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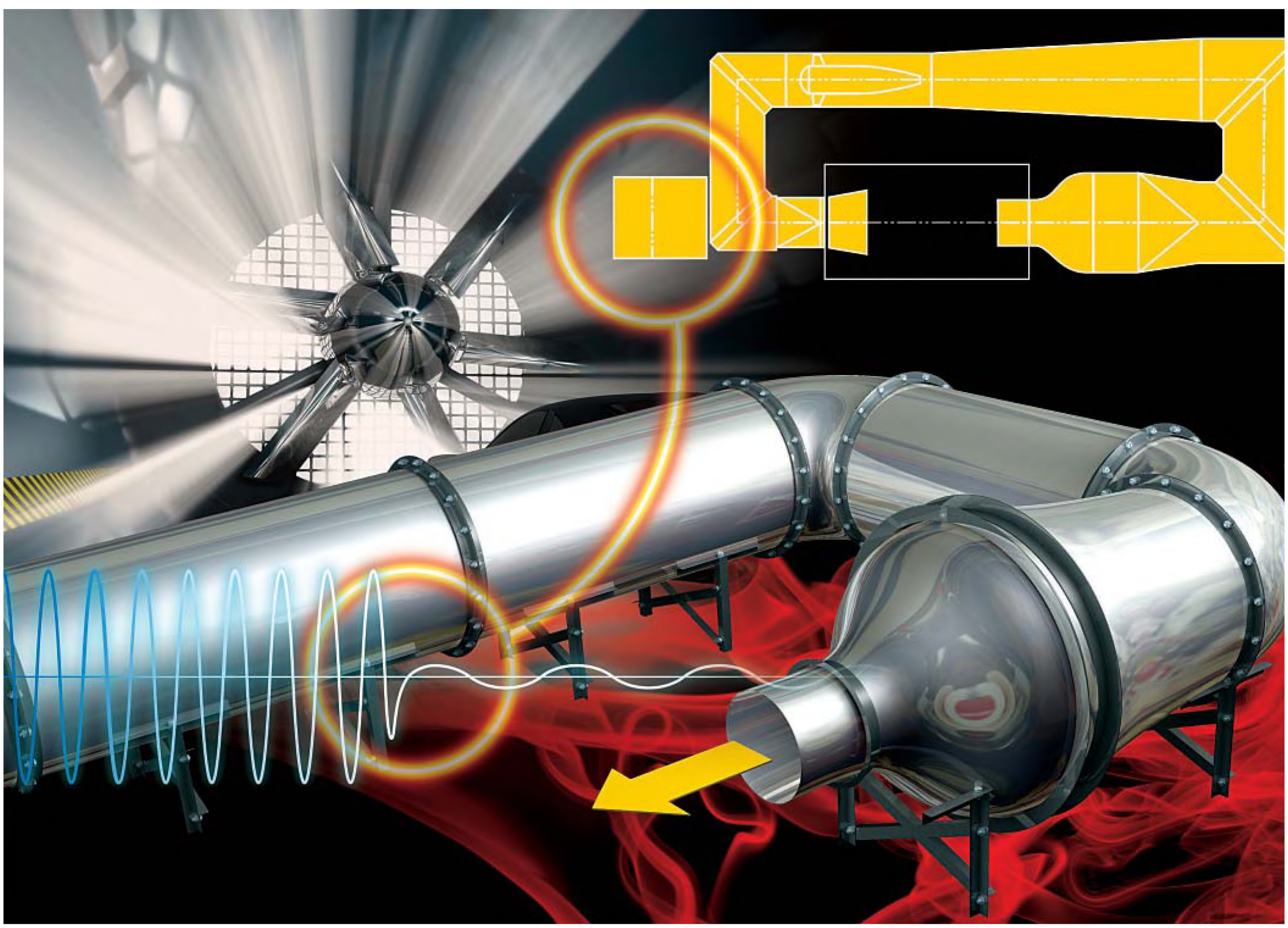

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THE CANADIAN ACOUSTICAL ASSOCIATION
P.O. BOX 1351, STATION "F"
TORONTO, ONTARIO M4Y 2V9

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

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EDITORIAL / EDITORIAL

Encore cette année, le rédacteur en chef m'a demandé de rédiger l'éditorial un plus tôt que prévu. Normalement, le numéro de juin est celui qui me permet de vous adresser un petit mot et de mettre l'emphase sur les articles soumis en français. Cette année, nous avons reçu moins de publications en français et le processus de révision est souvent ardu car nous avons de la difficulté à trouver des réviseurs francophones dans les champs de spécialités des auteurs. Si vous désirez nous donner un coup de pouce dans le processus de révision d'articles, n'hésitez pas à vous manifester, surtout si vous êtes une ressource francophone rare. Avec les agendas surchargés de tout le monde, je sens que Ramani et moi ne recevrons pas des tonnes de courriels, mais on ne sait jamais!

Contrairement à l'an dernier où tous les papiers provenaient de l'extérieur, vous pourrez lire dans ce numéro de mars, des articles provenant de nos collègues canadiens. En effet, nous publions deux articles en provenance de la région de Toronto, un de Kingston (en collaboration avec un collègue d'Halifax), et deux de Windsor. Les champs couverts par ces cinq articles sont très diversifiés, de l'usage des systèmes audio portatifs aux émissions des ventilateurs, en passant par la résonance des souffleries, l'acoustique sous-marine et la psycho-acoustique. Ce numéro est le reflet vivant de la richesse et de la diversité canadiennes dans le domaine de l'acoustique.

Une collègue de DRDC-Atlantique, Francine Desharnais, a gentiment accepté de faire pour vous la revue d'un livre intitulé « Histoire de l'acoustique sous-marine ». Elle semble avoir eu du plaisir à parcourir les nombreux chapitres de ce livre écrit par un collègue de la France. Deux autres critiques de livres anglophones complètent la section des revues de livres. Je profite de l'occasion pour vous inviter vous aussi à nous soumettre une revue critique, si vous mettez la main sur un livre qui a le potentiel d'intéresser nos lecteurs.

Enfin, l'équipe de rédaction a le plaisir de vous informer que vous pourrez prochainement accéder à un gabarit pour préparer les articles en format LaTex et les soumettre en format pdf directement sur le site web de l'association. En espérant que cette nouvelle initiative vous facilitera la tâche au moment de soumettre vos articles.

Au plaisir de vous retrouver en mars ou juin 2008!

Chantal Laroche
Rédactrice adjointe

I usually briefly address you and highlight articles submitted in French in the June editorial but, again this year, the chief-editor has asked me to write the editorial a bit earlier. We've received few French publications and the review process is often arduous given the difficulty of finding French reviewers matching authors' fields of expertise. Should you be interested in helping out in the review process, please do not hesitate to offer your services. Given everyone's overwhelmingly booked agenda, I fear Ramani and I won't be receiving loads of emails. Still, I just couldn't resist this opportunity to encourage your contribution!

In this issue, you can read articles submitted by Canadian colleagues, whereas last year's issue contained solely articles from other countries. Indeed, two published articles originate from the Greater Toronto Area, one from Kingston (in collaboration with Halifax) and two from Windsor, all covering very diverse fields of interest. From portable audio devices, to ventilator emissions, wind-tunnel resonances, underwater acoustics and psychoacoustics, this issue highlights the lively richness and diversity of Canadian acoustics.

Francine Desharnais from DRDC-Atlantic has kindly accepted to review a book entitled « Histoire de l'acoustique sous-marine » [History of underwater acoustics]. She seemingly enjoyed discovering the numerous book chapters written in France. Two additional reviews of English work complete the book review section. I delightfully take this opportunity to invite you to submit critical reviews of books likely to interest our readers.

Finally, the editorial team is pleased to inform you of a new initiative that should facilitate the process of submitting manuscripts. A LaTex template will soon be available to prepare manuscripts, which can be directly uploaded in pdf format on our website.

Looking forward to joining you again in March or June 2008!

Chantal Laroche
Associate editor

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Wind Tunnel Resonances and Helmholtz Resonators

Peter Waudby-Smith[1] and Ramani Ramakrishnan[2]
1- Aiolos Engineering Corp., 2150 Islington Avenue, Toronto, ON
2 - Ryerson University, Department of Architectural Science, 350 Victoria Street, Toronto, ON

ABSTRACT

Open jet wind tunnels can be impacted by low-frequency pressure fluctuations due to different feedback mechanisms coupling with vortices shed from the nozzle exit. These fluctuations can reduce the simulation quality of the wind tunnel and/or reduce the effective wind speed range of the facility. Considerable research has been conducted to understand the on-set of these strong pressure fluctuations and different mitigation methods have been attempted to reduce these fluctuations. The idea of using Helmholtz resonators to provide strong absorption in the low frequency regime has been around for a long time in different acoustical applications. It has been applied in aero-acoustic fields also to control cavity resonances. A parametric study of applying Helmholtz resonators to control the pressure fluctuations in open-jet wind tunnels was undertaken. An existing open-jet tunnel was modified to produce strong low-frequency pressure fluctuations. A control volume to represent a Helmholtz resonator was attached to the wind tunnel. The control volume was made adjustable to produce different tuning frequencies to control fluctuations at different wind speeds. The results of the study are presented in this paper.

SOMMAIRE

Les souffleries aérodynamiques à veine ouverte peuvent être affectées par diverses fluctuations de basse fréquence dues à différents mécanismes de rétroaction engendrés par la configuration de la veine. De telles fluctuations peuvent même endommager la structure de la soufflerie si de fortes résonances en résultent. Ces fluctuations, réduiront habituellement la qualité de simulation de la soufflerie et/ou réduiront la gamme efficace de vitesse de vent de l'installation. Des recherches considérables ont été conduites pour comprendre le debut de ces fortes fluctuations de pression, ainsi différentes méthodes de réduction ont été tentés pour réduire ces fluctuations. L'idée d'utiliser des résonateurs de Helmholtz pour fournir une forte absorption dans le régime de basse fréquence a été depuis longtemps employé dans différentes applications acoustiques. Cette approche a également été appliquée dans les domaines aéro-acoustique pour contrôler les résonances de cavité. Une étude paramétrique concernant l'utilisation d'un résonateur de Helmholtz pour contrôler les fluctuations de pression dans des souffleries aérodynamiques à veine ouverte a été entreprise. Une soufflerie aérodynamique à veine ouverte existante a été modifiée pour produire de fortes fluctuations de pression à basse fréquence. Un volume de contrôle, représentant un résonateur de Helmholtz a été fixé à la soufflerie. Le volume de contrôle a été rendu réglable pour produire différentes fréquences de mise au point et ainsi contrôler les fluctuations à différentes vitesses de vent. Cet article présente les résultats de l'étude paramétrique.

1. INTRODUCTION

Pressure fluctuations are an inherent characteristic of open-jet test sections in wind tunnels. The combination of vortices shed from the nozzle trailing edge combined with the development of shear layers between the main flow and relatively quiescent surrounding plenum region provide a source of flow unsteadiness that can couple with various resonant conditions in the wind tunnel. The result is a potential for generating large pressure fluctuations which can interfere and degrade the measurements on a test object in the test section. The various factors that can influence the severity of these fluctuations have been seriously studied for quite some time and are reported in the literature [1-2]. The above two papers by Michel and Froebel discuss the fluctuations as well as the lowest level that is possible by citing actual levels in wind tunnels. The suppression of these jet pulsations has been a key effort during the design and/or commissioning phases of other open-jet wind tunnels.

A brief review of the jet pulsations and their characteristics such as their on-set, amplitudes and dominant frequencies is presented in this paper. Some of the common suppression methods are also highlighted in this paper. A brief review of Helmholtz resonators as acoustic absorbers and their varied applications is also presented in this paper.

Measured data with a Helmholtz resonator assembly added to an existing open-jet wind tunnel at the National Research Council of Canada in the year 2002 are also described. The details of the flexible resonator and its performance in suppressing the open-jet pulsations are the main focus of the current paper. The results of the above study are discussed in this paper.

2. WIND TUNNEL PULSATIONS

2.1 Background

It is not unusual to experience low-frequency pressure and velocity fluctuations at distinct wind speeds in open-jet wind tunnels. These fluctuations have an impact on the aerodynamic quality of the test section flow, resulting in unsteady and inaccurate measurements of pressures and forces. Even the acoustic measurements are compromised as the noise generated by the test model is modulated at the frequency of the fluctuations.

Most open-jet wind tunnels suffer from the low-frequency fluctuations. Rennie et. al performed a detailed investigation into pressure fluctuations measured in the Hyundai Aero-acoustic Wind Tunnel (HAWT) [3]. The main features of the HAWT are: ¾ open-jet test section with 28 m^2 nozzle area; separate turntables for balance and dynamometer testing; and a relatively long, 18 m test section. Sample results, from Reference 3, of the pressure fluctuations measured in the original configuration of the HAWT test section are shown in Figure 1. It is seen that fluctuation pressure levels are 6% of the dynamic pressure at a speed of 80 km/h and 4.5% at a speed of 130 km/h. The main frequency of these fluctuations is a low value of 1.4 to 1.5 Hz. Fluctuations at a higher frequency of 2.6 Hz were also evident from the measurements, albeit, at a reduced amplitude. High amplitude fluctuations, in excess of 1.5%, are seen at a number of wind speeds, thereby reducing the value of the wind tunnel measurements. The different causes for these fluctuations must be properly accounted for before designing suitable control methods and are highlighted in the next section.

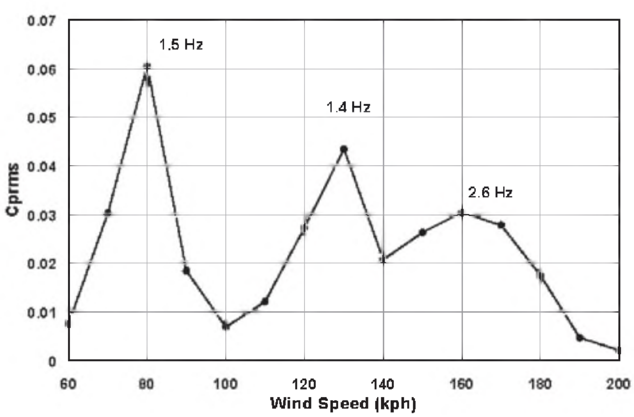

Figure 1. HAWT pressure fluctuations (from Reference 3)

2.2 Physical Mechanisms

Many ideas have been postulated to explain the on-set of the very-low-frequency oscillations in open-jet wind tunnels [4, 5]. Rennie conducted model tests to evaluate the feasibility and performance of different suppression methods to control the pressure fluctuations. The wind tunnel was a 1/7th scale version of the full scale HAWT. This pilot wind tunnel was constructed by modifying an existing tunnel at the Institute of Aerospace Research (IAR) of the National Research Council of Canada (NRC) in Ottawa, Ontario in the year 2000. [For details of the tunnel and its many features please see References 3 and 4]. Even though the main focus was on the development of suppression methods, Rennie conducted detailed analyses of the possible mechanisms for the pressure fluctuations [4]. Descriptions of the resonant frequencies are also given in Wickern et. al. [5].

It is postulated that whenever the large-scale vortices in the open jet mixing layer shed at frequencies that coincide with an acoustic resonant mode of the wind tunnel circuit, high amplitude pressure fluctuations result. There are four possible resonant conditions within the wind tunnel circuit. These conditions are outlined below.

1. Resonance of the complete wind tunnel circuit (organ pipe modes), with resonant frequencies evaluated as

$$f = \frac{n\,c}{2\,L_{circ}} \tag{1}$$

   where c is the speed of sound and L_{circ} is the total length of the circuit.

2. Resonances within the volume of the test section plenum, with frequencies calculated as

$$f = \frac{c}{2\sqrt{\left(\frac{l_x}{n_x}\right)^2 + \left(\frac{l_y}{n_y}\right)^2 + \left(\frac{l_z}{n_z}\right)^2}} \tag{2}$$

   where, lx, ly, and lz are the dimensions of the test section plenum.

3. The nozzle and test-section plenum chamber acting as a Helmholtz resonator. This can be calculated from the suggested equation by Wickern et. al. [5]

$$f = \frac{c\,D_h}{4\pi}\sqrt{\frac{\pi}{V\left(L_{noz} + \pi\frac{D_h}{4}\right)}} \tag{3}$$

   where D_h is the hydraulic diameter of the nozzle, V is the plenum volume, and L_{noz} is the nozzle length.

4. Edgetone feedback mechanism between the collector and the nozzle. A schematic detail of this mechanism is shown in Figure 2. Rennie showed that the feedback is strongly controlled by the jet length between the nozzle and the collector [4].

The measurements were conducted in both the full scale HAWT and the pilot tunnel at IAR of the NRC. The composite results are shown in Figure 3 (with the pilot tunnel frequencies multiplied by the scale ratio of 1/7). The figure also shows the resonance frequencies for the three circuit modes at 0.8 Hz, 1.6 Hz and 2.4 Hz, and the first three edgetone modes, which are a function of wind speed. It can be concluded from the composite data of Figure 3 that at a given wind speed, the pressure fluctuations have high amplitudes if the shedding frequencies are close to the junction of circuit organ pipe modes and the edgetone modes. Another useful conclusion, which has been found by others, is that a geometrically scaled pilot wind tunnel provides representative pressure fluctuation data in magnitude, with the frequencies adjusted by the geometrical scale ratio.

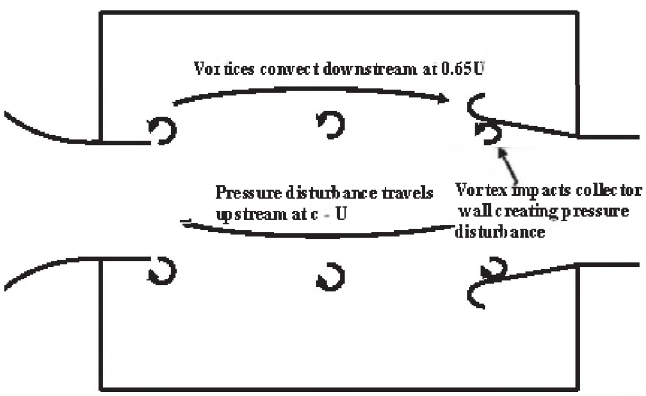

Figure 2. Edgetone feedback loop (from Reference 4)

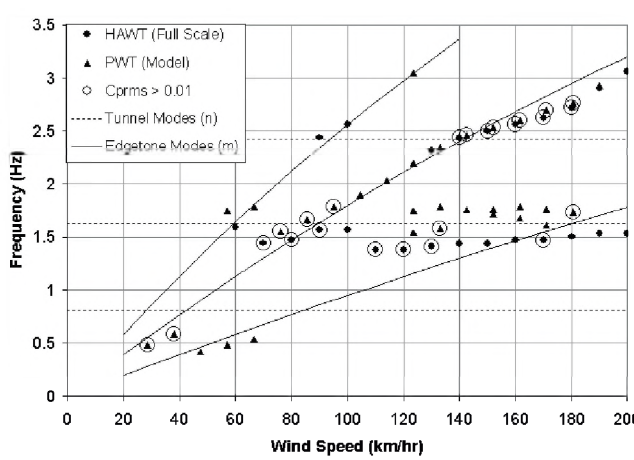

Figure 3. HAWT and 1/7'th scale Pilot Wind Tunnel pressure fluctuations (from Reference 3)

2.3 Suppression Mechanisms

Many different control methods have been attempted to suppress the pressure fluctuations in open jet wind tunnels. A brief review of these techniques will be presented below. Details of one of those techniques will be the main focus of this paper and will be presented in subsequent sections.

The first method applied in general aerodynamic wind tunnels is the installation of vortex generators on the nozzle lip. Vortex generators are a simple way of attenuating the amplitudes of the pressure fluctuations and their success has been proven in many wind tunnel applications. Vortex generators, however, induce a vena contracta type flow at the nozzle exit which results in a negative axial pressure gradient. In addition, they are significant sources of acoustic noise, which is unacceptable for the aero-acoustic demands of most open-jet test sections. As a result, vortex generators are undesirable for most aerodynamic and aeroacoustic open jet wind tunnel applications.

The results of Rennie showed that a properly selected jet length, usually small, can reduce the amplitude of the pressure fluctuations [3, 4]. However, a small jet length limits the size of the models that can be tested in modern automotive

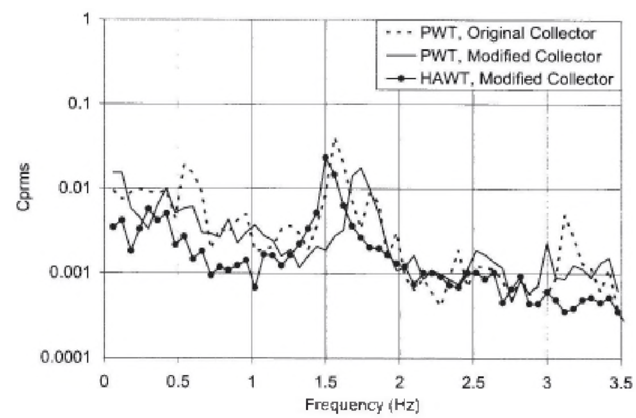

Figure 4a. Measured pressure fluctuation spectra - 80 kmph (from Reference 3)

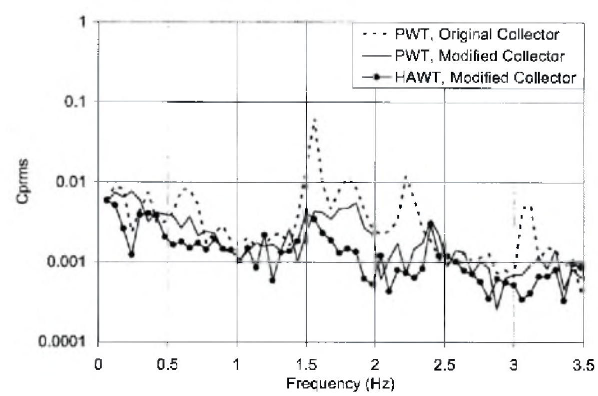

Figure 4b. Measured pressure fluctuation spectra - 130 kmph (from Reference 3)

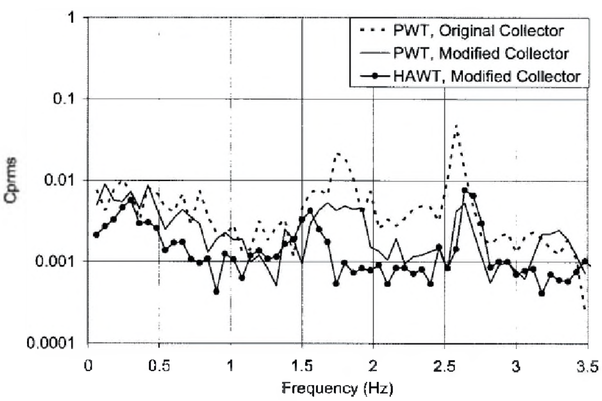

Figure 4c. Measured pressure fluctuation spectra - 160 kmph (from Reference 3)

wind tunnels. Hence, this method of jet length control is not practical in most applications.

