**$p < .01$

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EFFECT OF MICROPHONE POSITION ON VOICE QUALITY

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1. INTRODUCTION

In most recording and live amplification situations, the human voice is picked up by a microphone placed in close proximity to the subject. Various sonic problems unique to the interaction between voice and microphones can prevail due to such close placement. The most common problems in practice are popping from plosive consonants, exaggerated sibilance and biased timbral coloration.

Popping results from a brief turbulence of air exhaled from the mouth when pronouncing plosive consonants such as "b", "p", "t", "d", "k", "g"; and fricatives such as "h", "f" and "v" [1]. The microphone diaphragm becomes momentarily displaced in an extreme fashion producing an audible "thump" or "pop" sound as its output. This problem only occurs with directional pressure-gradient type microphones. The most common remedy is to place a "pop screen filter" device made of a fabric or metal mesh between the voice and microphone. This is not entirely effective however as too often some blasts of air still pass through the pop screen to the microphone.

The microphone can also translate sibilant consonants (s, sh) into distorted or whistle-like sounds - an exaggerated "flashing" of "s" sounds relative to the other accompanying consonants and vowels. These effects are further emphasized when common signal processing is applied such as electronic high-frequency equalization boosts and dynamic range compression.

It is well known that any resultant audio timbre is dependent on microphone orientation. However, this is entirely due to the directional properties of the voice which, depending on the point of microphone pickup, may have significantly different spectral characteristics. For example, some positions could result in a fuller sound, or, even impose an alias formant quality.

It is certain that different microphones will produce any of these problems to different degrees. In effect, the results of this investigation are influenced principally by the radiation patterns of the different voice phonemes at close range regardless of microphone characteristics. This research will attempt to establish the effect of microphone point of pickup on these 3 sonic distortions and to identify positions that might have a lower occurrence of these problems.

2. METHOD

Four different test subjects participated in this experiment - 2 females and 2 males. The subjects each sat on a chair facing a cluster of four of the same model pressure-gradient condenser type microphones. Microphones with a super-cardioid directional pattern were chosen in order to focus the range of pickup from the selected positions.

The microphones were positioned as depicted in Figure 1a and 1b. Each microphone was equally placed 10 centimetres away from the centre of the subject's mouth. (It was expected that the bass-boost "proximity-effect" would occur at such a close distance, but that it would be a given constant since all 4 positions are equidistant from the sound source). The four positions relative to the subject's mouth were: above, directly in front, below and to the left side. The first 3 positions were at 0-degrees azimuth with the above and below microphones being 40-degrees above and below the plane of reference of the mid-position microphone. Only one side microphone was set up assuming symmetry. It was placed (as in Fig. 1b) on the same horizontal plane with the subject's mouth but to the subject's left side by 40-degrees. All microphones were aimed at the subject's mouth.

Fig. 1a

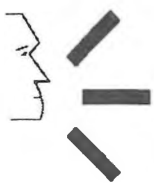
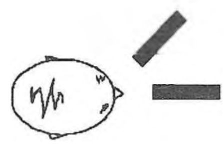


Fig. 1b



The room environment can best be described as semi-anechoic. In addition, the extreme close microphone placement to the sound source minimized the effect of room acoustics on the experiment.

The signals from all four microphones were recorded simultaneously into a desktop computer-based multitrack digital recording system.

To test for plosive consonants, the subject was scripted to speak a series of six separate sets of plosives and fricatives: b, p, t, k, h, and f. Each consonant was voiced with an attached set of vowel sounds: a (ah), e (eh), i (ee), o (oh), u (oo). For example: "Ba – Be - Bi – Bo - Bu".

The test stimuli for sibilant sounds involved the subject reciting the phrase, "many times loves lost" repeated twice.

The timbral test stimuli for the involved the subject voicing the vowel sounds, a, e, i, o, u as well as (nasal phonemes) "m" and "n". Each utterance was held at a constant pitch for about 3 seconds and repeated twice.

3. RESULTS

The plosive/vowel combinations that produced significant popping sounds from the microphone were quite apparent through listening as well as visually by the amplitude vs. time graphical waveform display. The results were tallied up in summary (Tables 1-4) with the maximum occurrences highlighted in grey:

Table 1: Consonant totals

B	P	K	T	H	F
10	29	9	25	31	21

Table 2: Vowel attachment totals

A	E	I	O	U
22	23	16	33	32

Table 3: Position totals

ABOVE	MID	BELOW	SIDE
28	62	23	12

Table 4: Detailed position totals

	B	P	K	T	H	F
above	0	5	1	6	3	13
mid	10	17	7	9	11	8
below	0	4	1	10	8	0
side	0	3	0	0	9	0

The sibilance tests were evaluated by simple listening via headphones and loudspeakers. The above microphone position gave the smoothest and most natural sibilant sound. The mid position gave the most exaggerated "s" sounds relative to the vowels and other consonants of the spoken phrase. Further validation is required through more blindfold listening evaluations from expert listeners, for example, sound engineers.

The most evident results in the timbral test were found with the nasal phonemes, "m" and "n". Here both aurally and visually (via the graphical waveform) the above position clearly had a fuller sound and higher amplitude output with the below position being the weakest.

Regarding the vowels (a,e,i,o,u), results show that the mid and side position timbres are virtually identical with the mid microphone position producing a slightly extended high-frequency content. The mid position showed the greatest proportion of high-frequency content of all 4 positions.