Rennie, in the pilot wind tunnel, experimented with collector shapes as well as installing a breather between the trailing edge of the collector and the inlet to the test-section diffuser [3, 4]. The results showed that an optimized collector geometry is one suitable method to reduce the amplitude of the pressure fluctuations (the optimization scheme for the collector is beyond the scope of this paper). The results with the modified collector are shown both for the full scale HAWT and the pilot wind tunnel in Figure 4a to 4c. The results show a large reduction in the amplitude of the resonant peaks (note the logarithmc scale). The pressure fluctuations are seen to be less than 1.5% for the three wind speeds of concern.

Waudby-Smith et.al. reported the results of implementing a similar collector design in the newly constructed GIE S2A aeroacoustic wind tunnel in Paris, a joint project of Renault, PSA Peugeot-Citroën and CNAM [6]. The results, summarized in Figure 5, clearly indicated that with proper design of the collector, the pressure fluctuations can be kept small. The pressure fluctuations at all wind speeds were below 1% except for two low wind speeds. The results were deemed acceptable, since these wind speeds were non-standard test conditions and were at low speed.

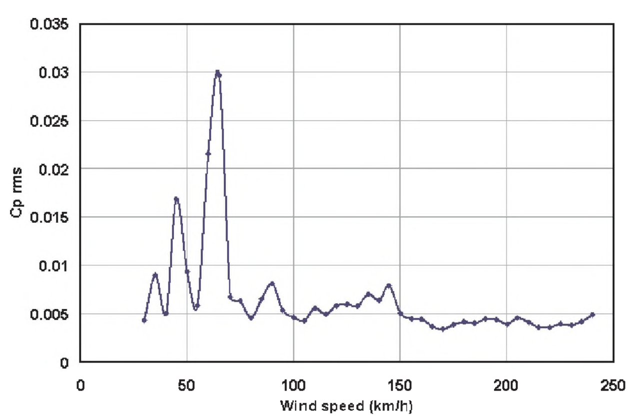

Figure 5. Overall pressure fluctuation amplitudes (from Reference 6)

Attempts were also made to modify the collector configuration by other researchers. Lacey evaluated the impact of an axially slanted collector [7]. Lacey's model tests showed that the slanted collector, with a slant as much as 45º from the vertical, was a promising control method to suppress pressure fluctuations, even for a long test section. According to Lacy the slanted collector does not work by "detuning" the collector feedback frequencies. This conclusion was based on a set of tests with a fixed collector and different test section lengths. In reality the collector size should have increased with increasing test section length. It is difficult to perform a truly parametric investigation into the effects (and optimization) of collector geometry.

Wickern et. al. studied the feasibility of implementing suitable suppression methods when the level of the pressure fluctuations in Audi's new aeroacoustic wind tunnel became a concern during the design stage [5]. It was decided to evaluate the impact of an active resonance control (ANC) system. Pilot wind tunnel studies showed the ANC to be very effective in reducing the wind tunnel fluctuations. The system consisted of a measuring microphone in the test section plenum and a large speaker driver system to generate '180º antiphase' sound levels at the offending frequencies to cancel the pressure fluctuations. The driver system was located in an adjacent room but communicated with Cross-leg 1 of the airline circuit. The sound level reductions at the full scale tunnel resonances at 2.4 Hz, 3.9 Hz and 6.8 Hz were measured to be 23 dB, 20 dB and 15 dB respectively. Velocity fluctuation measurements showed that the three main resonances were completely eliminated when the ANC system was switched on. The cost of the ANC mitigation method can be hundreds of thousands of dollars due to the cost of generating very low frequency sounds and hence ANC may not be cost effective.

Another active flow control method, this time using oscillating flaps at the nozzle exit, was studied by Heesen and Höpfer [8]. Attenuation up to 29 dB of the pulsation amplitudes was achieved in model-scale tests. The applicability of this system to an aeroacoustic wind tunnel, however, has not been demonstrated.

Finally, a Helmholtz resonator configuration has been developed to remove the pressure fluctuations that occur at distinct frequencies. Since this resonator absorber is the main focus of the current paper, a brief overview is given in Section 3. The application of the resonator absorber in the IAR pilot wind tunnel will be described in Section 4.

3. HELMHOLTZ RESONATORS

Helmholtz resonators have been around as both acoustic absorbers and acoustic amplifiers. They have been applied as side branch absorbers to attenuate noise in many different noise control schemes [9 through 15]. The physical characteristic of a Helmholtz resonator consists of a large volume communicating with the main noisy domain through a short neck (small neck area and neck length). The main acoustic characteristic of a Helmholtz resonator-absorber is the ability to tune the resonator to a single frequency. The tuning frequency of a single resonator is evaluated from:

$$f = \frac{c}{2\pi}\sqrt{\frac{S}{lV}} \tag{4}$$

where, f is the tuning frequency, c is the sound speed, S is the neck area, V is the cavity volume, and l is the effective length. The length l is given by

$$l = t + \delta \tag{5}$$

where, t is the neck length and $\delta = 0.85$ a, where a is the neck diameter.

The acoustic absorptive effect of the Helmholtz resonator operates as follows: The air in the neck of the resonator vibrates back and forth somewhat as a single mass, and the larger volume acts as a spring or restoring force. Frictional resistance is encountered by the alternating flow of air in and around the neck. Hence sound energy is absorbed, mainly in the region of the resonant frequency, 'f'. The larger the mass flow, the larger is the aborptive effect of the resonator.

Davis et. al. tested the use of side branch resonators in muffling systems for helicopter engines and evaluated the transmission characteristics [9]. Soderman used the Helmholtz resonators by creating an absorbing volume behind perforated sheets in silencer baffles for attenuating the fan noise of the wind tunnels [10, 11].

Ackermann et. al. developed a novel membrane absorber for attenuating noise in contaminated flow environments [12]. Multiple resonator volumes were created to broaden the frequency range of the absorbers and were successfully implemented in the turning vane-cross legs junction areas of the IVK wind tunnel in Stuttgart.

Ramakrishnan et. al. created an absorbing volume between the structural wall and facing-brick wall in an underground bus station in Ottawa, Ontario [13]. The facing bricks were mortared only horizontally and a small gap was created vertically, with the bricks set-apart from the structural wall. A Helmholtz absorber was thus created and provided absorption coefficient values between 0.7 and 0.85 in the frequency range of 60 to 300 Hz. The frequency range was also widened by placing fibreglass batts in the cavity.

Helmholtz resonators were used as active control of flow-induced cavity resonances [14, 15]. Hsu and Ahuja used simple pharmaceutical syringes as Helmholtz resonators to attenuate cavity resonances induced by flow in their simulated studies of bomb bays of planes [15]. Their attempts to use different volumes to broaden the frequency ranges were partially successful.

4. EXPERIMENTAL MODEL

4.1 Background

The review presented above shows that Helmholtz resonators have a wide application as acoustic absorbers. The current study was the first investigation that attempted systematically to evaluate the performance of Helmholtz resonators to reduce wind tunnel pressure fluctuations.

The motivation to apply the Helmholtz resonators was also based on a simple observation of a well-behaved wind tunnel designed (by engineers of Aiolos Engineering Company) and built more than 15 years ago. The tunnel under discussion had very low levels of pressure fluctuations and the tunnel had two features: a) a short test section and b) a large side room, built adjacent to the centre of Cross Leg 1 and in acoustic communication with the main wind tunnel circuit, that contained cross leg inserts. It was possible therefore that a Helmholtz resonator volume was serendipitously created thereby absorbing the low-frequency resonances of the tunnel. However, no measurements were undertaken to evaluate the performance of the side room functioning as a Helmholtz resonator absorber.

4.2 Pilot Wind Tunnel

As described in earlier sections, Rennie has outlined the design of a pilot wind tunnel which was used to study wind tunnel pulsations [3, 4]. The pilot wind tunnel (PWT) was designed and built as a (1/7)th scale model of the full scale HAWT.

The PWT was used to conduct the current study. The test section and contraction geometries of the PWT were modified following the HAWT tests. The test section remained representative of an automotive open-jet test section. No plenum walls were used (other tests required improved access to the surrounding flowfield), though this was of no consequence for these tests as the plenum resonance modes had higher frequencies than those of interest for the resonator. Initial "proof-of-concept" tests were performed with the resonator in July of 2002. More detailed tests were performed in December 2002 with the collector geometry "de-optimized" in order to enhance the pressure fluctuations in the baseline condition where no resonator is used.

The PWT had an observation window, just downstream of Corner 1 in Cross Leg 1. The observation window was removed and a Helmholtz resonator volume was created by attaching a room constructed from plywood. The window was replaced with a perforated sheet to provide acoustic communication between the resonator volume and the PWT circuit. The end wall of the room was movable, allowing the resonator volume to be modified to tune to different frequencies. The arrangement of the PWT and the resonator are shown in Figure 6.

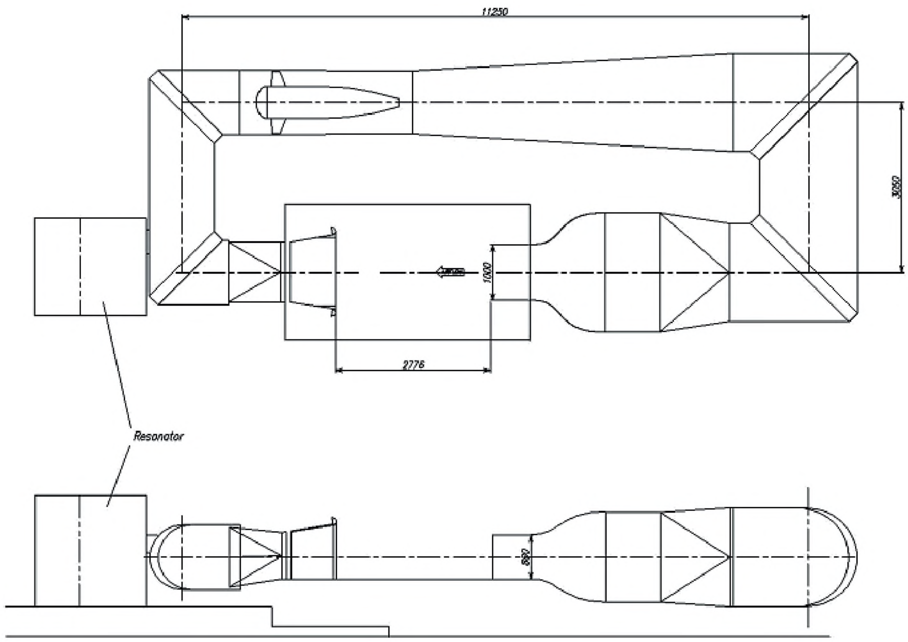

Figure 6. Arrangement of the Pilot Wind Tunnel and Resonator - Top - Plan; and Bottom - Elevation

The resonator used in these tests was located in the first cross-leg of the wind tunnel. This location was chosen because of the existing window, which was converted into the opening for the resonator, and the availability of sufficient space beside the wind tunnel. For other wind tunnels alternative locations are possible including the plenum that surrounds the test section of most wind tunnels.

5. RESULTS AND DISCUSSION

The in-flow static pressure fluctuations measured at the nozzle for the baseline condition (no resonator) are shown in overall Cprms form in Figure 7 and in spectral form in Figure 8. The measurements were made over the frequency range 0.3 Hz to 20 Hz with a 0.0625 Hz interval. The salient observations of these results are: a) Peak fluctuations at 15 m/s and 27 m/s correspond primarily to fluctuations at the first and second organ pipe modes of 6 Hz and 14 Hz respectively; b) Fluctuations can occur for both organ pipe modes at a single wind speed; c) Fluctuations are largest when a nozzle-collector feedback frequency is close to an organ pipe mode; d) Strong resonant frequencies were in the 6 Hz and 14 Hz bands; and e) The data are sufficiently repeatable.

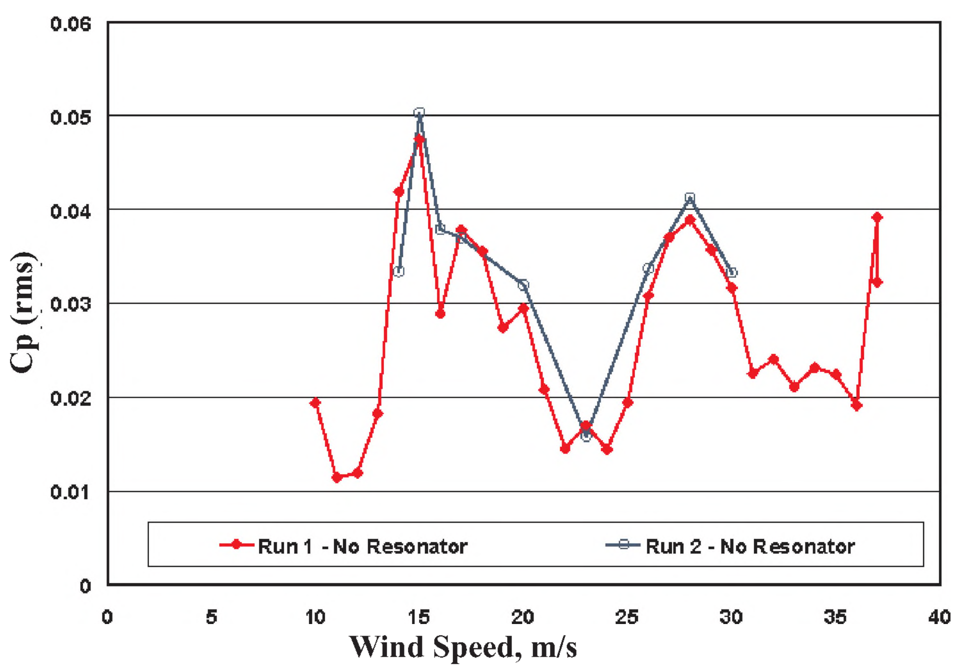

Figure 7. Overall Cp_{rms} values of the baseline conditions of the PWT

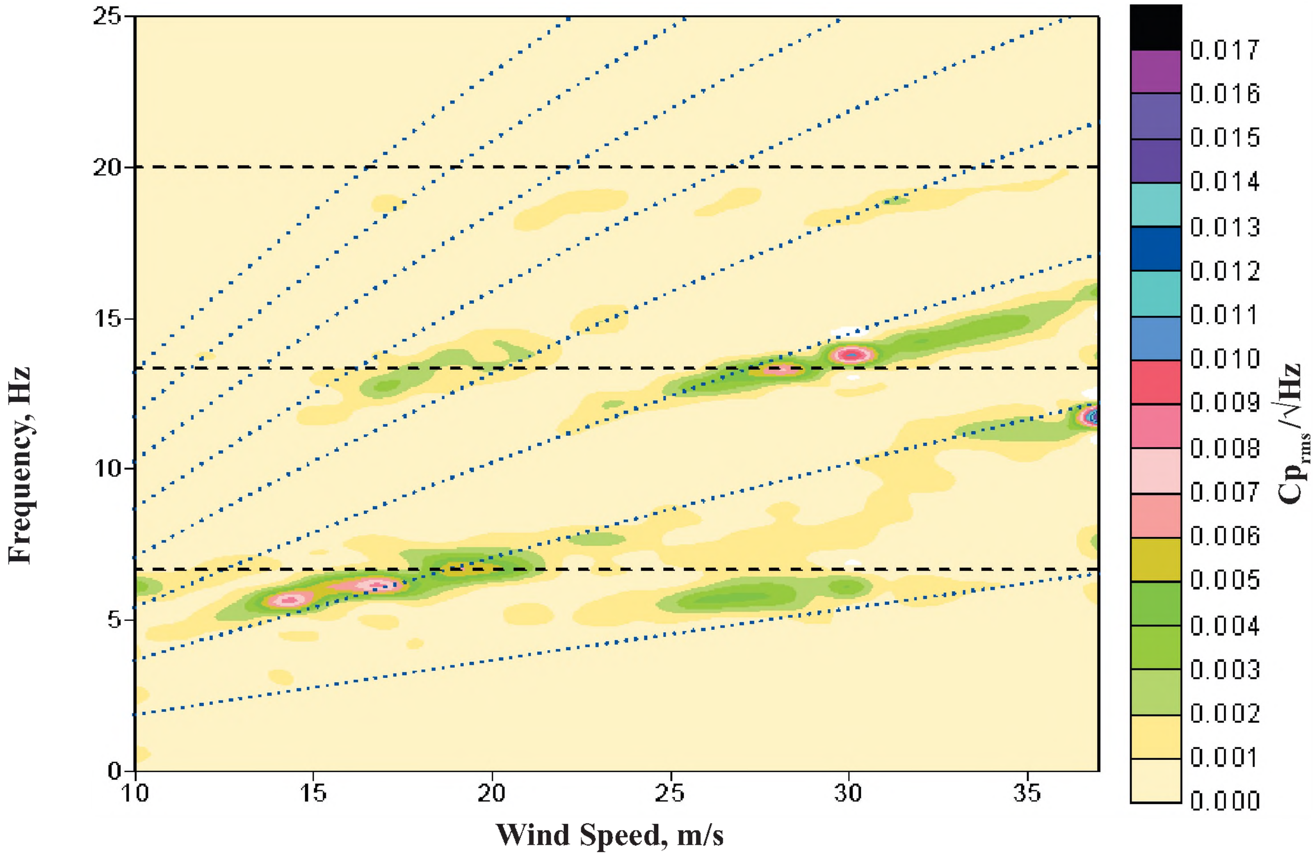

Figure 8. Map of pressure fluctuation spectra with wind speed - No Resonator
Horizontal dashed lines represent the organ-pipe modes and angled dotted lines represent the collector-feedback modes

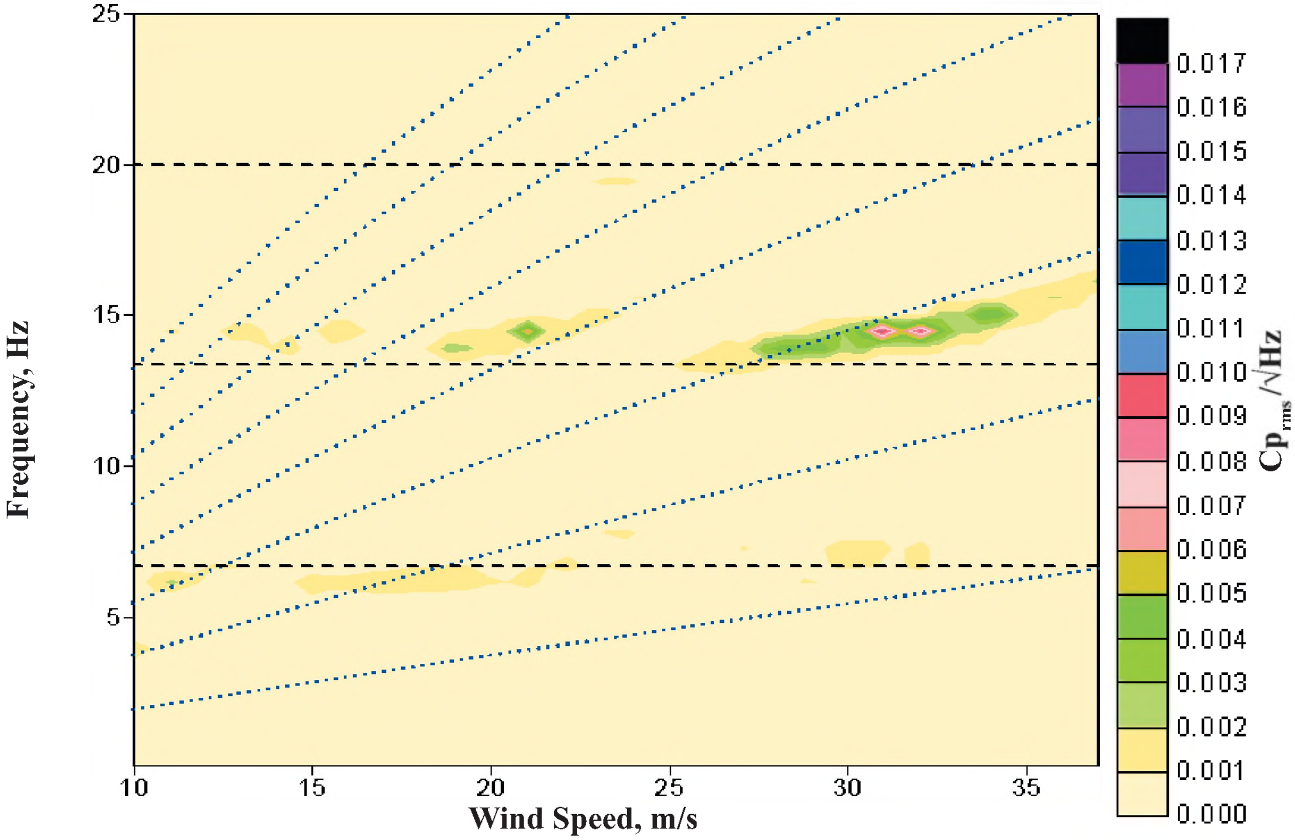

Figure 9. Map of pressure fluctuation spectra with wind speed - 6.5 m^3 resonator
Horizontal dashed lines represent the organ-pipe modes and angled dotted lines represent the collector-feedback modes

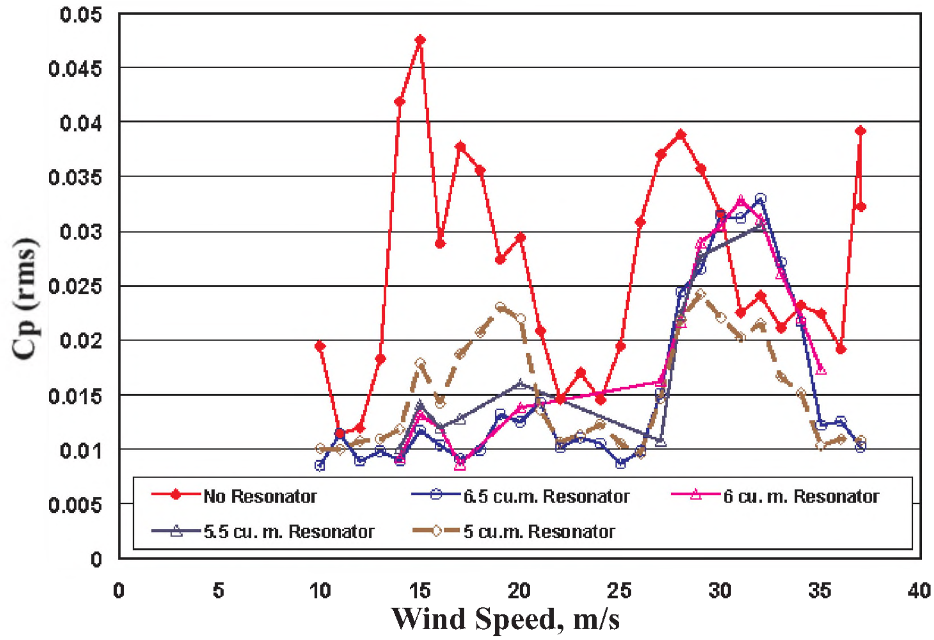

Figure 10. Overall Cp_{rms} values of the of the PWT with four large volume resonators

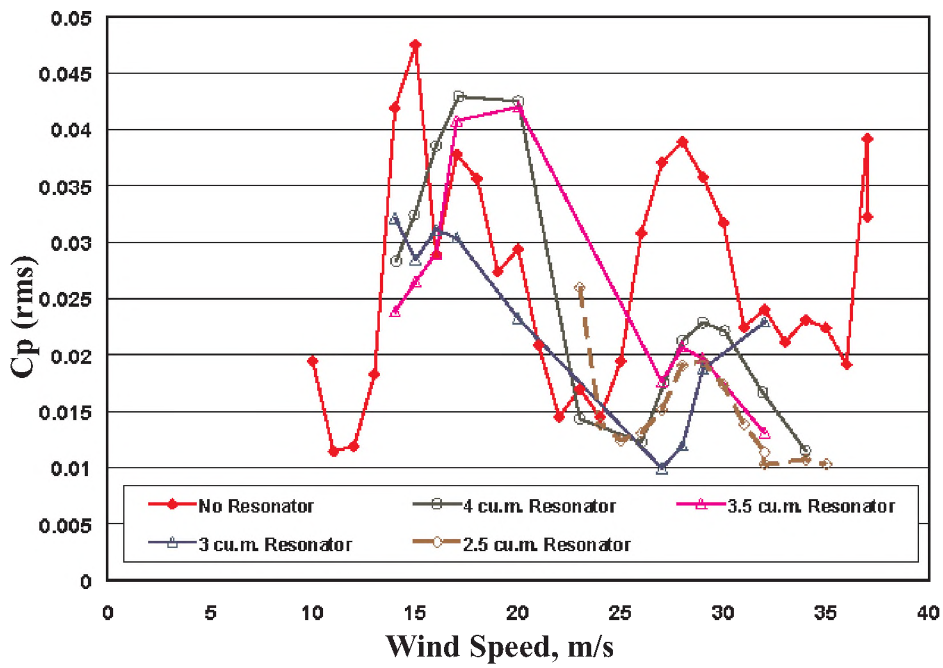

Figure 11. Overall Cp_{rms} values of the of the PWT with four small volume resonators

Measurements were performed across the wind speed range of the PWT for several different resonator volumes. Some of the salient results are presented in Figures 9 through 11.

The effect of the largest resonator, with a volume of 6.5 m^3 is shown in Figure 9. The results show that the large resonator was able to eliminate the pressure fluctuations in the 6 Hz band. However, the fluctuations generated at higher speeds in the 14 Hz band were still excited and the 6.5 m^3 volume appeared to have had little effect on these higher frequency fluctuations.

The volume of the resonator was adjusted sequentially downwards and the composite results for four large resonators are shown in Figure 10. The four resonator volumes were: 6.5 m^3, 6.0 m^3, 5.5 m^3, and 5.0 m^3 respectively. The 6.5 m^3 resonator was able to remove the lower frequency fluctuations successfully. As the resonator volume was reduced, the elimination of the lower frequency fluctuations became less prominent. At a volume of 5.0 m^3, the resonator was able to remove some of the higher frequency fluctuations at higher wind speeds but at the expense of reduced attenuation of the lower frequency components (and consequently higher wind speed fluctuations).

Results for further volume reductions are shown in Figure 11. The four small resonator volumes were: 4.0 m^3, 3.5 m^3, 3.0 m^3, and 2.5 m^3 respectively. The smaller resonators are seen to have reversed the trends of the larger resonators. These four resonators reduced the higher frequency fluctuations excited at the higher wind speeds but were ineffective at the lower wind speed resonances. Further, the attenuation of the pressure fluctuations provided by the larger resonators was more than the attenuation provided by the smaller resonators. The most efficient small resonator for this application was the one with a volume of 2.5 m^3.

6. CONCLUSIONS

A brief overview of mechanisms to suppress low frequency pressure fluctuations in open-jet wind tunnels showed that Helmholtz resonators could be a useful method. A pilot wind tunnel with a representative automotive test section was used, though with a collector that was not optimized for low pressure fluctuations. An adjustable resonator was installed and tests with variable volumes showed that Helmholtz resonators could be tuned to effectively suppress the key low frequency fluctuations.

7. ACKNOWLEDGEMENTS

The authors would like to thank Dr. Mark Rennie, who performed the tests in the IAR PWT. In addition, thanks are due to Gerrit Zorgdrager, who promoted the idea of using a resonator to control open-jet pressure fluctuations.

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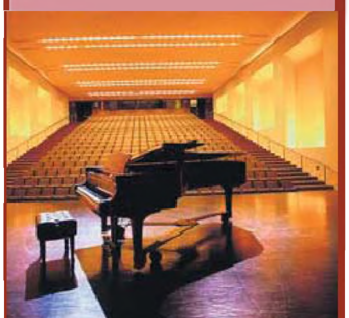

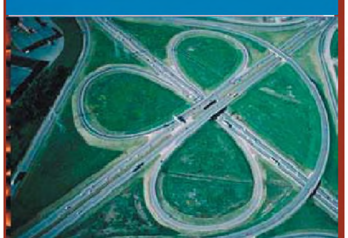
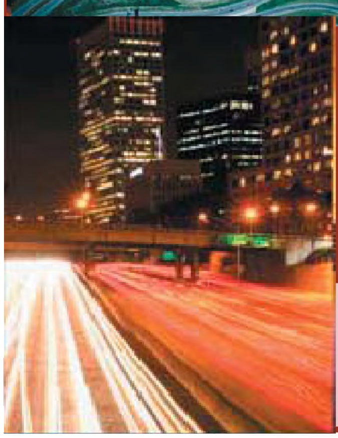

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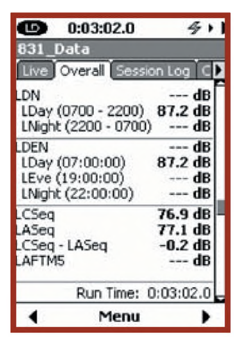

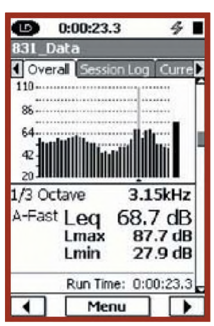

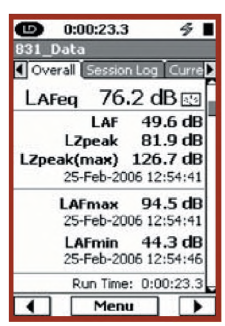

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HIGH-RESOLUTION B-SCAN BEARING ESTIMATION USING THE FAST ORTHOGONAL SEARCH

Donald R. McGaughey[1], Filip Bohac[2], and Richard F Marsden[1]
1 -Royal Military College of Canada, P.O. Box 17000, Station Forces, Kingston, Ontario
2 - Acoustic Data Analysis Centre (Atlantic), P.O Box 99000, Station Forces, Halifax, Nova Scotia

ABSTRACT

A directional DIFAR passive sonobuoy has an omni-directional channel and two orthogonal directional channels. Acoustic targets can be detected on a plot of the power-spectral density of the omni-directional channel called a LOFARGRAM. In addition, a bearing estimate can be formed by the arctangent of the sine channel over the cosine channel (B-scan estimate). Traditionally, the fast-Fourier transform has been used to perform the estimate of the power-spectral density. The fast orthogonal search (FOS) algorithm has been to shown give a higher resolution spectral estimate than the FFT algorithm. The power-spectral estimate of the FOS algorithm has been used instead of the FFT in generating the LOFARGRAM and B-scan bearing estimates. It is shown in simulation and experimental data that using the FOS algorithm allows two targets, whose frequencies are spaced closer than the FFT resolution, to be resolved. By resolving two targets, two bearing estimates are calculated and the resulting estimates are much more accurate than the FFT based B-scan bearing estimates.

SOMMAIRE

Un DIFAR directionnel à bouée sonar passive a un canal omnidirectionnel et deux canaux à directions orthogonales. Les cibles acoustiques peuvent être détectées à partir d'une représentation graphique de la densité spectrale de puissance du canal omnidirectionnel, appelée LOFOGRAM. En outre une estimation du roulement peut être obtenue a partir de l'arctangente du canal sinus divisé par le canal cosinus (estimation de type B-scan). L'algorithme de Fast Fourrier Transform est traditionnellement utilisé pour parvenir à une estimation de la densité spectrale de puissance. L'algorithme de Fast Orthogonal Search (FOS) produit une estimation de plus haute résolution spectrale que l'algorithme de FFT. L'estimation de la densité spectrale de puissance obtenue par l'algorithme de FOS a été utilisée à la place de la FFT pour la génération du LOFARGRAM et pour les estimations par B-scan de roulement. Il est démontré à partir de simulations et de données expérimentales que l'utilisation de l'algorithme FOS permet la détermination de deux cibles dont les fréquences sont plus proches que la résolution de la FFT. A partir de la détermination de deux cibles, deux estimations du roulement sont calculées de manière beaucoup plus précise que par l'estimation du roulement par B-scan basé sur la FFT.

1. INTRODUCTION

A directional frequency analysis and recording (DIFAR) sonobuoy is a passive acoustic sensor with three co-located hydrophones. The omni-directional sensor measures the overall acoustic pressure, while the sine and cosine sensors are orthogonal directional sensors [1]. Many Navies use the expendable DIFAR sonobuoy to detect and estimate the bearing of potential targets.

Traditionally, a power spectral density (PSD) of the omni channel is used to detect the presence of a target. The PSD is calculated over a short time-period and the PSD is plotted as a single line on a graph with the intensity of the PSD scaled as brightness. Each new PSD slice is placed at the bottom of the image – displacing all other slices upwards The general term for a frequency versus time plot is a low-frequency analysis and recording spectrogram (abbreviated as LOFARGRAM), or simply a gram [1].

The incident direction of an incoming acoustic signal can be computed using the arctangent of the cross-correlations of the sine channel over the cosine channel. A display of bearing versus frequency is known as a B-scan display [2]. Targets can be localized by operator identification of frequencies of interest and finding the corresponding bearing from the B-scan display.

These techniques require a trained operator to analyze the results. Although newer algorithms, such as the maximum likelihood adaptive beamforming (MLAB) [3], exist, the LOFARGRAM and B-scan algorithm are still used extensively. Thus in this paper, we present a simple modification to the LOFARGRAM and B-scan algorithms that improves the ability to detect and estimate the bearing of two targets that are narrowly separated in frequency.

The arctangent bearing calculation technique introduces errors when two sources in the same frequency bin lie at different bearings [3, 4]. In the case of sources of similar amplitudes, the single bearing will lie between the actual bearings of the two sources. If the two sources are of unequal amplitudes, the calculated bearing will lie closer to the stronger source, an effect known as bearing bias.

The PSDs required for the LOFARGRAM and B-scan are typically calculated using the fast Fourier transform (FFT) algorithm. It is well known that the frequency resolution of the FFT is inversely proportional to the record length of the time series [5, 6]. Thus, if two narrowband sources are in the same frequency bin, they will not be able to be separated in the LOFARGRAM and the B-scan will exhibit bearing bias as discussed above.

For a sampling frequency of f_s and a record length of N samples, the FFT resolution is known to be f_s/N [5, 6]. The fast orthogonal search (FOS) algorithm has been shown to give a PSD estimate with up to 10 times the resolution of the FFT [7-11]. Note that the FOS frequency resolution is dependent upon the signal and noise present in the record being modelled. Thus targets which may be in the same bin in the FFT spectral estimate may be resolved by the FOS spectral estimate.

In this paper the FOS spectral estimate has been used instead of the FFT spectral estimator in the LOFARGRAM and B-scan algorithms. The increased frequency resolution allows the resolution of two targets that are in the same FFT bin but not the same FOS bin. In addition, in this scenario, two unbiased directional estimates can be found by the B-scan algorithm.

In Section 2 the LOFARGRAM and B-scan algorithms are given in detail. In Section 3 a brief overview of the FOS algorithm for use in spectral estimation is provided. In Section 4, the modified B-scan and LOFARGRAM that use FOS for the spectral estimate are explained. Section 5 provides a comparison of the FOS-based and FFT-based algorithms on simulated data, while Section 6 compares the algorithms on real data from a sonobuoy. Section 7 provides results and conclusions.

2. LOFARGRAM AND B-SCAN ALGORITHMS

The output of the DIFAR sonobuoy is three separate time series: one for each the omni, sine and cosine channels respectively. The three time series are related by:

$$\begin{aligned} &x_{ok} && \text{Omnidirectional Channel} \\ &x_{sk} = x_{ok}\sin\theta && \text{Sine (East-West) Channel} \\ &x_{ck} = x_{ok}\cos\theta && \text{Cosine (North-South) Channel} \end{aligned} \quad (1)$$

where x_o, x_s, and x_c are the omni, sine and cosine channels respectively and k is the sample number.

Targets of interest are located by plotting the power spectral density (PSD) of the omni channel and locating the frequencies which contain the highest energies.

For a sampled time series, the discrete Fourier transform (DFT) can be computed using [5]

$$X(k) = \frac{1}{N}\sum_{n=0}^{N-1} x(n)e^{-2\pi jkn/N} \quad (2)$$

In general, the power spectral density (PSD) is obtained from [5]

$$\Phi(k) = X(k)X(k)^* = |X(k)|^2 \quad (3)$$

where *X(k)* is the Fourier transform of the time series x(n), and * indicates a complex conjugate.

The frequency resolution is defined as the minimum separation between two sinusoidal frequencies such that both frequencies can still be identified. For the DFT, the frequency resolution can be shown to be [5]

$$f_{res} = f_s/N = 1/NT_{sample} = 1/T_{record} \quad (4)$$

where f_s is the sampling frequency, N is the number of samples in the time series, and T_{sample} and T_{record} are the sampling and record period respectively. For example, a desired resolution of 0.25 Hz requires a sampling time of 4 seconds.

The LOFARGRAM requires the PSD of the omni channel which is given by

$$\hat{\Phi}_{oo} = \langle X_o X_o^* \rangle = \langle |X_o|^2 \rangle. \quad (5)$$

where the angle brackets denote ensemble averaging, and the caret (^) indicates an estimated value. The frequency index *k* has been dropped for convenience.

Likewise, a PSD is computed for the cross-spectra between the channels

$$\begin{aligned} \hat{\Phi}_{os} &= \langle X_o X_s^* \rangle = \langle X_o X_o^* \sin\theta \rangle \\ \hat{\Phi}_{oc} &= \langle X_o X_c^* \rangle = \langle X_o X_o^* \cos\theta \rangle \end{aligned} \quad (6)$$

where $\hat{\Phi}_{os}$ and $\hat{\Phi}_{oc}$ are the *omni-sine* and *omni-cosine* cross-spectra, respectively.

Since a single time-series is being processed, a time-average of non-overlapping segments is used instead of an ensemble average. It is known that the variance of the PSD estimate from one segment is proportional to the magnitude of the PSD [5]. To reduce this uncertainty, typically the PSD from M non-overlapping time sequences are averaged, reducing the variance of the estimate by 1/M. Windowing was not used in the PSD estimates of the FFT based LOFARGRAM.

The cross spectra are used to calculate the direction of arrival to a target of interest (TOI). The quotient of the omni-sine and omni-cosine PSD estimates yields a bearing from the simple trigonometric identity

$$\hat{\theta} = \arctan\left(\frac{\hat{\Phi}_{os}}{\hat{\Phi}_{oc}}\right), \tag{7}$$

where the caret (^) indicates an estimate.

The value of $\hat{\theta}$ provides a single direction estimate between 0 and 2π for each frequency bin using a four-quadrant arctangent calculation.

The output of the bearing calculation can be displayed similar to a LOFARGRAM. For a given PSD estimate, each frequency bin will have one bearing associated with it. With the frequency on the x-axis and the bearing on the y-axis, the plot indicates a bearing estimate for every frequency bin; this display format is referred to as a B-scan plot.

A user-selectable threshold can be applied to the B-scan display to show only bearings that have sufficient amplitude on the LOFARGRAM to be of interest. This threshold is generally set to remove all bearings from the B-scan display that are not of interest; this assumes that signals-of-interest are of significantly greater amplitude than undesirable narrowband or broadband acoustic noise.

3. FOS ALGORITHM

The fast orthogonal search (FOS) [7-15] is a general purpose modelling technique which can be applied to the estimation of difference equations, sums of exponential functions, and sinusoidal time-series models. The algorithm works by minimizing the mean-squared error (MSE) between a functional expansion of the time-series and the observed data.

An input function $y(n)$ is fit to a functional expansion

$$y(n) = \sum_{m=0}^{M} a_m p_m(n) + \varepsilon(n) \tag{8}$$

where $p_m(n)$ are arbitrary functions, a_m are the weights of the functional expansion, and $\varepsilon(n)$ is the modelling error.

FOS begins by creating a functional expansion using orthogonal basis functions such that

$$y(n) = \sum_{m=0}^{M} g_m w_m(n) + e(n) \tag{9}$$

where $w_m(n)$ is a selected orthogonal function, g_m is the weight of the function, and $e(n)$ is an error term. The number of terms (M) required to estimate the input signal with a chosen degree of error is defined as the model order.

The orthogonal functions $w_m(n)$ are derived from the candidate functions $p_m(n)$ using the Gram Schmidt (GS) orthogonalization algorithm. In this process, the GS algorithm computes a set of weights α_{mr}. The orthogonal functions are implicitly specified by the weights α_{mr} and need not be computed explicitly.

In its last stage, FOS calculates the weights of the functional expansion, a_m, from the weights of the orthogonal series expansion, g_m, and the weights α_{mr}, calculated by the GS algorithm.

A parsimonious model can be created by fitting terms which reduce the mean squared error (MSE) in order of their significance. The FOS search algorithm can be stopped:

(a) when a certain number of terms is fitted;

(b) when the ratio of MSE to the mean squared value of the input signal is below a threshold; or

(c) when adding another term to the model reduces the MSE less than adding white Gaussian noise.

Spectral analysis with FOS is accomplished by selecting candidates $p_m(n)$ that are pairs of sine and cosine terms at each of the frequencies of interest. Assuming a DC term is fitted as the first term, the candidate functions $p_m(n)$ are given by

$$\begin{aligned} p_0(n) &= 1 \\ p_{2m} &= \cos(\omega_m n) \\ p_{2m+1} &= \sin(\omega_m n) \end{aligned} \tag{10}$$

where $m = 1,\ldots,P$, ω_m is the digital frequency of the candidate pair and P is the number of candidate pairs.