4. DISCUSSION

The results gathered here reveal some clear trends in the interaction between microphone position and specific voice phonemes. Many audio engineers often refer to any given microphone position as possessing a certain sonic characteristic perhaps preferable to other positions. This practice is a generalisation that may need to be further qualified dependent on the nature of what the sound source (voice, in this case) is actually producing. As a sound source, the voice challenges these oversimplifications because it is a peculiar "instrument" that changes physical form with every different phoneme it produces (not taking into account pitch). At such a close microphone position range, this change of shape results in marked differences in directional radiation. For example, it might be believed that a certain position can yield a "fuller" sound – yet this may only be true for certain phonemes as seen here with "m" and "n".

It appears (see Table 3) that the side position has the best chance of avoiding "popping". This can be an interesting option since the vowel and sibilant sounds are virtually the same here as they are for the mid position. However, the mid position (which is the most common choice for voice pickup) also provides the highest chance of "popping" from plosives. Both the above and below positions are also superior to the mid position, with the above position having the added possible benefits of producing a fuller sound (for nasal phonemes) and the smoothest sibilant sounds. It is also proven here that H and P along with the O and U vowels are the most troublesome phonemes for "popping" a microphone. These findings may also have implications towards the design of more effective pop screen filters.

The sibilant tests reveal what might be expected because the mid position would capture the highest proportion of high frequency energy causing the sibilant sounds to be most prominent. As well, the mid position may in effect produce a standing-wave reflections between the microphone itself and the face of the subject at short wavelengths (around 2 cm. and less).

REFERENCES

[1] Wuttke, J. (1992). Microphones and Wind. *Journal of the Audio Engineering Society*. Vol. 40, no. 10.

[2] Marshall, A.H., Meyer, J., (1985). The Directivity and Auditory Impression of Singers. *Acustica*, Vol. 58.

PROGRAM ANNOUNCEMENT
COMMUNICATION, CULTURE AND INFORMATION TECHNOLOGY (CCIT): UNIVERSITY OF TORONTO AND SHERIDAN COLLEGE

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1. INTRODUCTION

University of Toronto at Mississauga (UTM) and Sheridan College have recently initiated a joint program in Communication, Culture and Information Technology (CCIT). A major aim of the program is to train students in the theory, design, application, and management of communication systems. Acoustics and the auditory system form a key link between human communication and communication technologies, and are of central relevance to the program.

The CCIT program combines the internationally recognized academic and research disciplines in the humanities and social sciences at UTM with the professional and applied strengths of Sheridan College — world renowned for programs in computing, visualization, digital media, design, performing arts, and journalism. The program is open to students admitted to the University of Toronto and students will graduate with a Bachelors degree (BA or B.Sc.).

One aim of the CCIT program is to provide students in arts or science programs with a second major that will give them valuable skills in applied areas, thereby increasing their employability (e.g., Psychology and CCIT; Anthropology and CCIT). There are also plans to offer *specialist programs* so that students can concentrate their studies in areas such as digital enterprise management; human communication; professional writing; theatre and media; and visual communication.

Graduates of CCIT will be sensitive to ethical, cultural, and social issues addressed by the humanities and liberal arts, and will also be comfortable working in an applied setting with new and emerging technologies. University of Toronto is extensively recruiting new faculty to manage the program and conduct research. Research on biological communication systems, for example, will take place in the Centre for Research on Biological Communication Systems (CRBCS), an initiative supported by a grant from the Canada Foundation for Innovation (CFI: See Schneider, this issue).

New buildings dedicated to the CCIT program are under construction at both the UTM and Sheridan College campuses. These buildings were made possible by grants from Ontario's SuperBuild Fund and from the City of Mississauga. At UTM, one floor of the CCIT building will be dedicated to the CRBCS. This 2000 m^2 facility will include equipment designed to measure, store, and analyze sound, as well as equipment designed to simulate the acoustic characteristics of natural environments. The CRBCS centre will also house a multimodal virtual-reality test station and several sound-attenuating chambers for experimental testing.

The CRBCS will support a range of studies in acoustics, auditory perception, speech, and multimodal communication. This component of CCIT is described by Schneider (this volume) and will not be reiterated here. With its promise of matching academic excellence with the world of quickly changing practical skills, CCIT will be a dynamic program, and its link to expertise in acoustics and audition will be essential to CCIT's success.

A number of *specialist programs*, currently under consideration for the CCIT curriculum, will provide students with various programs options that lead to different career destinations. A few of these programs are described below.

2.0 DIGITAL ENTERPRISE MANAGEMENT (DEM).

Communication technologies rely on the digitization and transmission of acoustic and visual information. Digital enterprises require the involvement of experts in communication theory, digital technologies, and multi-modal channels of communication.. Such enterprises are managed in ways that are fundamentally different from the management of other kinds of businesses. Students in this specialist program will be trained in the technologies associated with digital enterprises, the perceptual constraints on communication systems, and the challenges associated with managing such enterprises.

Students in the DEM stream will acquire skills needed to become managers and leaders of digital companies, technological entrepreneurs, E-content developers, and technology policy makers.

3.0 HUMAN COMMUNICATION ACROSS THE LIFESPAN (HCL).

Communication is an essential part of human life, and the auditory system is well adapted to the perception and cognition of auditory signals involved in human communication. Pre-linguistic infants communicate emotionally with their parents through lullabies and tone of voice. With maturation, children learn to perceptually segment spoken utterances into words and syllables, categorize words, and refine their skills at language production. For adults, communication skills are central to career success and social interaction. Management and leadership positions require an enhanced ability to communicate complex ideas, along with an ability to convey and interpret emotional meaning.