By fitting a sine and cosine pair at each candidate frequency, the magnitude and phase at the candidate frequency to be determined using the identities:

$$\begin{aligned} &A\cos(\omega_m n) + B\sin(\omega_m n) = C\cos(\omega_m n + \theta) \\ &C = \sqrt{A^2 + B^2} \\ &\theta = \tan^{-1}(B/A) \end{aligned} \tag{11}$$

FOS has been shown to be able to resolve two frequencies in the same FFT bin with up to 10 times the FFT resolution [10]. The improved resolution of FOS is achieved at the expense of computational complexity: the FFT computation is of order $n \log n$, FOS is of order Pn.

4. FOS BASED LOFARGRAM AND B-SCAN

The FOS algorithm can be used to estimate the PSD for the LOFARGRAM and B-scan algorithms. Unlike the FFT, FOS does not estimate the spectrum for every candidate frequency but fits a model with only significant, non-noise terms. The spectrum in the other frequency bins is assumed to be zero.

Although the omni, sine and cosine channels have the same frequency components, it has been observed that the FOS algorithm may select different candidate frequencies in the model for each channel due to the effects of noise. For the bearing estimate given in Eq. (7), the cross-spectral estimates of the omni-sine $\hat{\Phi}_{os}$ and omni-cosine $\hat{\Phi}_{oc}$ channels are required at exactly the same frequency. If the FOS models for the sine and cosine channels do not have calculated values at exactly the same frequencies, then the bearing estimates cannot be computed.

Since FOS creates a model based on the candidate functions made available to it, the algorithm can be forced to use certain candidates in the model, regardless of their MSE reduction.

From Eq. (1) it can be seen that the omni DIFAR channel will have the largest amplitude of the three channels. FOS will fit the strongest 25 frequency pairs in the omni-channel or it will stop when adding a new frequency pair fits less energy than adding a WGN term. These frequency pairs are then force-fit in the model for the spectral estimate of the sine and cosine channels. The result is a model that contains the same significant frequencies consistently across the three DIFAR spectral estimates allowing bearing estimates in these channels.

5. Simulation Results

In the first simulation, two target signals that are too closely spaced to be discernible by DFT (but are discernible by FOS) are present. The second case involves using a frequency spacing for the two signals such that they are un-resolvable by either FOS or DFT (i.e., they both lie in the same DFT and FOS bin). For B-scan, this implies that only one bearing can be found for the two sources.

Each target signal contains several narrowband frequency tones. Each signal consists of a time series created from a sum of sinusoids with noise added:

$$y(t) = \sum_{n=1}^{n_p} A_n \sin(2\pi F_n t + \varphi_n) + n(t), \tag{11}$$

where A_n and F_n and φ_n are the amplitude, frequency, and phase of the a single tone, n_p is the number of sinusoidal frequencies, and $n(t)$ is omni-directional background noise. The phase added to each individual target signal was uniformly distributed between 0 and 2π.

Two independent noise sources were added: the first to approximate propagation noise in the ocean, and the second to simulate receiver noise for each element of the array. Ambient noise is most certainly not white or Gaussian in nature [1], but it is a sufficient approximation for the purpose of this signal processing trial.

As seen in Eq. (1) the DIFAR signal is composed of 3 time series channels related to the omni channel such that

$$\begin{aligned} x_{\text{omni}}(t) &= \sum_{t=1}^{T} y_t(t) + n_{\text{o}}(t) + n_{\text{o,r}}(t) \\ x_{\sin}(t) &= \sum_{t=1}^{T} y_t(t)\sin\theta_t + n_{\text{o}}(t)\sin\theta_t + n_{\text{s,r}}(t) \\ x_{\cos}(t) &= \sum_{t=1}^{T} y_t(t)\cos\theta_t + n_{\text{o}}(t)\cos\theta_t + n_{\text{c,r}}(t) \end{aligned} \tag{12}$$

where *y(t)* is the synthetic signal at the source, *x(t)* is the signal at each channel, n_o is the ocean noise, $n_{o,r}$, $n_{s,r}$ and $n_{c,r}$ are the receiver noise for the omni, sine and cosine channels respectively. Here, θ_t is the chosen direction-of-arrival angle of the synthetic target.

The signal-to-noise ratio (SNR) is defined as

$$SNR = 10\log\left(\frac{\overline{x_{omni}(t)^2}}{\sigma^2_{noise}}\right) \tag{13}$$

where σ^2_{noise} is the variance of the noise and the overbar indicates the time-average. The variance of the noise approximating the ocean noise was adjusted to provide the amplitude of the noise *n(t)* to achieve the chosen SNR for each signal generated.

Two targets were introduced in the input signal whose powers are equal and bearing separation was varied from 180 degrees to 25 degrees at increments of 5 degrees. The detection was considered successful if the both estimated bearings were within 10 degrees of their respective input signal bearing. This was chosen since ±10 degrees is the operationally accepted accuracy of a DIFAR sonobuoy [2]. In both simulations, 1000 realizations at each bearing were simulated in calculating the percent success.

For two targets with a bearing separation of 20 degrees or less, the bearings are spaced closer than the error tolerance of ±10 degrees for each estimate. Hence 25 degrees was the minimum used in the simulations.

5.1. Two Sources - Separable

A time series was generated which contained two targets, one with a tonal at 54.9Hz and the other at 55.0Hz (i.e., a frequency separation of 0.1Hz). The bearing separation of the targets was varied from 180 to 25 degrees in 5 degree increments. The record length was 10 seconds (the entire time series required). Each 2-second time slice was extracted from the series as required, and the PSD for each channel was created from a five-slice average. The FFT and FOS resolutions (based on the input time series length) were 0.5Hz and 0.1 Hz respectively. FOS was stopped when 25 frequency pairs were fitted or when adding a term fitted no more energy than adding WGN

Sample plots of the bearing estimate with two sources separated by 90 degrees with an SNR of 3dB are in **Figure 1**. The figure shows the B-scan and LOFARGRAM displays for a representative spectral and bearing estimate. In the gram plots, a gray scale is used for the magnitude of FFT or FOS respectively. The gram plots are auto-scaled so that the largest magnitude in each gram is black and the lower magnitudes are shades or gray, while the lowest energy is white. In this case, the FFT bearing estimate shows two bearings, one of which appears correct. The bearing at 100 degrees (to the left of the input bearings) that appears correct is caused by spectral leakage, and as a result it is not in the bin where the signal is expected to be found. Since the energy in the Gram is low (below a threshold) this bearing estimate will discarded as noise and typically removed from the B-scan plot. By contrast, the FO estimate (at right) has the input (diamond) and estimate (› nearly co-located, indicating that both bearings are corre within ±10 degrees of the true bearing.

Figure 2 is a plot of the percent success for the B-sca technique for 1000 simulations at each bearing. Recall, a estimate is considered a "success" when the bearir estimate to both targets is within 10 degrees of the tru bearing of the respective target. Note that the FFT B-sca method does not provide any successful estimates fo SNR>0 dB. By contrast, the FOS B-scan estimates vai from nearly 100 percent success and decrease with SNI The plots are for four sample bearing separations of 75, 5: and 35, and 25 degrees, and nearly overlap. The variatic of the curves and their overlap is due to nois

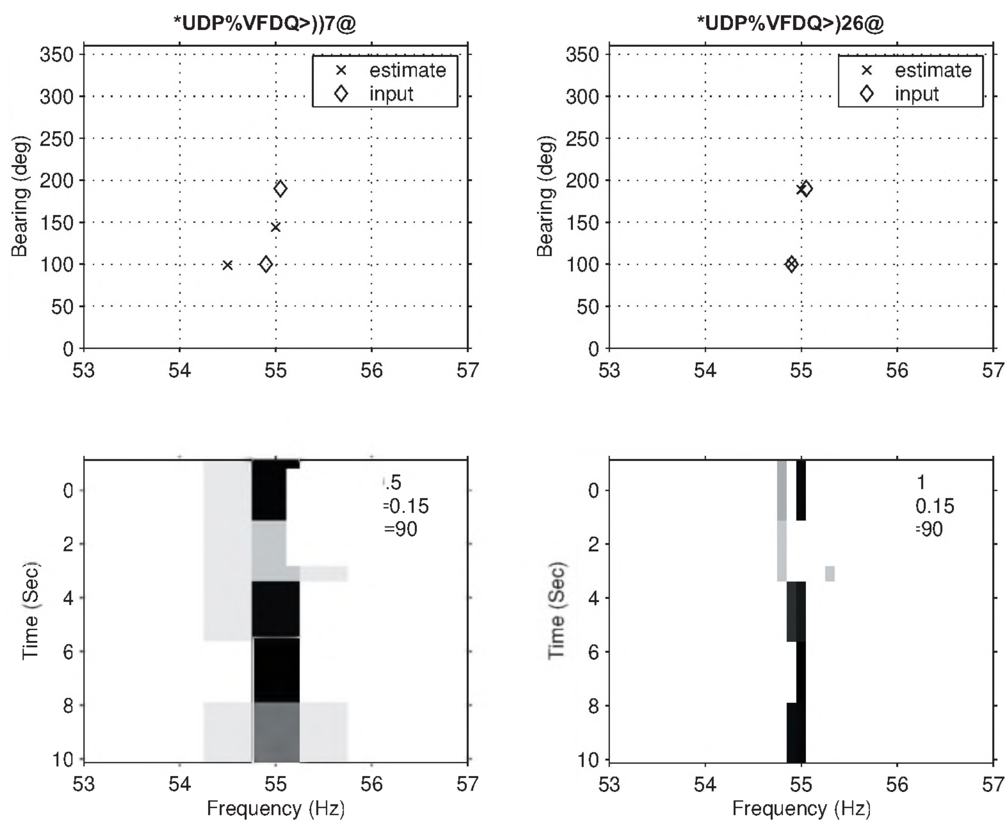

Figure 1 The top left plot is the B-scan and bottom left plot is the Gram for the FFT method. The top right is the B-scan and botto right is the Gram for the FOS method.

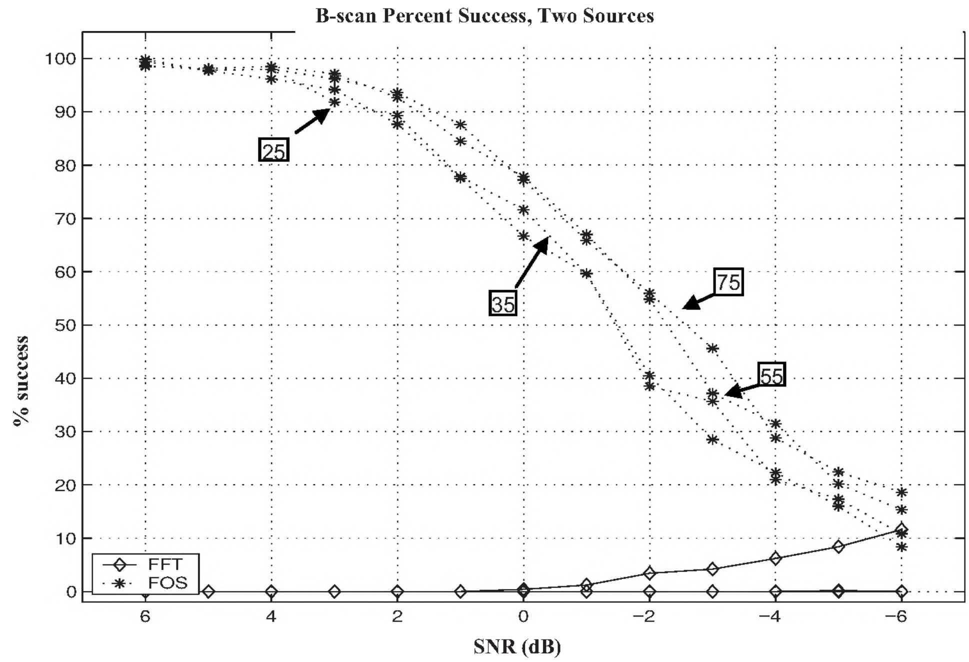

Figure 2: The four lines marked with the *, from top to bottom, are the FOS B-scan percent success versus the SNR for two sources with a separation of 75, 55, 35 and 25 degrees, respectively. There are four lines for the FFT-Bscan as well, however, three of the lines are overlapping with a 0% success. The FFT-Bscan line that rises above the other three corresponds to a separation of 25 degrees.

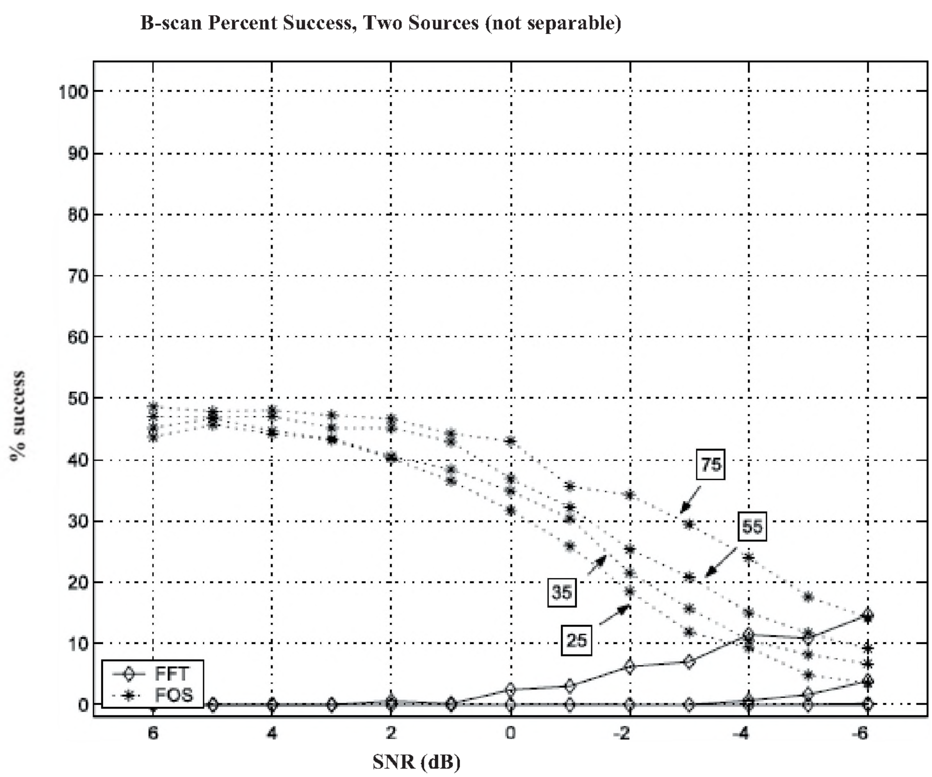

Figure 3:. FFT and FOS B-scan percent success for two non-separable sources, versus SNR (dB). The four lines, from top to bottom, are angular separations of 75, 55, 35, and 25 degrees, respectively.

Note that for the FFT B-scan estimates, the percent success increases for two sources separated by 25 degree with an SNR of 0dB or less. These "successful" estimates are due to chance. They are caused by the noise biasing the B-scan estimates of two adjacent FFT bins to be within the ±10 degree tolerance even though the signal is only present in one FFT bin.

5.2. Two Sources – Not Separable

In order to ensure consistency in the way FOS was employed, the frequency separation was decreased such that the two frequencies were in the same FOS bin. For this experiment the FOS resolution was 0.1Hz (as used in the previous experiment), while the target frequencies used were changed to 54.95Hz and 55.00Hz respectively (a separation of 0.05Hz).

As expected, neither B-scan was unable to resolve the two sources. The resulting bearing estimate for both FFT and FOS B-scan was generally a single estimate located between the two source bearings. The FFT B-scan percent success is zero at all SNR and all bearing separations. As shown in **Figure 3**, the FOS B-scan provided some successful estimates, but never more than one at a time (i.e., 50 percent success) given two input signals. Nonetheless, it clearly outperformed FFT B-scan again.

6. Experimental Results

The comparison of Fourier and FOS-based bearing estimators was extended to more complex DIFAR data. Two data sets were used to compare the two techniques. The first data file was generated in the Operational Mission Simulator (OMS) at 14 Wing Greenwood Nova Scotia, while the latter data set was recorded by a real DIFAR sonobuoy deployed in the Atlantic Ocean.

6.1. Simulated DIFAR data

The OMS is a signal generator built by CAE electronics Inc. which creates complex acoustic signatures for multiple surface and submerged contacts as well as generating an ambient noise field to simulate ocean noise. It also generates RADAR, electronic sensing measures (ESM) signals, and is a full tactical crew trainer designed for ASW.

The data set contains three targets, two of which are merchant vessels, while the third is a submerged submarine. The submarine did not produce a continuous acoustic signature and was used only to create transient signals. The two merchant contacts are located north-east and south of the sonobuoy, and their acoustic signatures consist of multiple harmonics of frequencies in the 10-400 Hz spectrum.

Figure 4 shows a plot of the relevant targets in the OMS scenario. Two merchant vessels and one submarine are present in the water. Sonobuoy positions are marked by δ with the sonobuoy radio frequency (RF) number next to them. Sonobuoys are assumed to be stationary.

The presence of these three sound sources at the same time allowed an examination of specific frequency bins which were known to contain signals incident from more than one direction. A given FFT frequency bin could therefore be subdivided to a higher frequency resolution using FOS in order to separate the multiple sources and compare the directional estimates.

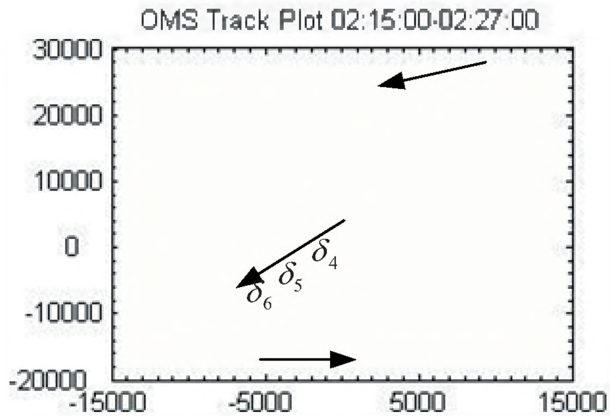

Figure 4. OMS track plot showing three targets and sonobuoys [4].

The PSD used for calculating bearing estimates was created from data that was averaged over 25 subsequent non-overlapped time slices of 5 seconds each. This results is an FFT resolution of 0.2Hz and 125 seconds of time-series data for each PSD point. The FOS resolution used in this section is 5 times the FFT resolution, or 0.04Hz. The FOS cross-spectral estimates are based upon the same length of time-series data.

From analysis not included in this paper, it was found that the signature of the vessel to the north-northeast (bearing of 020º) contains both discrete sources and broadband noise. The vessel to the south-southwest had prominent lines (bearing of 220º) at 27.4 and 42 Hz. There were clearly several instances where discrete frequency components of the two vessels overlap, (e.g. 42Hz).

Figure 5 is a comparison of a FFT and FOS generated Gram/B-scan display between the frequencies of 22 and 27 Hz as in this frequency range two of the targets have signals in the same FFT bin. The time segment chosen is known to contain discrete lines from the merchant vessel to the north, but also contains broadband noise from the vessel to the south. The data analyzed is from the eastern most sonobuoy, labelled δ_4.

The FFT gram (bottom-left) does not indicate any strong discrete spectral lines, and as such any bearings in the corresponding B-scan display would not necessarily be noted as significant by an operator. The FOS-generated gram (bottom-right) clearly shows two discrete sources present in the spectrum, and allows an assessment of their bearing in the corresponding B-scan display. Clearly identifiable in the FOS B-scan in **Figure 5** are two sources at B-scan bearings of 020º and 270º (the bearings are circled).

As FOS does not add model terms that are white Gaussian noise, the visible "salt-and-pepper" effect throughout the FOS gram indicates the presence of broadband coloured noise. This broadband noise is being fit into the FOS model.

The two targets in this scenario are at bearings of 20º and 220º respectively. Since B-scan bearings are biased by multiple sources present in a single frequency bin, the FOS B-scan bearing visible in the 22.6Hz at 270º is possibly a biased result due to the sources 220º and the broadband noise seen in the FOS B-scan.

Thus, the FOS B-scan was able to detect two bearings in the same FFT bin although the bearing for the target at 220º was biased due to the presence of broadband noise.

6.2. DIFAR Sonobuoy data

The data in this DIFAR recording are from a sonobuoy deployed from the CFAV QUEST during an acoustic trial. The set contains three significant sources: two merchant vessels as well as an acoustic projector towed by the Quest. The sound sources are well distributed around the sonobuoy, with one vessel to the north heading easterly, one to the south heading westerly, and the QUEST towing an acoustic projector heading east-south-easterly. These data were used by Desrochers [3]. A surface plot of the geographical location of known targets for the data set is shown in **Figure 6**. The arrows indicate position and approximate motion of the vessels in the area. The sonobuoy positions are indicated by the ŏ symbol. The data analyzed The DRDC data is from the eastern-most buoy just above the label 20:13 in **Figure 6**.