Disabilities in communication may occur following brain injury (e.g., due to stroke and trauma) and as a consequence of aging. Emerging technologies, however, provide an opportunity to treat such disabilities. Students in the HCL program will learn how human communication is influenced by cognitive processes, social and cultural factors, and technology. They will develop skills leading to careers in the use of technology in human communication, or in the health services sector (gerontology, speech pathology).

4.0 PERFORMANCE, THEATRE, AND MEDIA STUDIES (PTM).

A performance is a prepared presentation of a character, whether in the context of a political speech, a business presentation, a musical recital or a theatrical event. Performances may be mediated through stage, film, the internet, and television, and these media are continuously transformed by new technologies.

Students in PTM will study aspects of performance including acting, editing, music scoring, and sound design, and will gain an understanding of how performances are mediated by contemporary technologies. Graduates of PTM will be expert communicators, producers, and interpreters of performance in a number of media, and can apply these skills in the arts industry, education, marketing, public relations, or advertising.

5.0 VISUAL COMMUNICATION.

Although the auditory system is particularly well adapted for communication (i.e., speech and music), our culture is increasingly dominated by visual communication. Visual forms of communication include art, advertising, photography, film, web sites, medical images, and simulations of events used for the purposes of research. Are such visual forms of communication analogous to spoken languages or do they each have their own special "language"? What are the most recent techniques for creating visual forms of communication?

These two questions are addressed in two proposed specialist areas — one in Visual Culture and Communication (VCC) and the other in Computer Visualization (CV). Students in the VSS and CV programs will learn to design and interpret visual forms of communication, but whereas the VSS program will emphasize the cultural and historical significance of visual communication, the CV program will train students to be skilled practitioners who use applied visual semiotics to communicate information in visual forms. Graduates of these programs will have skills leading to careers in advertising, graphic design, and visual information technology.

6.0 FURTHER INFORMATION

Details on the CCIT program are provided on the CCIT web site (http://ccit.erin.utoronto.ca/) or may be obtained by calling 905-569-4732.

INDEX OF AUTHORS

CONFERENCES

The following list of conferences was mainly provided by the Acoustical Society of America. If you have any news to share with us, send them by mail or fax to the News Editor (see address on the inside cover), or via electronic mail to francine.desharnais@drdc-rddc.gc.ca

2002

10-12 September: 32nd International Acoustical Conference – EAA Symposium, Banská Stiavnica, Slovakia. Contact: M. Culik, Physics and Applied Mechanics Department, TU Zvolen, Masarykova 24, 96001 Zvolen, Slovakia; Fax: +421 45 532 1811; Web: http://alpha.tuzvo.sk/skas/acoustics

11-13 September: 10th International Meeting on Low Frequency Noise and Vibration, York, UK. Contact: W. Tempest, Multi-Science Co. Ltd., 5 Wates Way, Brentwood, Essex CM15 9TB, UK; Fax: +44 1277 223 453; Web: www.lowfrequency2002.org.uk

11 September: Dutch and Belgian Acoustical Societies Joint Meeting, Utrecht, The Netherlands. E-mail: info@nag-acoustics.nl

16-18 September: International Conference on Noise and Vibration Engineering, Leuven, Belgium. Fax: +32 1632 2987; Web: www.isma-isaac.be

16-20 September: 7th International Conference on Spoken Language Processing (ICSLP 2002), Interspeech 2002, Denver, CO. Contact: ICSLP 2002, Centennial Conferences, 801 Main St., Suite 010, Louisville, CO 80027; Tel.: 303-499-2299; Fax: 303-499-2599; E-mail: icslp@centennialconferences.com; Web: www.icslp2002.org

16-21 September: Forum Acusticum 2002 (Joint EAA-SEA-ASJ Meeting), Sevilla. Fax: +34 91 411 7651; Web: www.cica.es/aliens/forum2002

26-28 September: Autumn Meeting of the Acoustical Society of Japan, Akita, Japan. Contact: Acoustical Society of Japan, Nakaura 5th-Bldg., 2-18-20 Sotokanda, Chiyoda-ku, Tokyo 101-0021, Japan; Fax: +81 3 5256 1022; Web: wwwsoc.nacsis.ac.jp/asj/

9-11 October: Acoustics Week in Canada, Charlottetown, PEI, Canada. Contact: A. Cohen, Department of Psychology, University of Prince Edward Island, 550 University Avenue, Charlottetown, PE, C1A 4P3, Canada; Fax: 902-628-4359; Web: http://caa-aca.ca/PEI-2002.html

26-28 October: 6th National Congress of the Turkish Acoustical Society, Kars, Turkey. Contact: Türk Akustik Dernegi, YTÜ Mimarlik Fakültesi, 80750 Besiktas-Istanbul, Turkey; Fax: +90 212 261 0549; Web: http://www.takder.org/kongre-2002/kongre2002.html

29-31 October: Oceans 2002, Biloxi, MS. E-mail: oceans@jspargo.com; Web: www.oceans2002.com

CONFÉRENCES

La liste de conférences ci-jointe a été offerte en majeure partie par l'Acoustical Society of America. Si vous avez des nouvelles à nous communiquer, envoyez-les par courrier ou fax (coordonnées incluses à l'envers de la page couverture), ou par courriel à francine.desharnais@drdc-rddc.gc.ca

2002

10-12 septembre: 32e conférence internationale d'acoustique – Symposium EAA, Banská Stiavnica, Slovaquie. Info: M. Culik, Physics and Applied Mechanics Department, TU Zvolen, Masarykova 24, 96001 Zvolen, Slovakia; Fax: +421 45 532 1811; Web: http://alpha.tuzvo.sk/skas/acoustics