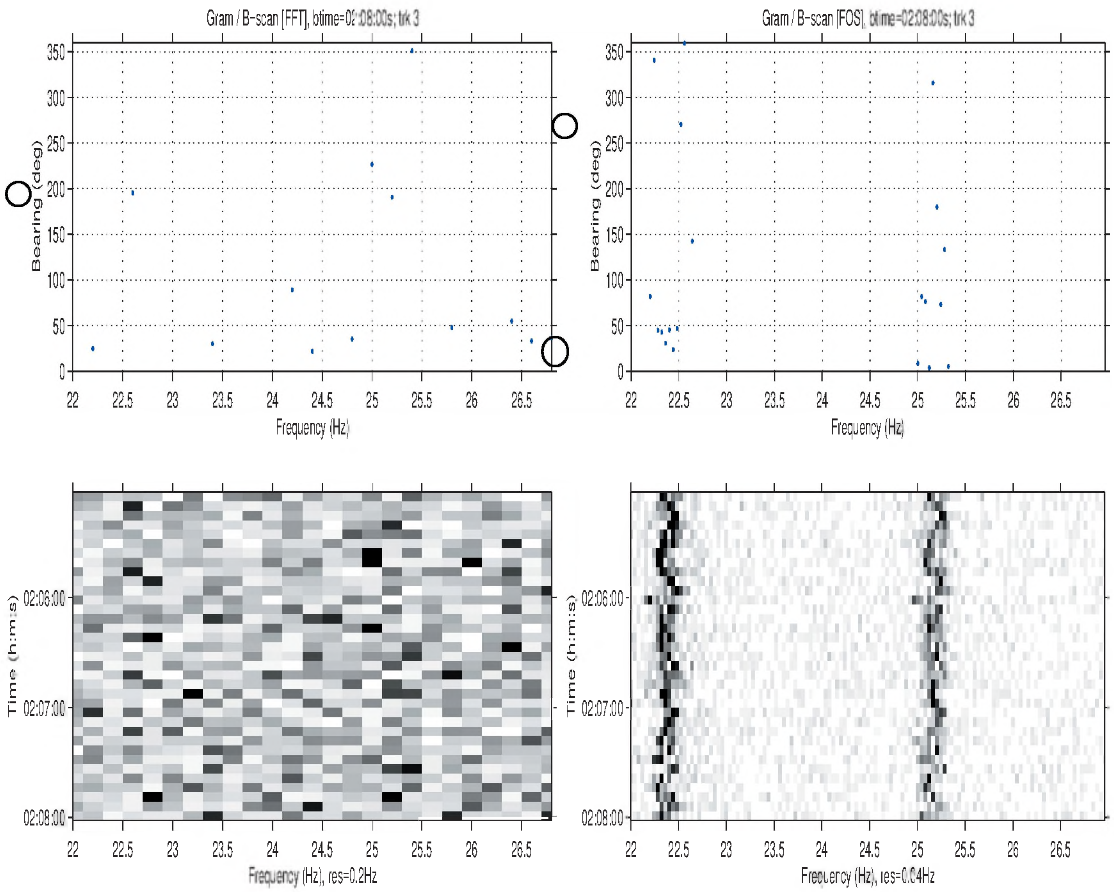

Figure 5: The B-scan (top left) and Gram (bottom left) using the FFT, and the B-scan (top right) and Gram (bottom right) using FOS for OMS DIFAR data.

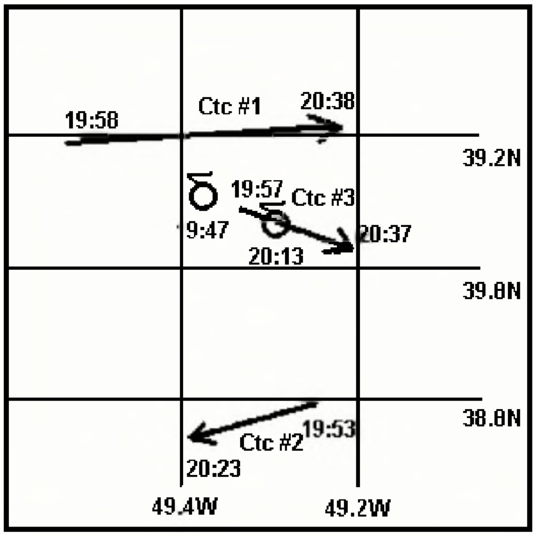

Figure 6: Surface plot of DRDC(A) DIFAR data. Extracted from Desrochers [3].

The DIFAR data set is 15 minutes long. The PSDs were estimated using a sample length of 5 seconds, resulting in a frequency resolution of 0.2 Hz. The data was averaged over 25 time slices for a total time of 125 seconds of time-series data for each row of the LOFARGRAM.

From analysis not included in this paper it was found that the towed projector generated discrete frequencies at 15, 17, 47, 49 and 147Hz at a bearing of approximately 130º. Broadband noise between 40 and 70Hz was found to be at a bearing of about 330º. Higher frequency broadband sources (above 80Hz) are found to the north-northwest, while an unknown source in the 30Hz range is at approximately 250º. A few bearings in the 80 to 110Hz range indicate the vessel known to be to the south of the sonobuoy.

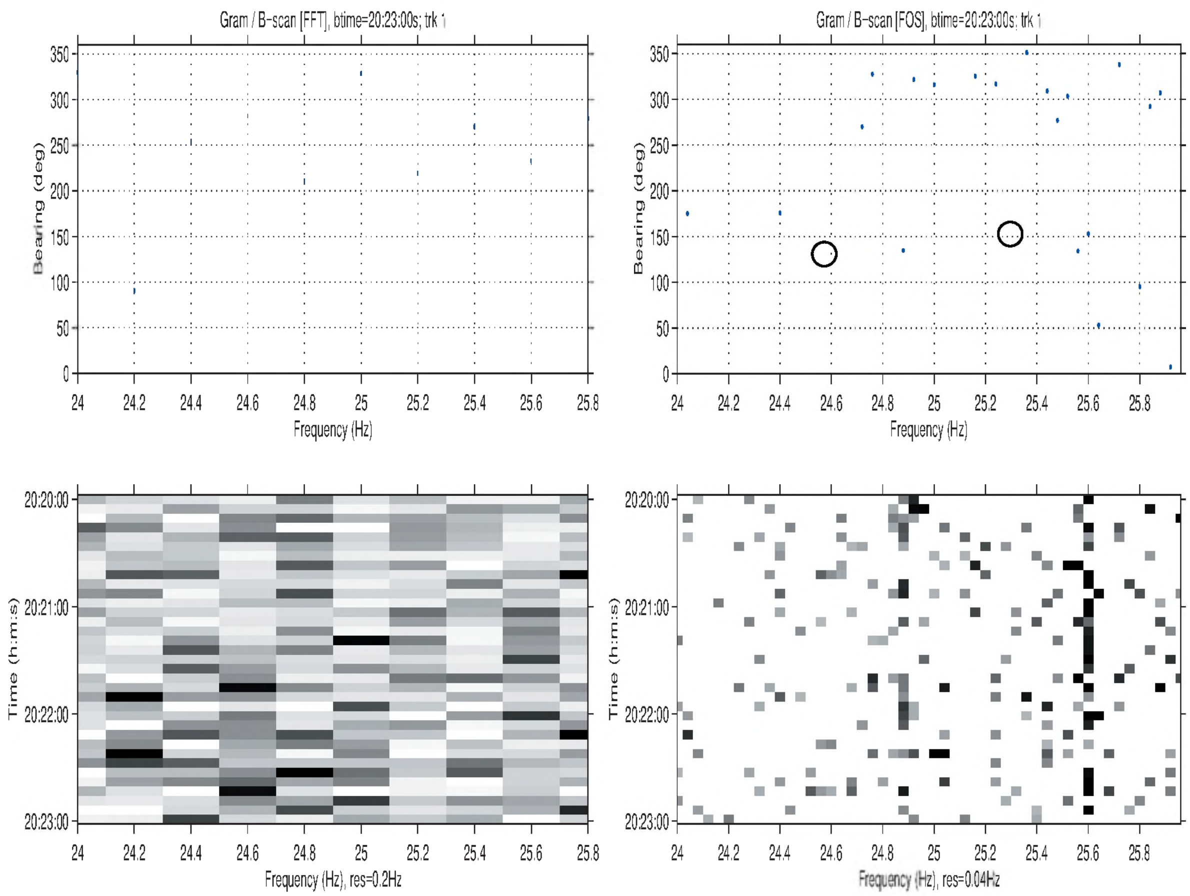

Figure 7: Comparison of FFT gram (top-left) and B-scan (bottom-left) and FOS gram (top-right) and B-scan (bottom-right) plots, for the DRDC(A) sonobuoy DIFAR data.

The frequency range between 22 and 26 Hz was examined where a known source existed but was not easily identifiable due to the presence of interference. The interference in this case was broadband noise from the merchant vessel to the north of the sonobuoy.

Twenty-five individual time slices of 5 seconds each were averaged to create the cross-spectral matrix (CSM) used for bearing calculations. The difference between the FFT-gram and FOS-gram and their corresponding B-scan estimates is seen in **Figure 7**. The FFT-gram does not show any discrete sources, while the corresponding B-scan plot shows diffuse bearings between 200º and 300º. By contrast, the FOS-gram shows the presence of two narrowband sources and identifies the corresponding bearings as 130º and 150º (circled on figure). These bearings indicate the source could be the QUEST, to the south-east of the sonobuoy. The broadband bearings are more clearly identified as being from a bearing of approximately 320º, and therefore from the merchant vessel to the north.

7. Conclusions

The FOS spectral estimation algorithm was used to perform a high resolution spectral estimate for use in the LOFARGRAM and B-scan algorithms. It was demonstrated that the FOS based algorithms could distinguish between two sources whose frequencies were in the same FFT bin, if the sources were separable by FOS. The probability of success was 0.7 or higher when the signals' SNR was 0 dB or greater. FOS B-scan was able to estimate two bearings , whereas the FFT B-scan could only estimate one bearing. Additionally, FOS provided one accurate estimate for a pair of signals in one FOS bin, where an FFT estimate yielded no accurate estimates. The improved resolution of the FOS based algorithms was demonstrated in simulation and on experimental DIFAR data.

8. Acknowledgement

The author's thank 14 Wing Greenwood Nova Scotia for the OMS simulated data. The DIFAR sonobuoy data was provided by Dr. Ian Fraser at DRDC(A) and all the data was demodulated by DRDC(A). The work was supported by an NSERC Discovery Grant and the Academic Research Program at RMC.

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A Critical Analysis of Loudness Calculation Methods

Jeff J Defoe, Colin J Novak, Helen J Ule, and Robert G Gaspar
Mechanical, Automotive and Materials Engineering, University of Windsor, 401 Sunset Ave., Windsor, Ontario

ABSTRACT

The physical meaning and methods of determining loudness were reviewed. Loudness is a psychoacoustic metric which closely corresponds to the perceived intensity of a sound stimulus. It can be determined by graphical procedures, numerical methods, or by commercial software. These methods typically require the consideration of the 1/3 octave band spectrum of the sound of interest. The sounds considered in this paper are a 1 kHz tone and pink noise. The loudness of these sounds was calculated in eight ways using different combinations of input data and calculation methods. All the methods considered are based on Zwicker loudness. It was determined that, of the combinations considered, only the commercial software dBSonic and the loudness calculation procedure detailed in DIN 45631 using 1/3 octave band levels filtered using ANSI S1.11-1986 gave the correct values of loudness for a 1 kHz tone. Comparing the results between the sources also demonstrated the difference between sound pressure level and loudness. It was apparent that the calculation and filtering methods must be considered together, as a given calculation will produce different results for different 1/3 octave band input. In the literature reviewed, no reference provided a guide to the selection of the type of filtering that should be used in conjunction with the loudness computation method.

SOMMAIRE

La signification physique et les méthodes de déterminer la sonie ont été examinées. La sonie est une mesure psychoacoustique qui correspond étroitement à l'intensité perçue d'un stimulus sonore. Elle peut être déterminée par procédure graphique, par des méthodes numériques, ou par des logiciels commerciaux. Toutes ces méthodes exigent la considération du spectre de tiers d'octave du son d'intérêt. Les sons considérés en cet article sont un ton de 1-kHz et un bruit rose. La sonie de ces sons a été calculée de huit manières, en utilisant des différentes combinaisons des données d'entrée et des méthodes de calcul. Toutes les méthodes considérées sont basé sur la sonie Zwicker. On a déterminé que seulement le logiciel commercial dBSonic et le procédé de calcul de la sonie dans le standard allemand DIN 45631 utilisant la filtration de tiers d'octaves d'ANSI S1.11-1986 donnent les valeurs correctes de la sonie basées sur les valeurs théoriques pour les tons de 1-kHz. Comparant les résultats pour les différents bruits démontrent également la différence entre le niveau de pression acoustique et la sonie d'un bruit. Il est évident que la méthode de calcul ne puisse pas être séparée de la méthode de filtrage, car un calcul donné produira différents résultats pour des tiers d'octaves différents. Aucune mention n'est faite dans la littérature examinée de quel type de filtrage devrait être employé pour les calculs de la sonie.

1. INTRODUCTION

As the field of acoustics moves closer to the forefront of engineering, particularly in product development, it is becoming increasingly apparent that the sound pressure level (SPL) of a noise source is not the only important metric. Of equal, if not more importance, is the sound quality, or psychoacoustic characteristics of a noise source. The study of psychoacoustics involves the quantification of the human perception of sound. Psychoacoustics began in earnest with the study of correlations between acoustic stimuli and the sense of hearing in the 1930s [1], and had attracted serious attention by the 1950s [2].

One of the most commonly used psychoacoustic metrics is loudness. This metric aims to quantify how loud a sound is perceived to be in comparison to a standard sound [2]. It accounts for both frequency-sensitivity and masking effects. The loudness of a sound is most commonly computed from whole or 1/3 octave band sound pressure levels measurements of the sound source of interest. However, the calculation procedure is non-trivial and poorly understood despite having been standardized in ISO 532 (1975) [3], DIN 45631 (1991) [4], ANSI S3.4-1980 [5] (outdated), and ANSI S3.4-2005 [6]. ISO 532A defines a procedure for determining loudness based on octave band measurements of a sound source, from Stevens' method [7]; ANSI S3.4-1980 is also based on Stevens' work. ISO 532B and DIN 45631 require 1/3 octave band inputs and are based on Zwicker's method [2, 8]. The updated ANSI S3.4-2005 is based on Moore's method [9] and is not considered in this paper; nor are the standards based on Stevens' work. Only Zwicker methods are considered. There are commercial software packages that calculate loudness, some according to ISO 532B and some using non-standard methods. In this paper, the 01dB software packages dBFA and dBSonic were considered. For this investigation, two noise sources are used in comparing the software packages to a public domain code and two codes written by the authors – one based on ISO 532B, the other on DIN 45631. A 1 kHz tone and pink noise were used. The purpose of this comparison is to evaluate the validity of the results obtained from the various methods, as well as to gain insight into the shortcomings of the relevant standards. In addition, comparisons of results amongst the different sound types will serve to illustrate the differences between the well-accepted metric of sound pressure level and loudness.

Before a complete definition of loudness can be presented, further details need be provided on the concept of masking and on the frequency characteristics of the human hearing system. In general, a sound will prevent other sounds of lower sound pressure level but with similar frequency content from being heard. Another type of masking can occur when one sound follows very closely after another in time. The second sound may at times not be heard. This is known as temporal masking. Typically, sound level meters and frequency analyzers will present the frequency content of a measured signal in terms of fractional octave bands. Whole, 1/3, 1/12 and 1/48 octave band filters are commonly encountered. Unfortunately, the human hearing system does not use fractional octave band filtering. The major range of human hearing (20 Hz to 16 kHz) is more properly filtered into bands based on the frequency ranges in which masking will occur – that is, if two sounds occur with frequency content within one band of each other, masking will take place [2]. These bands can be related to positions along the basilar membrane in the cochlea [10]. This band system was first proposed by Fletcher [1], and later refined by Zwicker [11], terming the divisions "critical bands" and assigned to them the units of Bark, in honor of Barkhausen (inventor of the loudness level unit "phon"). The critical bands are not of uniform width. For frequencies below 500 Hz, the bands are approximately 100 Hz wide [2, 11]. This includes the approximation that the first band begins at 0 Hz instead of the human hearing threshold of 20 Hz. Above 500 Hz, the bandwidth is approximately 20% of the centre frequency. The conversions from the critical band rate (z) to frequency (f) can be found in references [2] and [11]; analytical approximations to the critical bands exist and are also given in references [2] and [12]. Moore's work [9] re-examined the concept of the critical band and came up with a set of ERB (equivalent rectangular bandwidth) filters. These filters are similar to critical band filters at frequencies above 500 Hz, but below this they differ in that the ERB filters continue to narrow with decreasing centre frequency. Comparisons of results obtained using the Moore and Zwicker models show increased accuracy for sounds with significant low-frequency content for Moore's model [9]. Therefore, the calculation methods considered in this paper should not be used for sounds with dominant low-frequency content.

Excitation is the representation of the effect of a sound occurring within a specific critical band [2] – some excitation occurs outside of the critical band in which the sound occurs. So, "similar" frequency content means that one sound's critical-band spectrum is (at least partially) overshadowed by the masking sound's critical-band spectrum. This is illustrated schematically in figure 1. Loudness is "the sensation that corresponds most closely to the sound intensity of the stimulus" [2, p. 205]. The loudness of a 1 kHz tone at an SPL of 40 dB is 1 sone. This sound is termed the reference sound. All other loudness values are in comparison to the reference. The loudness of a sound is dependant on the sound's frequency content and bandwidth. Thus, a weighting scheme such as the A-weighting scale is too simplistic to accurately determine the loudness of a sound [2, 13]. In fact, SPL as reported in dB or dBA can actually be misleading regarding the perceived loudness of a sound, as will be shown later. Because of the sloped tails on the critical-band filters in the human ear, as shown from the excitation pattern in figure 1, the proximity (in terms of frequency) of two sounds affects the total perceived loudness. In essence, unless two sounds are distantly separated in frequency, the total loudness will not be the sum of the sounds' individual loudnesses [8]. The concept of "specific loudness" is employed to mean the contribution to the total loudness of a specific slice of the critical band spectrum. Mathematically speaking,

$$N = \int_0^{24Bark} N' \cdot dz \tag{1}$$

where N is the total loudness and N' is the specific loudness in critical bandwidth dz.

Critical-band filtering is not widely available in most software, and so a procedure was developed for use with 1/3 octave band data [2, 8]. The true loudness of a sound accounts for both frequency and time masking; however, since octave band filtering requires the sound to be recorded over a finite time span, the procedure assumes a steady-state sound and thus accounts only for frequency masking. Only stationary signals are considered in this paper.

2. CALCULATION PROCEDURES

Both the graphical and numerical calculation procedures for loudness are described below.

2.1 Calculation Procedure - Graphical

Determining the actual loudness of a sound involves a number of steps: conversion of the data from 1/3 octave bands to approximated critical bands, determination of specific loudness, and then the summation to get the total loudness.
The procedure is a graphical one, standardized in ISO 532 B [3]. It is also included in DIN 45631 [4], which additionally

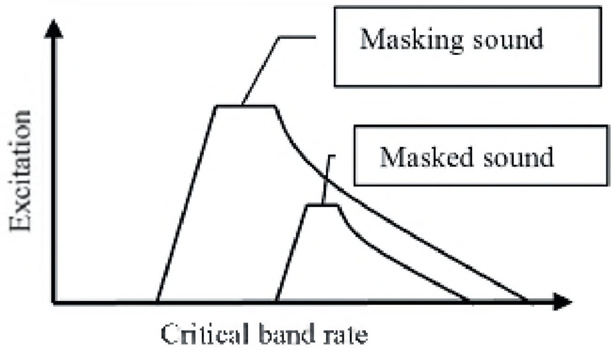

Figure 1. Schematic of masking from excitation patterns.

contains the code for a computer program to calculate loudness based on its 1/3 octave data. A simplification that is included in the procedure is the approximation of the lower slopes of excitation patterns as being infinitely steep, which is justified since they are much steeper than the upper slopes (see Figure 1) [8]. The procedure can account for sounds measured in both free and diffuse fields [3, 7].

The first step, converting 1/3 octave data to critical band data, requires that all bands up to 90 Hz be combined into the first critical band, that the three bands from 90 to 180 Hz be combined into the second critical band, and that the two bands from 180 to 280 Hz be combined into the third critical band. Above 280 Hz, the 1/3 octave bands approximately correspond to critical bands [2, 3, 7]. The levels in each critical band are then plotted on graphs (provided as part of the standard) as horizontal lines within the appropriate band at the given level [3].

The second major step, determining the specific loudness across the critical band rate scale, involves joining the horizontal lines drawn in step one in a specific way. Moving from the low to high frequency, the following rule is applied [3]: if the level in a band is higher than that of the previous band, the horizontal segments are to be joined by a vertical line. This is the infinitely-steep lower slope approximation mentioned earlier. If the level in a band is lower than that of the previous band, the reduction is not immediate but follows a curve of the type shown in Figure 2 until it intersects another of the drawn horizontal lines. This gives the Ns = f(z) curve, the area under which is the total loudness [3].

The third step, summation or integration, is simply the determination of this area by any appropriate graphical means.

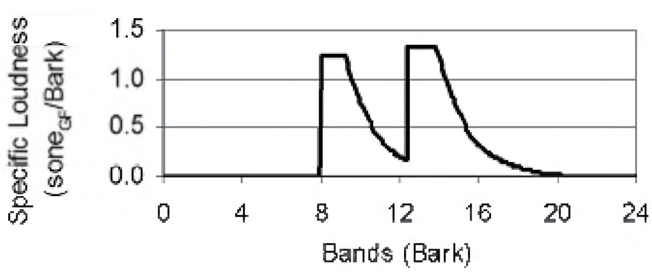

Figure 3. Sample graph (specific loudness curve).

It is to be noted that the loudness obtained is to be given in units of soneG as opposed to sone, where the G indicates that the loudness was determined graphically rather than by jury testing [2].

2.2. Calculation Procedure - Numerical

The graphical procedure described above is somewhat tedious to use in practice, and has some inherent loss in accuracy due to the interpolations required by the user. Two computer programs were written to automate this procedure. One was written by Paulus and Zwicker [8] for FORTRAN and later updated in reference [14] to run in BASIC. The other was developed as part of DIN 45631 (for BASIC) and was published in a slightly different programming format in reference [15].