11-13 septembre: 10e rencontre internationale sur le bruit et les vibrations à basse fréquence, York, Royaume-Uni. Info: W. Tempest, Multi-Science Co. Ltd., 5 Wates Way, Brentwood, Essex CM15 9TB, UK; Fax: +44 1277 223 453; Web: www.lowfrequency2002.org.uk

11 septembre: Rencontre combinée des Sociétés hollandaise et belge d'acoustique, Utrecht, Pays-Bas. Courriel: info@nag-acoustics.nl

16-18 septembre: Conférence internationale sur le génie du bruit et des vibrations, Leuven, Belgique. Fax: +32 1632 2987; Web: www.isma-isaac.be

16-20 septembre: 7e Conférence internationale sur le traitement de la langue parlée (ICSLP 2002), Interspeech 2002, Denver, CO. Info: ICSLP 2002, Centennial Conferences, 801 Main St., Suite 010, Louisville, CO 80027; Tél.: 303-499-2299; Fax: 303-499-2599; Courriel: icslp@centennialconferences.com; Web: www.icslp2002.org

16-21 septembre: Forum Acusticum 2002 (Rencontre conjointe EAA-SEA-ASJ), Séville. Fax: +34 91 411 7651; Web: www.cica.es/aliens/forum2002

26-28 septembre: Rencontre d'automne de la Société japonaise d'acoustique, Akita, Japon. Info: Acoustical Society of Japan, Nakaura 5th-Bldg., 2-18-20 Sotokanda, Chiyoda-ku, Tokyo 101-0021, Japan; Fax: +81 3 5256 1022; Web: wwwsoc.nacsis.ac.jp/asj/

9-11 octobre: Semaine canadienne d'acoustique 2002, Charlottetown, IPE, Canada. Info: A. Cohen, Department of Psychology, University of Prince Edward Island, 550 University Avenue, Charlottetown, PE, C1A 4P3, Canada; Fax: 902-628-4359; Web: http://caa-aca.ca/PEI-2002.html

26-28 octobre: 6e congrès national de la Société turque d'acoustique, Kars, Turquie. Info: Türk Akustik Dernegi, YTÜ Mimarlik Fakültesi, 80750 Besiktas-Istanbul, Turkey; Fax: +90 212 261 0549; Web: http://www.takder.org/kongre-2002/kongre2002.html

29-31 octobre: Oceans 2002, Biloxi, MS. Courriel: oceans@jspargo.com; Web: www.oceans2002.com

13-15 November: Australian Acoustical Society Conference 2002, Adelaide, Australia. Contact: AAS 2002 Conference Secretariat, Department of Mechanical Engineering, Adelaide University, SA 5005, Australia; Fax: +61 8 303 4367; Web: www.mecheng.adelaide.edu.au/aasconf

15-17 November: Reproduced Sound 18: Perception, Reception, Deception, Stratford-upon-Avon, UK. Contact: Institute of Acoustics, 77A St. Peter's Street, St. Albans, Herts AL1 3BN, UK; Fax: +44 1727 850553; Web: www.ioa.org.uk

21-22 November: New Zealand Acoustical Society 16th Biennial Conference, Auckland, New Zealand. Contact: New Zealand Acoustical Society, PO Box 1181, Auckland, New Zealand; E-mail: graham@marshallday.co.nz

2-6 December: Joint Meeting: 9th Mexican Congress on Acoustics, 144th Meeting of the Acoustical Society of America, and 3rd Iberoamerican Congress on Acoustics, Cancun, Mexico. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; Web: asa.aip.org/cancun.html

9-13 December: International Symposium on Musical Acoustics (ISMA Mexico City), Mexico City. Fax: +52 55 5601 3210; Web: www.unam.mx/enmusica/ismamexico.html

2003

17-20 March: German Acoustical Society Meeting (DAGA2003), Aachen, Germany. Fax: +49 441 798 3698; E-mail: dega@akuphysik.uni-oldenburg.de

18-20 March: Spring Meeting of the Acoustical Society of Japan, Tokyo, Japan. Contact: Acoustical Society of Japan, Nakaura 5th-Bldg., 2-18-20 Sotokanda, Chiyoda-ku, Tokyo 101-0021, Japan; Fax: +81 3 5256 1022; Web: wwwsoc.nii.ac.jp/asj/index-e.html

7-9 April: WESPAC8, Melbourne, Australia. Web: www.wespac8.com

28 April – 2 May: 145th Meeting of the Acoustical Society of America, Nashville, TN. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; Web: asa.aip.org

8-13 June: XVIII International Evoked Response Audiometry Study Group Symposium, Puerto de la Cruz, Tenerife, Canary Islands, Spain. Fax: +34 922 27 03 64; Web: www.ierasg-2003.org

16-18 June: ACOUSTICS – Modeling & Experimental Measurements, Cadiz, Spain. Contact: Acoustics03, Wessex Institute of Technology, Ashurst Lodge, Ashurst, Southampton SO40 7AA, UK; Fax: +44 238 029 2853; Web: www.wessex.ac.uk/conference/2003/acoustics/index.html

13-15 novembre: Conférence 2002 de la Société australienne d'acoustique, Adélaïde, Australie. Info: AAS 2002 Conference Secretariat, Department of Mechanical Engineering, Adelaide University, SA 5005, Australia; Fax: +61 8 303 4367; www.mecheng.adelaide.edu.au/aasconf

15-17 novembre: Sons reproduits 18: Perception, Réception, Déception, Stratford-upon-Avon, Royaume-Uni. Info: Institute of Acoustics, 77A St. Peter's Street, St. Albans, Herts AL1 3BN, UK; Fax: +44 1727 850553; Web: www.ioa.org.uk

21-22 novembre: 16e conférence bisannuelle de la Société d'acoustique de la Nouvelle-Zélande, Auckland, Nouvelle-Zélande. Info: New Zealand Acoustical Society, PO Box 1181, Auckland, New Zealand; Courriel: graham@marshallday.co.nz

2-6 décembre: Rencontres combinées: 9e Congrès mexicain d'acoustique, 144e rencontre de l'Acoustical Society of America, et 3e Congrès ibéro-américain d'acoustique, Cancun, Mexique. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél: 516-576-2360; Fax: 516-576-2377; Courriel: asa@aip.org; Web: asa.aip.org/cancun.html

9-13 décembre: Symposium international sur l'acoustique musicale (ISMA Mexico City), Mexico, Mexique. Fax: +52 55 5601 3210; Web: www.unam.mx/enmusica/ismamexico.html

2003

17-20 mars: Rencontre de la Société allemande d'acoustique (DAGA2003), Aachen, Allemagne. Fax: +49 441 798 3698; Courriel: dega@akuphysik.uni-oldenburg.de

18-20 mars: Rencontre de printemps de la Société japonaise d'acoustique, Tokyo, Japon. Info: Acoustical Society of Japan, Nakaura 5th-Bldg., 2-18-20 Sotokanda, Chiyoda-ku, Tokyo 101-0021, Japan; Fax: +81 3 5256 1022; Web: wwwsoc.nii.ac.jp/asj/index-e.html

7-9 avril: WESPAC8, Melbourne, Australie. Web: www.wespac8.com

28 avril – 2 mai: 145e rencontre de l'Acoustical Society of America, Nashville, TN. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél.: 516-576-2360; Fax: 516-576-2377; Courriel: asa@aip.org; Web: asa.aip.org

8-13 juin: XVII Symposium international du Groupe expérimental sur l'audiométrie des potentiels évoqués, Puerto de la Cruz, Tenerife, Iles Canaries, Espagne. Fax: +34 922 27 03 64; Web: www.ierasg-2003.org

16-18 juin: ACOUSTICS – Modélisation et mesures expérimentales, Cadiz, Espagne. Info: Acoustics03, Wessex Institute of Technology, Ashurst Lodge, Ashurst, Southampton SO40 7AA, UK; Fax: +44 238 029 2853; Web: www.wessex.ac.uk/conference/2003/acoustics/index.html

29 June – 3 July: 8th Conference on Noise as a Public Health Problem, Amsterdam-Rotterdam, The Netherlands. Contact: Congress Secretariat, PO Box 1558, 6501 BN Nijmegen, The Netherlands; Fax: +31 24 360 1159; E-mail: office.nw@prompt.nl

7-10 July: 10th International Congress on Sound and Vibration, Stockholm, Sweden. Contact: Congress Secretariat, Congrex Sweden AB; Tel: +46 8 459 66 00; Fax: +46 8 8 661 91 25; E-mail: icsv10@congrex.se; Web: www.congex.com/icsv10

14-16 July: 8th International Conference on Recent Advances in Structural Dynamics, Southampton, UK. Web: www.isvr.soton.ac.uk/sd2003

6-9 August: Stockholm Music Acoustics Conference 2003 (SMAC03), Stockholm, Sweden. Contact: www.speech.kth.se/music/smac03

25-27 August: Inter-Noise 2003, Jeju Island, Korea. Contact: Dept. of Mechanical Engineering, KAIST, 373-1, Kusong-dong, Yusong-gu, Taejon 305-701, Korea; Fax: +82 42 869 8220; Web: www.icje-ju.co.kr

1-4 September: Eurospeech 2003, Geneva, Switzerland. Contact: SYMPORG SA, Avenue Krieg 7, 1208 Geneva, Switzerland; Fax: +41 22 839 8485; Web: www.symporg.ch/eurospeech2003

7-10 September: World Congress on Ultrasonics, Paris, France. Web: www.sfa.asso.fr/wcu2003

15-17 October: 34th Spanish Congress on Acoustics, Bilbao, Spain. Contact: Sociedad Española de Acústica, Serrano 144, 28006 Madrid, Spain; Fax: +34 91 411 7651; Web: www.ia.csic.es/sea/index.html

10-14 November: 146th Meeting of the Acoustical Society of America, Austin, TX. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; Web: asa.aip.org

2004

22-26 March: Joint Congress of the French and German Acoustical Societies (SFA-DEGA), Strasbourg, France. Contact: Société Française d'Acoustique, 23 avenue Brunetière, 75017 Paris, France; Fax: +49 441 798 3698; E-mail: sfa4@wanadoo.fr

5-9 April: 18th International Congress on Acoustics (ICA2004), Kyoto, Japan. Web: ica2004.or.jp

24-28 May: 75th Anniversary Meeting (147th Meeting) of the Acoustical Society of America, New York, NY. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; Web: asa.aip.org

29 juin – 3 juillet: 8e conférence sur le bruit, un problème de santé publique, Amsterdam-Rotterdam, Pays-Bas. Info: Congress Secretariat, PO Box 1558, 6501 BN Nijmegen, The Netherlands; Fax: +31 24 360 1159; Courriel: office.nw@prompt.nl