2.2.1 Method 1: from references [8] and [14]

In the development of the program found in reference [8], the authors strive for improved accuracy compared to the graphical procedure by basing the program not purely on the graphs in ISO 532 B, but rather on an understanding of the underlying phenomena. As in the graphical procedure, the overall idea is to convert 1/3 octave band levels into a specific loudness curve and then integrate this curve from 0 to 24 Bark to get the total loudness. The equations and values used are empirically derived [8] and this requires the use of lookup tables for data rather than the use of functions. The calculation process is outlined below.

Consider first a simple case. For a sound with a main excitation lying within one critical band, the specific loudness is determined from [8]

$$N' = 0.064 \cdot 10^{0.025 L_{EHS}} \cdot \left\{ \left[1 + \frac{1}{4} 10^{0.1(L_E - L_{EHS})} \right]^{0.25} - 1 \right\} \qquad (2)$$

Broadband sounds are treated as a type of summation of several, critical-band wide, sounds [8]. However, at each critical band rate value z, only the largest specific loudness is retained. The others are discarded. This provides the masking characteristic observed in the human ear.

The sloped curves used for determining the decreasing upper tails of specific loudness in reference [4] do not have a mathematical representation, a fact which is very relevant for programming a loudness calculator [8]. Lookup tables must be used to determine the proper value of the slope at a given critical band rate and level. Actually, the slope is frequency-dependent below about 900 Hz [8]; above this frequency it is a function of level only. To this point, it was assumed that the available input data was in the form of critical band levels. Since this is quite uncommon and also in order to more closely mirror ISO 532 B, modifications are included to permit the use of 1/3 octave band levels instead.

The first step is to combine the low-frequency 1/3 octave bands as in the graphical procedure. However, an additional factor is considered: the threshold of hearing is taken into account for the lowest frequencies (up to 63 Hz) and if the 1/3 octave level is below the threshold, no contribution to loudness occurs [8]. After the first three critical bands are approximated in this way, subsequent critical bands are approximated by full 1/3 octave bands as in the graphical procedure. This approximation introduces an error into the loudness calculation which is neglected in the graphical procedure due to its own inherent inaccuracies. For the numerical method, an additional correction is introduced based on the ratio of the approximated and exact critical bands [8].

It was noted by Paulus and Zwicker [8] that the loudness level determined in the graphical procedure will differ from the calculated one if the loudness is less than 0.5 soneG. This is due to additional nonlinearities in the human hearing system that the equation-based method does not inherently account for.

Because of the fact that lookup tables are used to determine the upper slopes of specific loudness, the slopes are constant over each segment and thus an accurate integration may be done using a simple trapezoid rule method [8]. Finally the loudness level can be calculated from

$$L_N = 10\log_2 N + 40 \tag{3}$$

The results obtained from the program agree with those from the ISO graphical procedure within the repeatable accuracy of the graphical method for all sounds with loudness above 0.5 soneG. Below this loudness, additional calculations to account for the nonlinear scale in the graphical procedure below 0.5 soneG would need to be included. This was addressed by Zwicker et al. [15]. Reference [14] contains an identical program written in BASIC instead of FORTRAN. Both programs are able to account for the differences in loudness that result from the sound field being diffuse as opposed to free [7, 15].

2.2.2 Method 2: from reference [15]

While the programs in references [8] and [14] aimed to produce results in close agreement with ISO 532 B, a more recent standard, DIN 45631, exists for calculating loudness. This standard contains a BASIC program for calculating loudness that is different than the program developed by Zwicker et al. [14]. A different program, also by Zwicker et al. [15], which produces the same results as the original DIN standard's program was published in 1991 and claims to calculate loudness values that are "in line" with ISO 532 B. In reference [15], much of the lookup table data is slightly different than in reference [14], the most significant of these differences being that the lookup tables for the upper slopes for accessory loudness contain 18 division levels instead of 16 as they do in reference [8]. Also, the program in reference [15] contains a modified loudness level formula for cases when the loudness is less than 1.0 soneG. This low-loudness formula is

$$L_N = 40(N + 0.0005)^{0.35} \tag{4}$$

with a further correction that if the loudness level so calculated is less than 3 phonGF, then the loudness level in set equal to 3 phonGF. Other than these differences, the programs are similar.

3. SOUNDS TESTED, EXPERIMENTAL SETUPS AND PROCESSING SYSTEMS USED

A comparison of the results for loudness computed from various methods was investigated in this study. To do so, two sounds were used as inputs. Both are described along with the measurement method used below. The data and calculation methods used are also described.

3.1 Sounds Tested and Experimental Setups

Both a 1 kHz tone and pink noise were tested. These are commonly used sounds in psychoacoustic evaluations.

3.1.1 1 kHz Tone

Given that the reference sound for loudness is a 40 dB SPL, 1 kHz tone, this sound was included in the test to see how close the various processing systems came to achieving the theoretical value of 1 sone. In addition, a 1 kHz tone at 80 dB SPL was also tested in order to compare with the pink noise tested. Both tones were generated via a signal generator and acquired at calibrated levels using a data acquisition system.

3.1.2. Pink Noise

Pink noise was played through a Peavey PR10 speaker located on a table 0.73 m off the floor. A microphone was placed directly in line with the speaker at a distance of 2.0 m and at a height of 1.03 m, corresponding to the vertical centre of the speaker. Data was recorded using both a Larson-Davis System 824 sound level meter (SLM) and a multi channel 01dB Orchestra acquisition system. Two 30 second samples were recorded. The overall A-weighted SPL during the tests was 80 dBA.

3.2. Loudness Calculation Processing Systems

A total of eight combinations of 1/3 octave input data and processing methods (See Table1 for details) were used for the pink noise, and six combinations were used for the pure tones in order to be able to properly compare all available methods.

3.2.1. Input Data

Two basic sources of input data were available. The SLM

Combination	Input Data	Processing Method
1	SLM 1/3 oct.	VB from [4, 5] (ISO)
2	dbFA 1/3 oct. (IEC)	VB from [4, 5] (ISO)
3	SLM 1/3 oct.	VB from [7] (DIN)
4	dbFA 1/3 oct. (IEC)	VB from [7] (DIN)
5	dBFA 1/3 oct. (IEC)	dBFA
6	dBSonic 1/3 oct.	dBSonic
7	MATLAB 1/3 oct. (ANSI)	MATLAB (DIN)
8	dbFA 1/3 oct. (IEC)	MATLAB (DIN)

TABLE 1. Combinations considered.

directly provided 1/3 octave band levels of the measured sources, while the 01 dB system using the Orchestra and computer records raw data for later analysis. This file can be processed to give 1/3 octave band levels. This filtering process was accomplished in several ways. The 01dB software, dBFA [16], filters the data according to IEC 61260 [17]. The 01dB software dBSonic uses an unknown internal filtering algorithm. The MATLAB program [18], written by Hastings, filters according to ANSI S1.11-1986 [19]. An update to this standard was published in 2004 [20] which is essentially identical to the IEC standard.

3.2.2. Processing Methods

Five processing methods were used for this study. These were:

- a Microsoft Excel Visual Basic program adapted from reference [14]
- a Microsoft Excel Visual Basic program adapted from reference [15]
- the 01dB software dBFA
- the 01dB software dBSonic
- a MATLAB program [18] based on reference [15]

The two Visual Basic (VB) programs were written by the present authors. dBFA, dBSonic and the MATLAB program process the signal in a two-step process. First, the Orchestra data is converted to 1/3 octave data. Then, this data is fed into the loudness calculator. With dBFA and dBSonic, both of these steps occur within the program with the 1/3 octave data also available for output. With the MATLAB code, the Orchestra to 1/3 octave data conversion can be skipped and the 1/3 octave band levels can be supplied directly.

3.2.3. Combinations used

In order to compare the processing methods and input data, eight combinations were developed. Table 1 lists the combinations in terms of which 1/3 octave band levels and processing system was used for each. These combinations prove sufficient to allow a detailed analysis of the results obtained. For the 1 kHz tone, combinations 1 and 3 were not considered for the reasons outlined above.

4. RESULTS AND DISCUSSION

This section presents the results, analysis and discussion of the tests.

4.1. Results

The loudness results for both types of sounds are shown in Figures 3 and 4. Note that the loudness metric is soneGF, where the G indicates that the loudness was determined from 1/3 octave bands and the F indicates that the sound field was in the free-field condition. Also, when examining the figures, the legend information is to be read from left to right sequentially to correspond to the amplitude bars.

For the 40 dB tones, the Excel (ISO) method using IEC 1/3 octave data from dBFA, dBSonic, and the Hastings (DIN) method using ANSI 1/3 octave data yield the correct result of exactly 1 soneGF. Hastings (DIN) with IEC filtering gives only 0.93 soneGF, which is identical to the Excel (DIN) with IEC filtering result. dBFA over-predicts the loudness to be 1.05 soneGF. For the 80 dB tones, no combination gives exactly 16 sone, which is the theoretical loudness for this sound. However, dBFA, dBSonic and Hastings with ANSI filtering give results closest to the theoretical value: 16.7, 16.5 and 16.4 soneGF respectively. All other methods significantly under-predict the loudness: Excel or Hastings (DIN) with IEC filtered data give 14.3 soneGF, and Excel (ISO) with IEC filtering gives 14.8 soneGF. Thus for the two sets of tones – the simplest sounds for which one can compute the loudness – only dBSonic and DIN with ANSI filtering give consistently accurate loudness values.

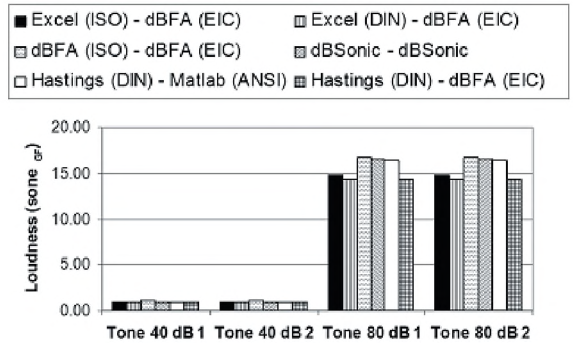

Figure 3. Loudness results for tone measurements.

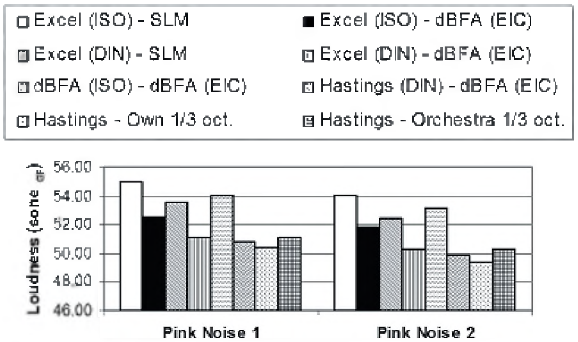

Figure 4. Loudness results for pink noise measurements.

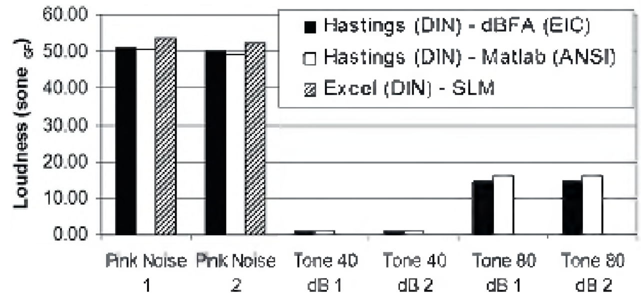

Figure 5. Loudness calculated using DIN with IEC, ANSI and SLM 1/3 octave band filtering.

In addition to the methods used for the tones, the pink noise was also analysed using the sound level meter (SLM) which contains its own filtering method. These results were processed through ISO and DIN loudness routines. Based on the results from the tones, dBSonic and DIN with ANSI filtering will be taken as "correct" or reference values of loudness, against which the other methods will be compared. The two agree well for the pink noise measurements, deviating by at most 0.5 soneGF. The numerical values are presented in Table 2. For the pink noise, DIN with IEC filtering yields results that are quite close to the reference values – 51.1 soneGF and 50.3 soneGF for sample 1 and 2, respectively. dBFA over-predicts for both samples by about 3 soneGF. ISO with IEC filtering and both ISO and DIN with the SLM filtering all over-predict the results as well.

In the next section, these results will be analyzed in-depth.

4.2 Analysis and Discussion

Several aspects of the results raise the need for further analysis or detailed discussion. These include the dependence of the results on the 1/3 octave filtering method used, the sources of error in the "incorrect" methods, the differences between loudness and SPL, the differences in loudness between ISO and DIN, and the practical precision limits on loudness.

4.2.1. Filtering Dependency

By looking at the results of the DIN loudness calculations using 1/3 octave data filtered via IEC 61260, ANSI S1.11, or using the SLM, it becomes apparent that there is more than just the loudness calculation routine that affects the final value obtained. From Figure 5 and Table 3 the differences in the results are clear.

Sample:	1	2
dBSonic	50.8	49.9
DIN + ANSI	50.4	49.4

TABLE 2. Pink noise results for dBSonic and DIN with ANSI (soneGF).

Sound	IEC	SLM	Maximum
Pink Noise 1	1.37%	6.29%	6.29%
Pink Noise 2	1.84%	6.26%	6.26%
Tone 40 dB 1	-7.30%	N/A	-7.30%
Tone 40 dB 2	-7.30%	N/A	-7.30%
Tone 80 dB 1	-12.86%	N/A	-12.86%
Tone 80 dB 2	-12.86%	N/A	-12.86%

TABLE 3. Error between ANSI, IEC and SLM results (reference: ANSI) for DIN loudness

Between IEC and ANSI, the differences are small but significant (> 1%) for all sounds but the large deviations for the tones should be noted. This is not what would be expected, since if any sound could be expected to be accurately predicted by any loudness calculation, it would be the 1 kHz tone which is the reference sound for loudness. The differences between the SLM and ANSI data are even more significant. The implication is that differences in the frequency spectra of the sounds are responsible for the differences in the errors obtained. The filtering method plays a significant role in defining the accuracy of the loudness calculation. Thus, the same loudness calculation procedure will yield different results when a different method of 1/3 octave band filtering is used. While this result may seem obvious when stated this way, it is often overlooked. There is no mention of the method whereby the 1/3 octave band levels used in the computation should be acquired in ISO 532 B [3], nor in the numerical methods described in references [8], [14] and [15]. Recall that reference [15] contains an identical program to DIN 45631 [4]. This is a significant oversight in the specification of these standards and does little to instill confidence in the results obtained via their use.

4.2.2. Sources of Error

The large deviations of the majority of the loudness calculation methods from the two determined to be "correct"

– dBSonic and DIN with ANSI filtering – require some investigation. For the pink noise, the "incorrect" methods all over-predict the loudness by varying amounts. It is interesting that this over-prediction is not unilateral. The loudness values for the tones are under-predicted by the same methods, except dBFA which always over-predicts the loudness. Sources of error include the differences in filtering methods (as discussed above), as well as differences in lookup table data between the programs given in references [8] and [15].

4.2.3. Loudness vs. Sound Pressure Level

In Section 3 it was stated that all the sounds gave an A-weighted SPL of 80 dBA. The "A" scale was chosen as it is the most commonly used weighting used in acoustics when attempting to deliver a better impression of human perception than a linear SPL [21]. Figure 6 and Table 4 show the loudness values obtained for the different sounds. In the table, the values for mean loudness were calculated by taking the average of the loudness for the samples and across both "correct" methods: dBSonic and DIN with ANSI filtering. The loudness values for these sounds vary greatly: the pink noise has a loudness about three times higher than the 1 kHz tone. Thus, it is evident that overall SPL does not illustrate the entire story. This result should not be surprising given that the bandwidth of the pink noise is large compared to that for the 1 kHz tone. Figure 7 shows the specific loudness vs. critical band rate curves for the tone and pink noise, respectively. The area under the pink noise curve is much larger than the area under the tone's curve, leading to the greater loudness value.

4.2.4. ISO vs. DIN

It is interesting to compare results for ISO and DIN given the same 1/3 octave band inputs. Even though the ISO with SLM and IEC filtering was shown to give incorrect loudness values as compared to the reference methods, this comparison is still useful since ISO is after all an international standard in wide use. Tables 5 and 6 provide these comparisons for SLM and IEC filtering, respectively. The error varies from 2.63% to 7.95%. The statement in reference [15] that DIN 45631 gives results "largely identical" to ISO 532B is somewhat of an exaggeration. Also, it can be noted that ISO always produces a higher loudness than DIN for the same input data.

4.2.5. Precision and Accuracy Limits of Loudness

As a final observation, given the empirical nature of the data used in the calculation of loudness and the dependence of the final values obtained on calculation and filtering method(s), it may not be meaningful to report loudness values with too great of a precision. Single decimal place precision is the best that one should be expected to report and expect to be significant.

In practice, the best accuracy that should be expected is about

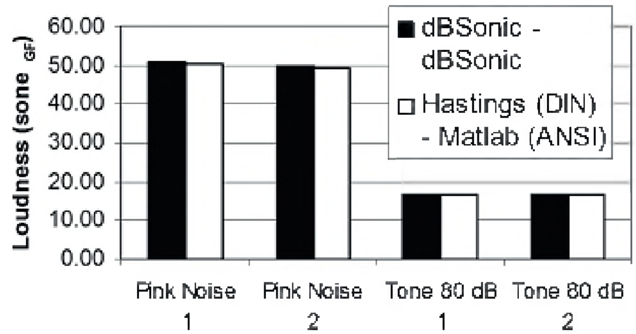

Figure 6. Loudness comparisons.

Sound	Pink Noise	80 dB, 1 kHz tone
Loudness (sone$_{GF}$)	50.1	16.5

TABLE 4. Comparison of mean loudness for the sounds tested

±0.5 sone, though often the margin of error will be even larger. Recall that loudness is a measure of the human perception of the intensity of a sound and that commonly, it is said that a 3 dB change in SPL is the minimum perceptible by the human hearing system. Also, for mid-frequency tones above 40 phons, a 10 dB-increase in the SPL corresponds approximately to a doubling of its loudness [2]. Consider the 80 dB, 1 kHz tones presented. With an SPL of 80 dB, the loudness is about 16.5 soneGF. According to the power law described in [2], a just-perceptible change to 83 dB would result in a new loudness of 19.7 soneGF. This is a change of 3.7 sone! So, here the practical accuracy limit would be a range of this magnitude, that is, ±1.8 sone. There is clearly a significant error margin in all loudness computations.

While the same doubling formula cannot really be applied to sounds other than tones, it can be assumed to give a rough estimate and so the ±1.8 sone accuracy can still be assumed as a first approximation. Even given this relatively generous margin of error, the deviations amongst the results are

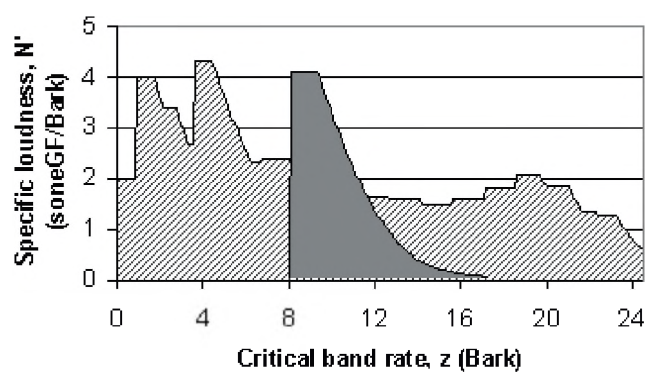

Figure 7. Specific loudness vs. critical band rate for the 80 dB, 1 kHz tone (filled) and pink noise (crosshatched).

	ISO (sone$_{GF}$)	DIN (sone$_{GF}$)	% error (reference: DIN)
Pink Noise 1	54.9	53.5	2.63%
Pink Noise 2	54.0	52.5	2.99%

TABLE 5. ISO and DIN loudness for SLM data.

	ISO (sone$_{GF}$)	DIN (sone$_{GF}$)	% error (reference: DIN)
Pink Noise 1	52.6	51.1	2.96%
Pink Noise 2	51.8	50.3	3.09%
Tone 40 dB 1	1.0	0.9	7.95%
Tone 40 dB 2	1.0	0.9	7.95%
Tone 80 dB 1	14.8	14.3	3.84%
Tone 80 dB 2	14.8	14.3	3.84%

TABLE 6. ISO and DIN loudness for IEC data.

Sound	Maximum deviation (sone)
Pink Noise 1	4.6
Pink Noise 2	4.7
Tone 80 dB 1	2.4
Tone 80 dB 2	2.4

TABLE 7. Maximum deviations from correct values for 80 dBA sounds,

still significant as can be seen in Table 7. The maximum deviation was calculated by taking the difference between the highest and lowest loudness values for a sound. Although these deviations are outside of what should be considered as a reasonable margin of error, they do not seem quite so large when viewed in this light.

5. CONCLUSIONS

The physical meaning of and calculation procedures for determining loudness were reviewed. 1 kHz tones at 40 and 80 dB(A), and pink noise at 80 dBA were used to compare several combinations of loudness calculation methods and 1/3 octave band filtering techniques. It was determined that the only two combinations to give accurate results were the dBSonic (using internal filtering) and DIN with (old) ANSI filtering methods. The program based on ISO does not yield an accurate measurement in most cases.

The calculation of loudness from 1/3 octave band levels cannot be separated from the 1/3 octave band filtering process, as different methods of filtering (SLM, IEC, ANSI) all result in different loudness values even when processed using a single calculation method. This dependence is ignored in the literature reviewed.

The results also highlight the difference between SPL and loudness. While all the sounds tested had an SPL of 80 dBA, their loudness varied from 16.5 to 50.1 soneGF, as would be expected due to differences in frequency content. Finally, one should be cautioned not to state values with too great an accuracy when dealing with loudness. Generally the error will be in the range of ±0.5 to 2.0 sone.