7-10 juillet: 10e Congrès international sur le bruit et les vibrations, Stockholm, Suède. Info: Congress Secretariat, Congrex Sweden AB; Tél.: +46 8 459 66 00; Fax: +46 8 8 661 91 25; Courriel: icsv10@congrex.se; Web: www.congex.com/icsv10

14-16 juillet: 8e Conférence internationale sur les développements récents en dynamique structurelle, Southampton, Royaume-Uni; www.isvr.soton.ac.uk/sd2003

6-9 août: Conférence 2003 d'acoustique musicale de Stockholm (SMAC03), Stockholm, Suède. Info: www.speech.kth.se/music/smac03

25-27 août: Inter-Noise 2003, Île Jeju, Corée. Info: Dept. of Mechanical Engineering, KAIST, 373-1, Kusong-dong, Yusong-gu, Taejon 305-701, Korea; Fax: +82 42 869 8220; Web: www.icjeju.co.kr

1-4 septembre: Eurospeech 2003, Genève, Suisse. Info: SYMPORG SA, Avenue Krieg 7, 1208 Geneva, Switzerland; Fax: +41 22 839 8485; Web: www.symporg.ch/eurospeech2003

7-10 septembre: Congrès mondial sur les ultra-sons, Paris, France. Web: www.sfa.asso.fr/wcu2003

15-17 octobre: 34e Congrès espagnole d'acoustique, Bilbao, Espagne. Info: Sociedad Española de Acústica, Serrano 144, 28006 Madrid, Spain; Fax: +34 91 411 7651; Web: www.ia.csic.es/sea/index.html

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2004

22-26 mars: Congrès conbiné des Sociétés française et allemande d'acoustique (SFA-DEGA), Strasbourg, France. Info: Société Française d'Acoustique, 23 avenue Brunetière, 75017 Paris, France; Fax: +49 441 798 3698; Courriel: sfa4@wanadoo.fr

5-9 avril: 18e Congrès international sur l'acoustique (ICA2004), Kyoto, Japon. Web: ica2004.or.jp

24-28 mai: 75e rencontre anniversaire (147e rencontre) de l'Acoustical Society of America, New York, NY. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél.: 516-576-2360; Fax: 516-576-2377; Courriel: asa@aip.org; Web: asa.aip.org

13-17 September: 4th Iberoamerican Congress on Acoustics, 4th Iberian Congress on Acoustics, 35th Spanish Congress on Acoustics, Guimarães, Portugal. Contact: Sociedade Portuguesa de Acústica, Laboratório Nacional de Engenharia Civil, Avenida do Brasil 101, 1700-066 Lisboa, Portugal; Fax: +351 21 844 3028; E-mail: dsilva@lnec.pt

29 November – 3 December: 148th Meeting of the Acoustical Society of America, San Diego, CA. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; Web: asa.aip.org

13-17 septembre: 4e Congrès ibéro-américain d'acoustique, 4e Congrès ibérien d'acoustique, 35e Congrès espagnol d'acoustique, Guimarães, Portugal. Info: Sociedade Portuguesa de Acústica, Laboratório Nacional de Engenharia Civil, Avenida do Brasil 101, 1700-066 Lisboa, Portugal; Fax: +351 21 844 3028; Courriel: dsilva@lnec.pt

29 novembre – 3 décembre: 148e rencontre de l'Acoustical Society of America, San Diego, CA. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél.: 516-576-2360; Fax: 516-576-2377; Courriel: asa@aip.org; Web: asa.aip.org

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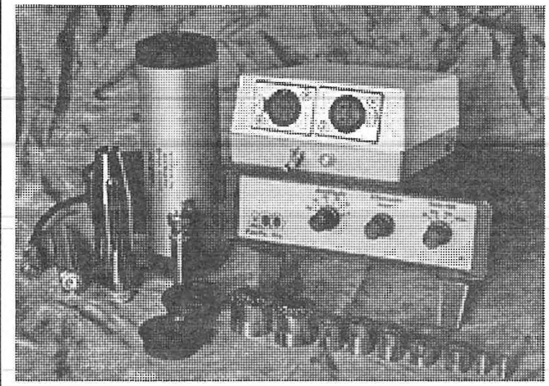

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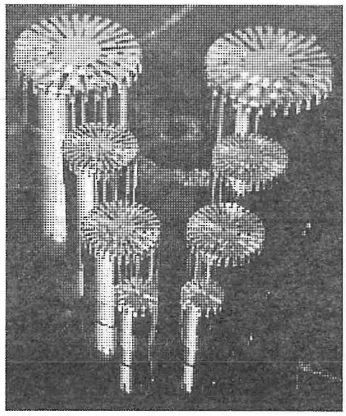

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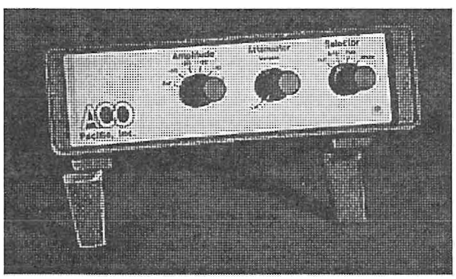

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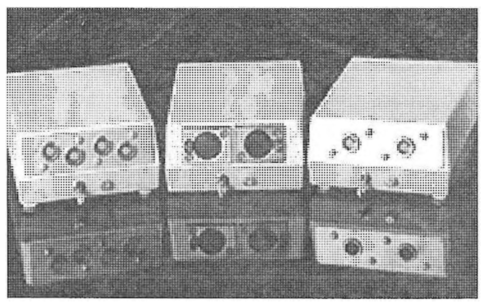