Topics for further study include an in-depth analysis of the 1/3 octave spectra as well as the specific loudness curves of the sounds for the different filtering methods. The goal would be to pinpoint the contributions that lead to different loudness values, as well as the consideration of other combinations such as ISO loudness with ANSI filtering.

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Use Of Portable Audio Devices By University Students

Shazia Ahmed, Sina Fallah, Brenda Garrido, Andrew Gross, Matthew King, Timothy Morrish, Desiree Pereira, Shaun Sharma, Ewelina Zaszewska, and Kathy Pichora-Fuller
University of Toronto at Mississauga, Department of Psychology, 3359 Mississauga Rd N, Mississauga, Ontario

ABSTRACT

New digital portable audio devices such as the Apple iPod have caused renewed concerns that recreational noise exposure may pose a danger to the hearing health of young adults. In this study, 150 undergraduates completed a survey about their use of portable audio devices and about other factors that could affect their hearing health. In addition to completing the survey, 24 students also participated in an experimental session. In the experimental session, hearing thresholds up to 14 kHz were measured and objective acoustical measures of output of the iPod were obtained. Participants listened to music and adjusted an iPod to their preferred setting in five conditions: in quiet and in two types of background noise, traffic or multi-talker babble background, at a high and a low level. A Brüel and Kjær dummy head and PULSE sound analysis system were used to measure the output of the iPod at the preferred settings of the students and at predetermined volume and equalizer control settings. It was found that most students use portable audio devices, but the pattern of their usage seems to be potentially hazardous only for a minority. The importance of education about safe usage of this technology is emphasized

SOMMAIRE

De nouveaux appareils audio digitaux, tel le Ipod, ont renouveler les inquiétudes que peut amener le bruit récréatif à l'ouïe de jeunes adultes. Dans la présente étude, 150 étudiants au baccalauréat ont complété un questionnaire concernant leur utilisation d'appareils audio portatifs ainsi que d'autres facteurs pouvant affecter leur santé auditive. En plus de répondre au questionnaire, les 24 étudiants ont également participé à une session expérimentale. Lors de cette session, les seuils auditifs atteignant 14 kHz ont été mesurés et des mesures acoustiques objectives du iPod ont été obtenues. Les participants ont écouté de la musique et ont ajusté leur iPod au niveau qu'ils préféraient dans cinq conditions. L'une de ces conditions était silencieuse et les deux autres avaient un bruit de fond (du traffic ou plusieurs personnes qui parlaient) à des niveaux de sons haut et bas. Une tête de mannequin Bruel et Kjaer ainsi qu'un système d'analyse de son PULSE ont été utilisés afin de mesurer le output du iPod aux réglages de son favorisés par les participants et des niveaux de volume egaux. Il a été trouvé que la plupart des étudiants utilisent des appareils audio portatifs, mais le patron d'utilisation de ces devis ne pose un danger que chez une minorité d'entre eux. L'article met une emphase sur l'importance d'offrir une éducation sur l'usage sûr de la technologie.

1. INTRODUCTION

For decades, noise has been recognized as a hazard that can damage hearing (Clarke & Bohne, 1999). Concerns about industrial and military noise have dominated research and practice regarding the prevention of noise-induced hearing loss. Nevertheless, it is widely held that exposure to noise in recreational activities could affect hearing health (e.g., Chung et al. 2005; Health & Welfare Canada, 1988; Williams, 2005), with youth being vulnerable (e.g., Ciona & Cheesman, 2000; Lees, Roberts, & Wald, 1985).

In modern everyday life, people are continuously bombarded with noise that is potentially detrimental to hearing health. One of the sources of recreational noise that has received media attention is portable audio devices (e.g., Fearn & Hanson, 1989). These devices have been popular for decades, especially among adolescents and young adults. The media drew attention to the potential risks of using the Sony Walkman in the 1980's. In the 1990's, studies investigated headphone/portable CD players (Bly, Keith & Hussey, 1998; 1999, 2001). Very recently somewhat similar coverage has been given to the potential risk of using the Apple iPod (Fligor, 2006; Hawaleshka, 2006; Spencer, 2006). Some fear that the use of digital devices is excessive and more dangerous than older portable audio technology. The purpose of our study was to investigate if this new technology poses a significant risk to hearing health.

Given the newness of digital portable audio technology, little research has yet been conducted to investigate how it is actually used by young adults. A recent study for the American Speech and Hearing Association gauged the potential risk of portable audio devices based on data gathered in a telephone-based survey focused on high school students (Zogby, 2006). Compared to adults, high school students listened to their audio devices for longer periods at higher settings compared to adults, and they reported having more symptoms of hearing loss, including turning up the TV, tinnitus, and difficulty communicating. As in previous studies, factors that resurfaced were the low level of awareness of the risk of loud sound to hearing health, and the low level of worry about hearing health, at least for those not educated about hearing loss prevention (e.g., Williams, 2005; Chung, 2005; Zogby 2006).

The specific aim of the study was to examine the relationship between the use of portable audio devices and hearing health in university students using both subjective and objective measures. Subjective measures were gathered by administering a web-based questionnaire to determine how portable audio devices are used by university students, how their use interacts with other sources of noise exposure, and whether their patterns of use raise concerns about hearing health. Objective measures included audiometric testing of hearing and acoustical measurement of the output produced by an iPod under control conditions and at the preferred settings of users when listening to two types of music in quiet and in different noisy background conditions.

2. METHOD

2.1 Participants

A questionnaire was administered to 150 participants. The participants included 126 undergraduate students who received a credit towards their Psychology 100 course at the University of Toronto at Mississauga (UTM) for their participation. The others were undergraduate students who volunteered to complete the study with no monetary compensation. Participants from PSY 100 were recruited using the course website and they were tested in groups in a computer lab at the university that was reserved for the study. The others, students involved in other projects in the same research facility, completed the survey individually on a computer in the lab. All participants provided informed consent. The survey took less than 30 minutes to complete and was followed by an information session about the effects of noise on hearing and how to conserve hearing. All participants were young adults, most being between the ages of 16 and 20 years (71.3%) or between 21 and 25 years (28%). Just over half (56%) were male. Almost all were single (92%) and still lived at home with family (82%).

Twenty four of the students who had completed the questionnaire volunteered to attend a second one-hour session at which the objective measures were collected. The measures included audiometry and acoustical measurements of output at the preferred iPod settings for listening to music heard in quiet and in two levels of background noise. Most students earned one course credit for their participation in the second session and a few volunteered without compensation because of their interest in the topic. All participants in session two provided informed consent.

2.2 The Survey

A 124-item online survey was designed to probe items that would provide information on users of portable audio devices in the university student population. Items were designed to investigate a number of topics, including: demographic characteristics, transportation usage patterns, work environments, personal and family hearing history, recreational activities (including noisy hobbies, frequency of attendance at bars, concerts, and sporting events), as well as questions on the use of portable devices. A number of items probed the participants' subjective estimations of the volume levels to which they set their own devices, and their subjective perceptions of their hearing abilities. Thus, the survey was intended to aid in establishing trends relating hearing loss to the degree of use of portable audio devices.

Two items were used to identify the participant number and date. In addition, there were 70 main questions and 52 sub-questions. All participants were asked to complete 48 main questions and 12 associated sub-questions. Only respondents who owned an iPod were asked to complete another 19 main questions and 15 associated sub-questions. Only iPod users who usually adjusted the equalizer settings were asked a further 3 main questions and 25 associated sub-questions. All main questions and associated sub-questions and response options are provided in Appendix A.

2.3 Audiometry

Audiograms were measured for the 24 participants who completed session two. Thresholds were tested for pure-tones of .25, .5, 1, 1.5, 2, 3, 4, 6, 8, 10, and 14 kHz. Participants sat in a sound-attenuating double-walled IAC booth. Standard audiological test frequencies were delivered to Telephonics TDH-50P HB7 headphones from a Grason-Stadler GSI-61 audiometer. High-frequency testing (above 8 kHz) was conducted using a special option on the audiometer and Sennheiser HAD 200 headphones. The ear tested was the one that the participant believed to be of lesser ability or if both ears were believed to be equally good then the left ear was tested.

Clinically normal results for hearing thresholds are considered to be between 0-25 dB HL (Mencher, 1997); following the recommendation of Mencher, if the thresholds of any participant exceeded 20 dB HL at 3 kHz or 30 dB HL at 4 kHz, and if the person wished to have a diagnostic hearing test, then a referral to an audiologist would have been made; however, no participant met these criteria.

2.4 Acoustical Measurement of Output

Acoustical measurements of output were obtained under a range of control conditions using a dummy head and at the preferred settings of the 24 participants in session two.

Equipment

All testing was conducted in a 10 x 12 foot IAC double-walled soundbooth. Music was presented to a dummy head and 24 participants from the same black 30-GB Apple iPod Video MP3 player (MA146LL/A), with standard earbuds. At the start of each session, a calibration test was done in which the output was checked with the iPod volume set to maximum and the equalizer set to the default position.

The output from the iPod was measured by placing the headset on a Brüel and Kjær (B & K, 2006) Sound Quality Head and Torso Simulator (HATS) type 4128-C-001 with binaural microphones and Zwislocki couplers. The output to the ears of the dummy head was measured using a B & K PULSE Sound Analyzer 9.0, Labshop version 9.0.0.352.

Stimuli

The same stimuli were used to obtain the output measurements in the control and user preference conditions. Two samples of music were used, each representative of a particular genre, either Hip-Hop or Electronica. These genres were chosen because they are very popular with undergraduates. A typical 30-second segment of each song was presented to calculate the mean dB (L_{eqA}) level output in each condition. Hip Hop songs are known for their strong percussions, thus most of their energy is concentrated in the low-frequency range between 1 and 4 kHz. The particular sample of Hip Hop tested was from the song "Cop that disk" by Missy Elliott and Timbaland and Magoo which includes vocals. In contrast, a typical Electronica song features synthetized sounds with more energy from 1 to 12 kHz. The particular sample used was from the song "Area 51" by Infected Mushroom, with no vocals. Average spectra for these two clips are shown in Figure 1. The time waveform of the Hip Hop clip is shown in Figure 2 and the time waveform of the Electronica clip is shown in Figure 3.

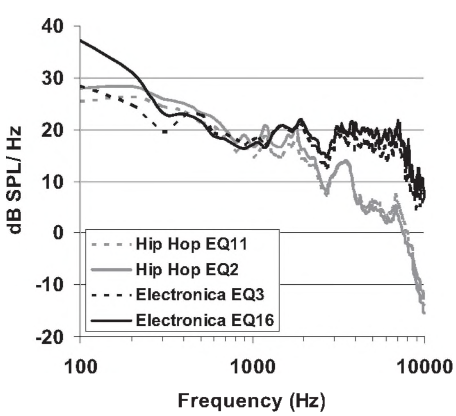

Figure 1. Spectra of Hip Hop clip played at volume 4 with equalizer settings 11 "latin" and 2 "bass booster", and of the Electronica clip played at volume 4 with equalizer settings 3 "bass reducer" and 16 "RNB".

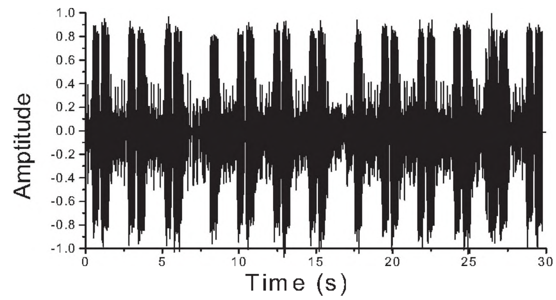

Figure 2. The time waveform of the Hip Hop clip.

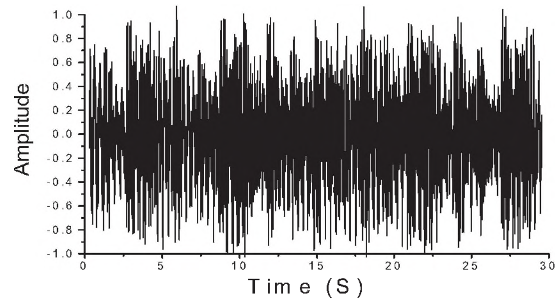

Figure 3. The time waveform of the Elecctronica clip.

2.4.1 Output at Control Settings

The B & K HATS was positioned on a small desk in a fixed location within the booth. There are 23 different equalizer settings options on the iPod and each of them was coded with numbers from 0 through 22. A marker divided into four equal segments was positioned below the iPod's visually displayed volume meter and it was used to divide the volume setting into one of four different ranges: 0-25, 25-50, 50-75 or 75-100%. Thus, the sound level in dBA was recorded at 25, 50, 75, and 100% volume settings for each of the 23 equalizer settings.

The output was measured for three different earbud – style headsets coupled to the iPod, one sold by Apple and two alternatives sold by other companies. The brands of headsets tested and their specifications are as follows:

1. 30-GB iPod Video Earphones with a frequency range of 100Hz - 20kHz, sensitivity of 90 ± -3 dB, impedance of 32 Ω ± -15% and maximum power input of 10mW.
2. Mirai Earphones (MI-SL-730BV-Black) with a frequency range of 20Hz - 20kHz, sensitivity of 113dB ± 3dB, impedance of 32 Ω and maximum power input of 60 mW.
3. Panasonic Earphones (RP-HV288) with a frequency range of 10Hz - 25kHz, sensitivity of 104 dB/mW, impedance of 16 Ω and maximum power input of 50 mW

The standard iPod earphones were the first headset to be assessed. After positioning the earphones in the B & K HATS, a 30-second Hip Hop sound clip was presented to the dummy. The average dBA level was recorded. The initial equalizer and volume setting were adjusted to 0=default and 25%, respectively. The measurements were obtained in the same way for the three other volume levels at the default equalizer setting. Then the next equalizer setting was tested at each of the four volume levels until all 23 equalizers had been tested using the Hip Hop song. The procedure was repeated with the Electronica test stimulus.

For each condition, the average difference in dBA between Hip Hop and Electronica was calculated. The largest and smallest differences at each volume level were used to determine which equalizer settings made the most difference (see Results section). Based on these findings, two equalizer options were selected for further testing with the other two headphones. Specifically, Mirai and Panasonic earphones were tested at the equalizer settings 10=jazz and 13=lounge across the four different volume levels.

2.4.2 Output at User Preference Settings

The 24 participants listened to each sample of music in five background conditions: quiet, multi-talker babble at 50 and 70 dB SPL, and traffic noise at 50 and 70 dB SPL. The levels of noise were chosen to typify mildly and moderately adverse conditions representative of everyday situations.

Participants sat in the center of the booth facing two loud speakers, one in the back left and the other in the back right corner. The background noise was delivered over both loud speakers, positioned at 45^0 on either side of the listener. The delivery of background sound was controlled using a TDT System III. The participants wore the iPod and standard headset in both ears to listen to the music samples.

Participants completed the experiment in one of four orders. In two orders, Electronica was tested first and Hip-Hop second, and the types of music were reversed in the other two orders. Within each music type condition, half of the time traffic noise was tested first and multi-talker babble was tested second, and the types of background noise were reversed otherwise.

Each participant was told he or she would listen to ten music clips, five from each of the two genres. For each clip in each background condition, the participant was asked to adjust the iPod to listen to the song "the way you like it the best". The participants were shown how to adjust the iPod volume and equalizer controls, but they were not required to do so unless they wanted to. At the start of each condition, the equalizer was set to the default position. After the participant was satisfied with his or her adjustment in each condition, the experimenter noted the volume setting with reference to a pre-marked scale from 0 (minimum, 1-25%) to 4 (maximum, 75-100%). The earbuds were then placed in the dummy's ears and the 30-second clip was analyzed using the PULSE Sound Analyzer to determine the average output in dBA for each ear at the user's preferred settings.

3. RESULTS AND DISCUSSION

The main subjective and objective findings will be reported first, and then relationships between the findings in the present study and the study of high school students (Zogby, 2006) will be examined to illuminate whether or not the use of digital portable audio devices poses a risk to the hearing health of university students and how this compares to the findings for high school students.

3.1 The Survey

Portable Audio Device Use

Most (82%) of the participants owned a portable audio device, with the most popular devices being MP3 players (42.7%), iPods (34%), CD players (32.7%), and cell phones (29.3%) (Figure 4). Some students (N=26) listen more often to their device through loudspeakers, but it is more common for students to listen with a headset. About as many (N=51) use standard earbuds as use headphones (N=43), with only 5 students reporting that they use extended range earbuds. About half of the device owners reported using their portable audio device for 5 to 7 days per week, and for a duration of 2 hours or more per listening session. Of the device users, 35 reported listening to their devices as frequently as seven days per week, and 7 listened to their device for as long as 4 to 8 hours in a typical single session. Figure 5 shows the typical length of listening sessions.

The median volume level for students using a headset was 60% on a scale from 0 to 100%, with 100% being the maximum volume; however, there were 17 individuals who reported setting the sound level in the 80-100% range (Figure 6). The median level to which the students using a loudspeaker adjusted their devices was also 60%; however, there were 25 individuals who reported setting the sound level in the 80-100% range when using the device in this fashion. This initial evidence suggests that most students use their devices frequently, but in mid volume ranges, regardless of how frequently the device is used. Nevertheless, there is a minority who may use their devices very frequently for long sessions and at high volumes.

Gender differences were observed (Figure 7). More females than males reported setting the volume in the 25-50% range. Curiously, more males than females preferred the highest ranges, but of those who preferred the lowest volume, more were males.

On average, undergraduates seem to use their devices for less time and at lower levels than high school students. This finding is consistent with age-related trends in frequency and during of participation in other noisy leisure activities (Cheesman, Ciona, Mendoza & Grew, 2001). The most likely explanation seems to be that university students have less leisure time than high school students.

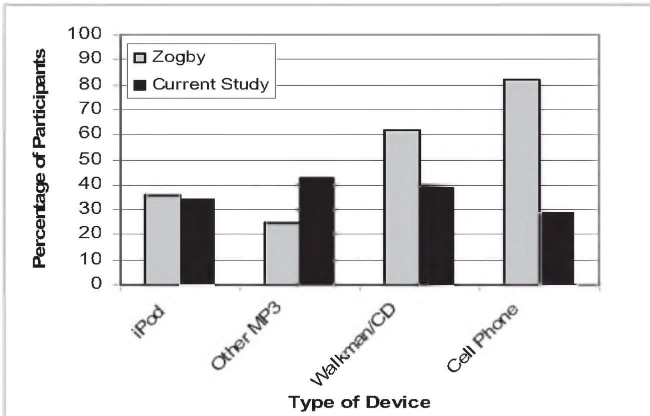

Figure 4. Percentage of respondents in the current study (N = 118) and percentage of 301 high school students (Zogby, 2006) who own various popular portable audio devices.

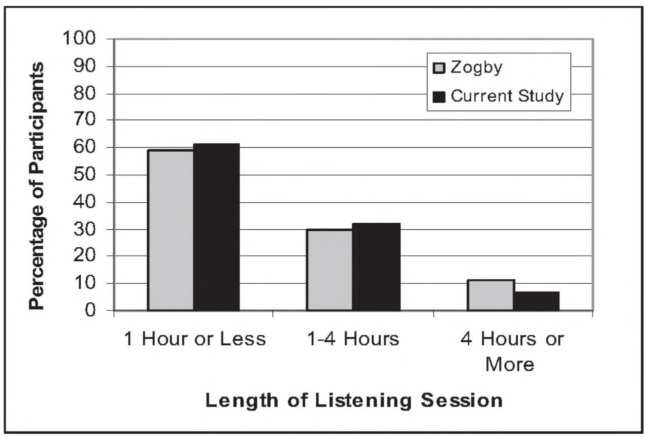

Figure 5. Percentage of respondents owning devices in the current study (N = 118) and percentage of 301 high school students (Zogby, 2006) reporting typical lengths of a listening session.

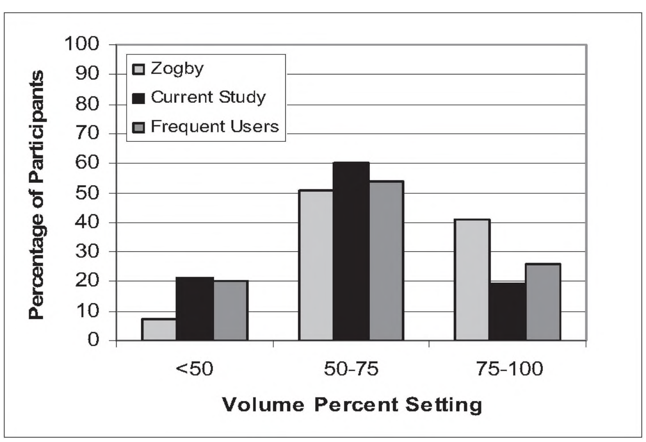

Figure 6. Percentage of students reporting their usual volume setting (in percent of scale) in the study of high school students by Zogby (2006) and in the current study for all respondents who owned a device (N = 121), and for our 35 frequent users who reported using their device 7 days per week.

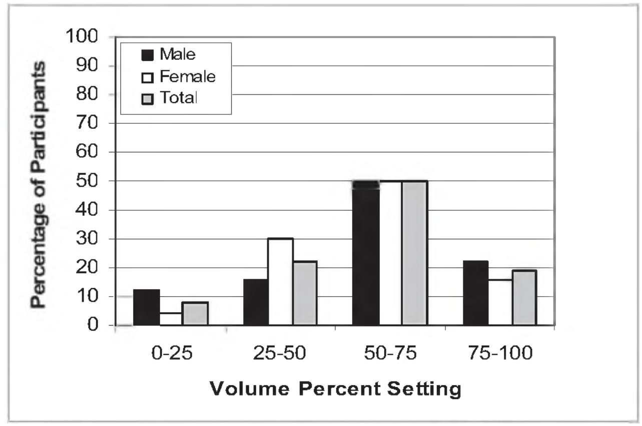

Figure 7. Percentage of male and female respondents owning of portable audio devices (N = 121) reporting volume preferences with volume divided into quadrants.