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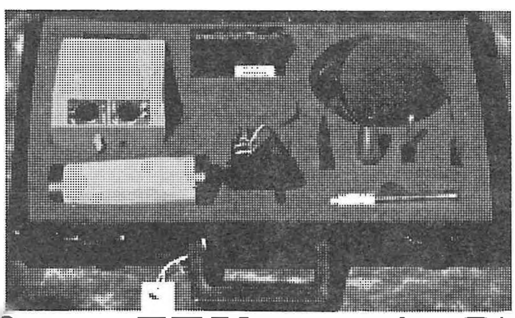

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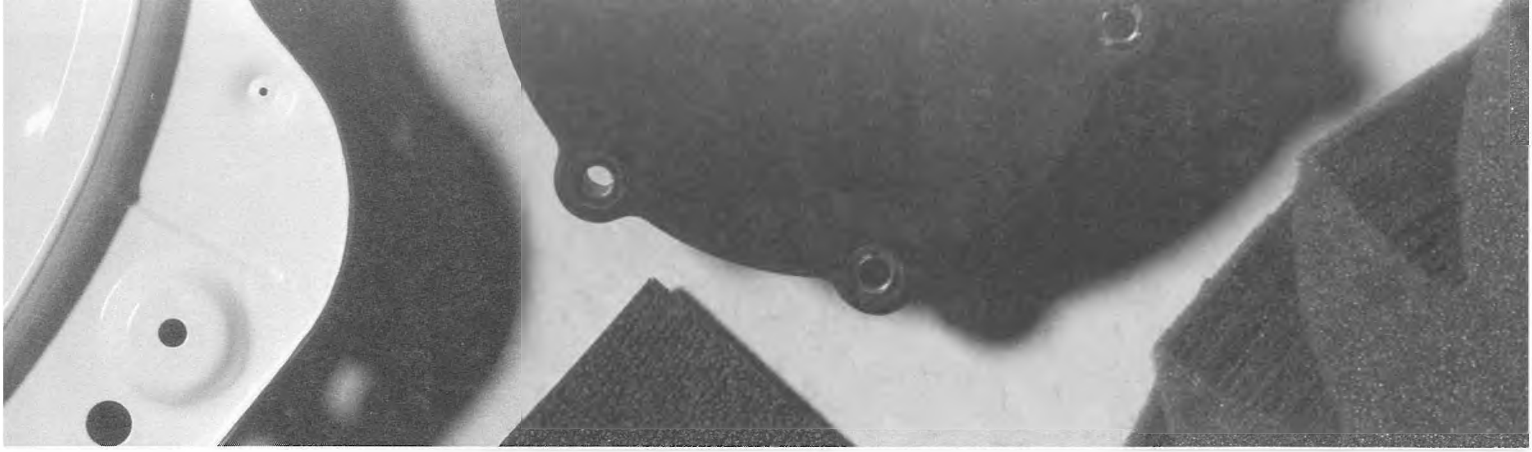

CAA Student Travel Subsidy and Student Presentation Award

Application Form
Deadline for Receipt, August 14, 2002

Procedure
Complete and submit this application to:
Dr. Reina Lamothe, Student Awards CAA-2002
Department of Psychology
University of Prince Edward Island
Charlottetown, PE C1A 4P3

Subsidy cheques will be mailed directly to you by Nov. 30, 2002.

Eligibility Requirements
In order to be eligible for the Travel Subsidy, you must meet the following requirements:

1. Full-time student at a Canadian University
2. Student member in good standing with the Association
3. Distance travelled to the Conference must exceed 100 km (one way)
4. Submit a summary paper for publication in the Proceedings Issue of *Canadian Acoustics* with the applicant as the first author
5. Present an oral paper at the Conference. Due to limited funding, a travel subsidy can only be given to the presenter of the paper though there may be more than one student author.

Section A: All applicants must complete this section
Name of Student: ______________________________
Address (where the cheque is to be sent)______________________________
____
Title of the proposed paper:______________________________
____
Is the paper to be judged in the Student Presentation Awards [Yes/No]
Name and Location of the University ______________________________
Faculty and Degree Being Sought:

Section B: Complete this section only if you are applying for the CAA Student Travel Subsidy

I hereby apply for a travel subsidy from CAA

Proposed Method of Transport to conference ______________________________
Provide least expensive transportation cost______________________________

Date of Departure to and Return from the Conference______________________________

Other Sources of Funding (excluding personal) ______________________________

Signature of Applicant	Signature of University Supervisor
______________________________	______________________________
I certify that the information provided above is correct	I certify that the applicant is a full-time student
______________________________	______________________________
Print Name	Print Name

Subvention de l'ACA pour les Frais de Déplacement des etudiants et prix Récompensant les présentations d'etudiants

Formulaire d'inscription

Date limite de réception, 14 Août 2002

Procédure

Compléter le formulaire et le soumettre à:
Dr. Reina Lamothe, Student Awards ACA-2002
Department of Psychology
University of Prince Edward Island
Charlottetown, PE C1A 4P3

Les chèques de Subvention vous seront directement envoyés avant le 30 Novembre 2002.

Conditions d'Eligibilité

Pour avoir droit à la Subvention pour les Frais de Déplacement, vous devez remplir les conditions suivantes:

1. Etre étudiant à temps plein dans une Université Canadienne
2. Etre Membre de l'ACA
3. La distance parcourue jusqu'à la Conférence doit être supérieure à 100km (aller simple)
4. Soumettre un sommaire en vue de sa publication dans les actes d'Acoustique Canadienne, l'étudiant doit être le premier auteur du sommaire
5. Présenter une communication orale pendant la conférence. En raison du financement limité, une Subvention pour les Frais de déplacement ne peut être attribuée qu'à l'étudiant présentant la communication même si plusieurs étudiants sont auteurs du sommaire.

Section A: Tous les candidats doivent remplir cette section

Nom de l'étudiant:____________________________________

Addresse (où le chèque doit être envoyé):____________________________________

____

Titre de la communication proposée:____________________________________

____

La communication est elle inscrite au concours pour le Prix Récompensant les Communications d'Étudiants [Oui/Non]

Nom et adresse de l'université: ____________________________________

Faculté et niveau d'étude en cours:

Section B: Compléter cette section si vous postulez pour une Subvention des Frais de Déplacement
Je postule par le présent document à une Subvention de l'ACA pour les Frais de Déplacement

Moyen de Transport proposé pour se rendre à la conférence:____________________________________

Fournir le coût de transport le moins élevé:____________________________________

Date de Départ pour la Conférence et de Retour:____________________________________

Autres sources de financement pour le transport (excluyant les sources personnel)____________________________________

____

Signature du candidat	Signature du superviseur
____________________	____________________
Je certifie que les informations fournies ci dessus sont plein	Je certifie que le signataire est un(e) étudiant(e) à temps correctes
____________________	____________________
Nom	Nom

INSTRUCTIONS TO AUTHORS FOR THE PREPARATION OF MANUSCRIPTS

Submissions: The original manuscript and two copies should be sent to the Editor-in-Chief.

General Presentation: Papers should be submitted in camera-ready format. Paper size 8.5" x 11". If you have access to a word processor, copy as closely as possible the format of the articles in Canadian Acoustics 18(4) 1990. All text in Times-Roman 10 pt font, with single (12 pt) spacing. Main body of text in two columns separated by 0.25". One line space between paragraphs.

Margins: Top - title page: 1.25"; other pages, 0.75"; bottom, 1" minimum; sides, 0.75".

Title: Bold, 14 pt with 14 pt spacing, upper case, centered.

Authors/addresses: Names and full mailing addresses, 10 pt with single (12 pt) spacing, upper and lower case, centered. Names in bold text.

Abstracts: English and French versions. Headings, 12 pt bold, upper case, centered. Indent text 0.5" on both sides.

Headings: Headings to be in 12 pt bold, Times-Roman font. Number at the left margin and indent text 0.5". Main headings, numbered as 1, 2, 3, ... to be in upper case. Sub-headings numbered as 1.1, 1.2, 1.3, ... in upper and lower case. Sub-sub-headings not numbered, in upper and lower case, underlined.

Equations: Minimize. Place in text if short. Numbered.

Figures/Tables: Keep small. Insert in text at top or bottom of page. Name as "Figure 1, 2, ..." Caption in 9 pt with single (12 pt) spacing. Leave 0.5" between text.

Line Widths: Line widths in techincal drawings, figures and tables should be a minimum of 0.5 pt.

Photographs: Submit original glossy, black and white photograph.

Scans: Should be between 225 dpi and 300 dpi. Scan: Line art as bitmap tiffs; Black and white as grayscale tiffs and colour as CMYK tiffs;

References: Cite in text and list at end in any consistent format, 9 pt with single (12 pt) spacing.

Page numbers: In light pencil at the bottom of each page.

Reprints: Can be ordered at time of acceptance of paper.

DIRECTIVES A L'INTENTION DES AUTEURS PREPARATION DES MANUSCRITS

Soumissions: Le manuscrit original ainsi que deux copies doivent être soumis au rédacteur-en-chef.

Présentation générale: Le manuscript doit comprendre le collage. Dimensions des pages, 8.5" x 11". Si vous avez accès à un système de traitement de texte, dans la mesure du possible, suivre le format des articles dans l'Acoustique Canadienne 18(4) 1990. Tout le texte doit être en caractères Times-Roman, 10 pt et à simple (12 pt) interligne. Le texte principal doit être en deux colonnes séparées d'un espace de 0.25". Les paragraphes sont séparés d'un espace d'une ligne.

Marges: Dans le haut - page titre, 1.25"; autres pages, 0.75"; dans le bas, 1" minimum; latérales, 0.75".

Titre du manuscrit: 14 pt à 14 pt interligne, lettres majuscules, caractères gras. Centré.

Auteurs/adresses: Noms et adresses postales. Lettres majuscules et minuscules, 10 pt à simple (12 pt) interligne. Centré. Les noms doivent être en caractères gras.

Sommaire: En versions anglaise et française. Titre en 12 pt, lettres majuscules, caractères gras, centré. Paragraphe 0.5" en alinéa de la marge, des 2 cotés.

Titres des sections: Tous en caractères gras, 12 pt, Times-Roman. Premiers titres: numéroter 1, 2, 3, ..., en lettres majuscules; sous-titres: numéroter 1.1, 1.2, 1.3, ..., en lettres majuscules et minuscules; sous-sous-titres: ne pas numéroter, en lettres majuscules et minuscules et soulignés.

Equations: Les minimiser. Les insérer dans le texte si elles sont courtes. Les numéroter.

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Photographies: Soumettre la photographie originale sur papier glacé, noir et blanc.

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L'abonnement pour la présente année est dû le 31 janvier. Les nouveaux abonnements reçus avant le 1 juillet s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de l'Acoustique Canadienne de cette année. Les abonnements reçus après le 1 juillet s'appliquent à l'année suivante.

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