Work, Transportation, and Recreational Noise

About half of the students (53.6%) were employed at the time of the survey, with about a third of them working less than 10 hours per week, a third working less than 20 hours per week, and a third working more hours per week. About a fifth (21.2%) did volunteer work. The most common work and volunteer settings were community (sports complex, community center, mall, department store) and institutional (library, school, childcare, clinic) locations. Relatively few reported working in recreational settings such as clubs, bars, restaurants, or cinemas (7.9%) or industrial settings such as in factories, manufacturing, or warehouses (6.6%) that would be more likely to be noisy. Nevertheless, almost half (45%) considered their work/volunteer setting to be moderately noisy and another third (32.5%) considered their settings to be more than moderately noisy. The majority (78.2%) seldom or never wear a portable audio device while working or volunteering.

Most students (84.1%) travel to university more than three days per week, with the duration of the trip for most students being less than one hour (88.1%). About half (51%) of the respondents usually travel by car, but many (38.4%) travel by bus, with those using a portable audio device while traveling being distributed bi-modally into those who never listen (36.4%) and those who frequently listen (35.1%). A fifth (20%) reported that commuting was the most common situation in which they used their device.

About half of the students reported going to nightclubs (49.3%) or bars (50%), with about a third going at least once a month to clubs (30.6%) or bars (40.6%). As many as 65 students reported attending a concert once or twice a year, but only 26 students reported attending more than two concerts per year. By comparison, slightly more students (59.3%) attended a sporting event at least once a year, but almost all (98.7%) went to a movie at least once a year.

About a third (34%) play a musical instrument, but few (6.7%) were members of a band. The most popular noisy hobbies were motorcycling/go-karting (17.3%) or ski/sea-dooing (11.3%). However, the majority (76%) reported that they had no noisy hobbies. Regular past times were reported to be watching television (86.7%), sports or exercise (84.7%), reading (64.7%), and playing computer or video games (60%). About a fifth of the students (19.4%) used their portable audio device most frequently during recreational activities, including while exercising (22.7%), walking (14.7%), or during other leisure activities (12%).

Mood seemed to influence device use with frequent use being reported when students were bored (28%), followed by when they were experiencing the common positive or negative emotions of being happy (19.3%) or sad (17.3%), and followed next by when they were experiencing more extreme positive or negative emotions of being excited (15.3%) or depressed (15.3%). Use dropped when they were experiencing feelings such as being angry (13.3%), upset (12.7%), frustrated (10%), tired (9.3%), or hungry (7.3%).

Although some students (11.3%) reported using ear protection, more (N=20) used it while studying or sleeping than the number (N=18) who used it when operating noisy equipment or engaging in noisy hobbies. Only two reported wearing ear protection when attending concerts.

Self-reported Symptoms of Hearing Loss

Self-reports of concerns about hearing (responses 1-3 on question 25) indicated that 31.4% of the university students thought their hearing was worse now than five years ago (Figure 8). Also, 13.3% believed they had a hearing loss (question 26) and 12.7% believed it was noise related. Nevertheless, fewer (8%) reported that difficulty with hearing limited or hampered their personal or social life (responses 6-7 on question 32), and 10% reported that difficulty with their hearing upset them (responses 6-7 on question 33). Importantly, the findings did not differ greatly between the users and non-users of portable audio devices (Figure 9). It is interesting that more non-users than users of portable audio devices rated their hearing as being worse and suspected a hearing loss, yet only frequent users and those who preferred to set the volume high reported negative social or emotional effects of hearing difficulties.

Although our questions and those of Zogby (2006) are not identical, some comparisons can be made regarding self-perceived hearing health. The high school students responded on a scale ranging from "*very concerned*" to "*not at all concerned*" to the question *"How concerned are you about losing your hearing as you age?"*, with responses of "*somewhat"* or "*very concerned*" being taken by Zogby to indicate that the student was concerned.

Zogby (2006) examined three main symptoms of possible hearing loss: turning up the television, tinnitus, and communication problems. Comparisons to similar questions in our study are shown in Figure 10.

Overall, more university students (69%) than high school students (46%) reported no concern about their hearing (Figure 9), but fewer university students (25%) reported none of the symptoms of possible hearing loss compared to the high school students (49%). As shown in Figure 10, about a third (30%) of the university sample reported turning up the TV volume more than they used to (compared to 27% of high school students). Tinnitus following exposure to loud sound was reported by 51.3% of the university students, but 76% reported that usually they never or rarely have ringing. Tinnitus was reported by 17% of high school students.

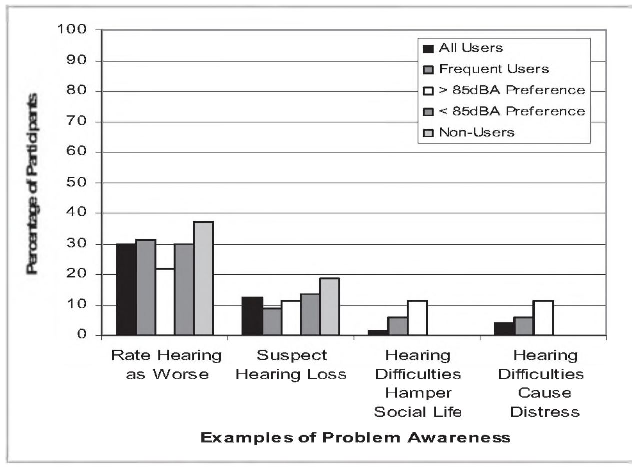

Figure 8. Percentage of students reporting awareness of hearing problems. All users include respondents owning a portable audio device (N = 123). Frequent users (N=35) reported using their device 7 days/week. The >85dBA preference group (N = 9) set the iPod volume at or over 85dBA at least once in session two of the current study. The <85dBA preference group (N = 15) always set the iPod volume below 85dBA in session two of the current study. Non-users (N = 27) are our survey respondents who do not own a portable audio device.

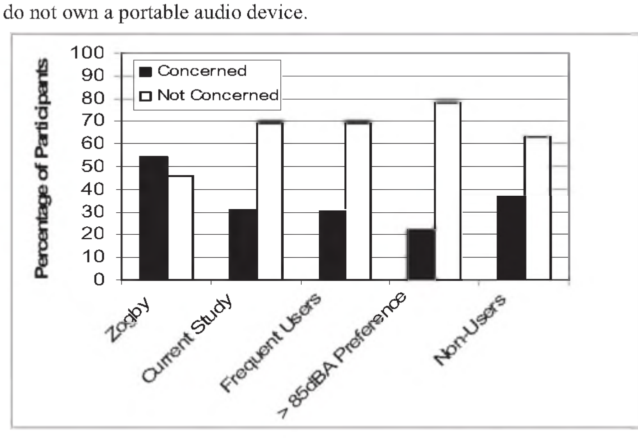

Figure 9. Percentage of students reporting concern about hearing problems for high school students (Zogby, 2006) and university students in the current study.

In terms of communication problems that may be symptoms of hearing loss, our survey probed whether or not students experienced various problems when they were or were not using their device. The responses to two of our items (questions 30 and 64) were averaged to determine an overall communication rating that could be compared to the communication item of Zogby (Figure 10).

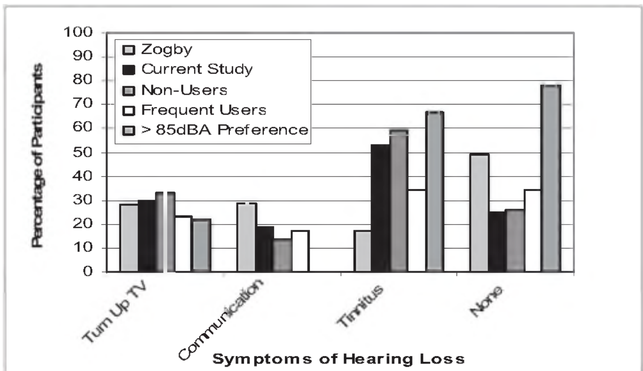

Figure 10. Percentage of students reporting symptoms of hearing loss for high school students (Zogby, 2006), and university students in the current study. Percentages are also shown for non-users (N = 27) and two possibly at risk subgroups, frequent users (N = 35), and those who set the iPod volume at or above 85 dBA in session two (N = 9).

Many aspects of communication are challenged when students are using a portable audio device (Figure 11). Evidence of a problem hearing the doorbell is given by a response of 1-4 on questions 27 or 60; problems in speech communication are suggested by responses of 1-5 on questions 28 or 61 (phone talk), questions 29 or 62 (whisper), questions 30 or 64 (talk in quiet), or questions 31 or 63 (talk in noise). Further comparisons between different types of users without their device (Figure 12) and with their device (Figure 13) suggest that when not wearing the device those who prefer higher volumes experience more problems than those who prefer lower volumes. In contrast, when the device is worn, those who prefer lower volumes experience more problems than those who prefer higher volumes.

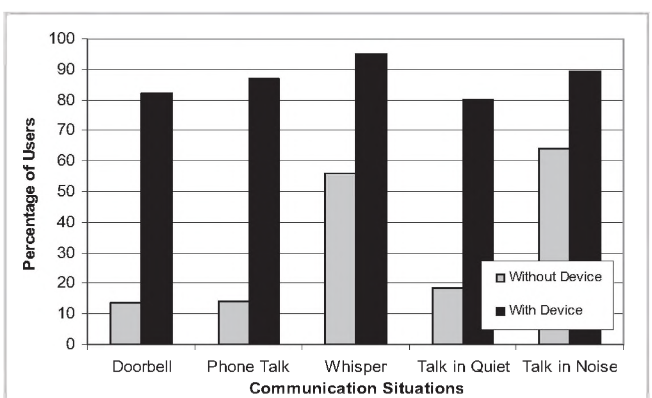

Figure 11. Percentage of students owning devices (N = 123) who report communication problems when wearing or not wearing their own device.

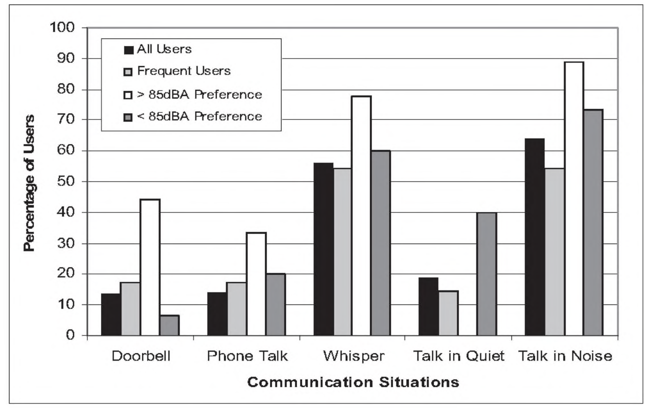

Figure 12. Percentage of students reporting communication problems when NOT wearing a portable audio device for university students in the current study who own a device (N = 123). Percentages are also shown for frequent users (N = 35), and for those who sometimes set the iPod volume at or above 85 dBA (N = 9) or always below 85 dBA (N = 15) in session two of the current study.

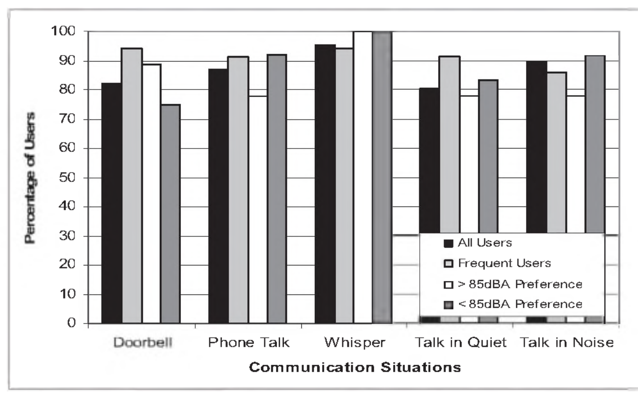

Figure 13. Percentage of students reporting communication problems when they DO wear a portable audio device for university students in the current study who own a device (N = 123). Percentages are also shown for frequent users (N = 35), and for those who sometimes set the iPod volume at or above 85 dBA (N = 9) or always below 85 dBA (N = 15) in session two of the current study.

3.2 Audiometry

The distribution of hearing thresholds (dB HL) measured at each of the test frequencies is shown in Figure 14. All 24 of the students who completed the second session had thresholds less than 10 dB HL at 1, 1.5, and 4 kHz. For all other frequencies, at least ¾ of the students had thresholds better than 10 dB HL, well within the normal range. Only two participants had a threshold falling outside of the range considered to be clinically normal (> 25 dBHL). One of the abnormal thresholds was 30 dB HL for .25 kHz, and the other was 40 dB HL at 14 kHz. Thus, there were no noteworthy early signs of hearing loss, including no suggestion of the classic 4 kHz notch that is usually taken as an early sign of noise-induced hearing loss in industrial hearing conservation programs.

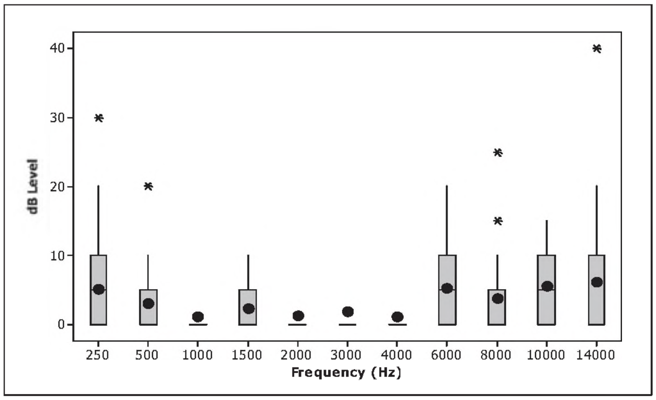

Figure 14. Box and whisker plots showing the mean dB HL threshold (black dots), median (centres of box), inter-quartile range (box ends), and minimum and maximum (ends of whisker lines) thresholds of hearing (dB HL) at each pure-tone frequency tested for the 24 university students who participated in session two of the current study. Outliers are indicated by *.

3.3 Acoustical Measurement of Output

The output of an iPod was measured for two samples of music played at pre-selected volume and equalizer control settings on the B&K HATS dummy head. The output was also measured for the same samples at the preferred settings of the 24 students who completed session two.

3.3.1 Control Settings

For the iPod earphones, the dBA measurements of both ears of the dummy were averaged and plotted for the 23 equalizer settings at four volume levels. Not surprisingly, dBA levels increased with volume. For Hip Hop, the average output (± SD) increased from volume 1 through 4 as follows: 52.5(±5.3), 61.9(±8.2), 77.9(±5.7) and 93.7(±5.7) dBA.

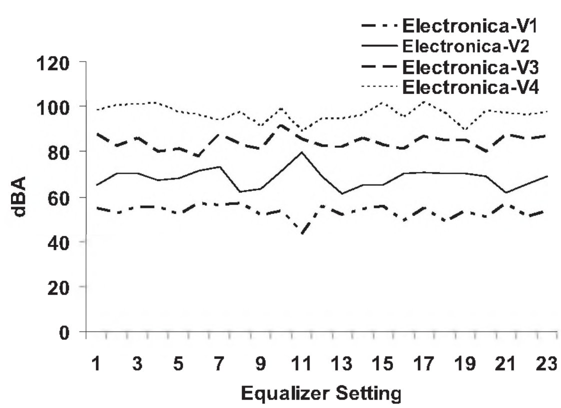

Figure 15. Average of left and right-ear dBA outputs for iPod earphones with the Electronica music at four volumes and 23 equalizer settings.

Interestingly, the average dBA output was greater for Electronica than for Hip Hop for volume levels 1 through 4 respectively: 53.6(±3.2), 68.4(±4.2), 84.3(±3.3), 96.7(±3.6) dBA (Figure 15). The output depended on equalizer and volume settings; however, the variability was greater for Hip Hop.

At each volume and equalizer setting, the difference due to genre was calculated by subtracting the dBA outputs (Hip Hop - Electronica); e.g., at volume 4, the largest negative difference was -19 dB (equalizer 13, "lounge") and the largest positive difference was +8.3 dB (equalizer 10, "jazz"). Thus, it was interesting to examine these two settings more closely because they seemed to illustrate how genre and equalizer settings might interact to affect output at different volume settings. Accordingly, we tested the two alternative transducers, the Mirai and Panasonic earphones, at the default setting and with the equalizer set at 10=jazz and 13=lounge. All tests with the Mirai and Panosonic earphones were conducted at a 50% volume setting because that was assumed to be a typical user setting, but of course, we do not know if a listener would adjust the volume in the same way in all conditions for all headsets.

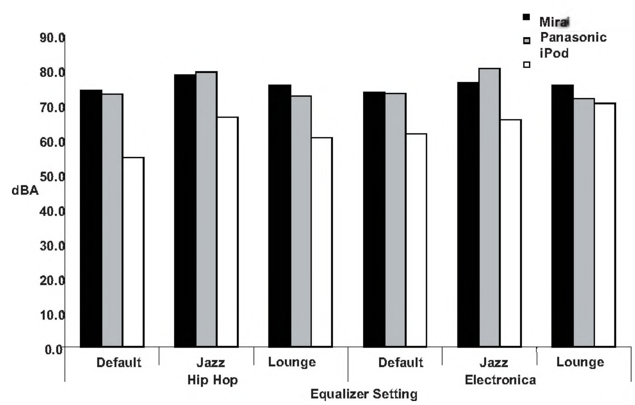

Figure 16. Left and right-ear average dBA outputs for three earphones at 50% volume with default, jazz and lounge equalizer settings for Electronica and Hip Hop.

The dBA output from the Mirai and Panasonic earphones were comparable for each equalizer setting across the two test stimuli (Figure 16). They always yielded dBA output levels between 70 and 80 dBA. The largest difference between them was 4.1 dB in the jazz setting for Electronica. Importantly, the outputs were consistently greater than the outputs from iPod earphones. At the default and lounge settings, the output of the Mirai exceeded that of the Panasonic, and both exceeded the output from the iPod earbuds. This trend, however, was not maintained at the jazz setting where the output of the Panasonic earbuds was slightly greater than that of the Mirai earbuds, but again both alternative transducers produced higher outputs than the standard iPod earbuds. Compatible results were reported by Fligor and Ives (2006) who also found reductions when noise cancellation headsets were tested at lower volumes.

3.3.2 User Preference Settings

On average, listeners adjusted the iPod to 67.6 dBA, but their preferred listening level depended on background noise, and less so on type of music and the ear (Figure 17).

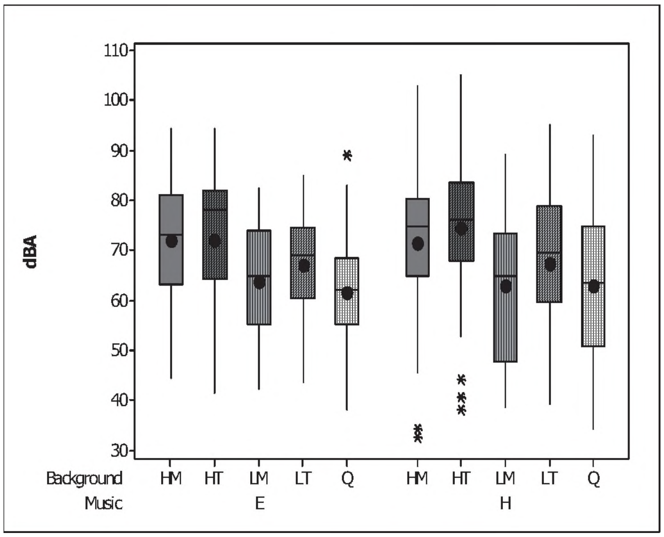

Figure17. Box and whisker plots showing the means (black dots), medians (lines in boxes), inter-quartile ranges (box ends), and minimum and maximum values (ends of whisker lines) for dBA outputs measured when the iPod was adjusted by listeners to their preferred settings for Electronica (E) and Hip-Hop (H) music under five background conditions (HM = high,-level multi-talker babble, HT high-level traffic noise, LM = low-level multi-talker babble, LT = low-level traffic noise, Q= quiet). Results for the two ears are averaged. Outliers are indicated by *.

Output was lowest when there was no background noise (62.1 dBA) or when there was a low level of multi-talker babble (63.4 dBA). Output was greater when there was a low level of traffic noise (67.2 dBA). Output was greatest when there was high level noise, with little difference between multi-talker babble noise (71.7 dBA) and traffic noise (73.3 dBA). Curiously, mean output was 2.7 dB greater for the right than for the left ear.

These descriptions were confirmed by an Analysis of Variance with three within-subjects factors: Ear (right or left), Music (Hip-hop or Electronica), and Background (quiet, low-level multi-talker babble, high-level multi-talker babble, low-level traffic noise, high-level traffic noise). There were significant main effects of ear $F(1, 23) = 35.2$, $p < .01$), and background, $F(4,92) = 21.3$, $p < .01$. A Student-Newman-Keuls test of multiple comparisons confirmed that there was no significant difference between the outputs preferred in the quiet and low-level babble, but that the outputs in these conditions were significantly lower than in low-level traffic noise, and that the outputs in high-level noise conditions were significantly greater ($p = .05$).

3.4 Relationships between Objective and Subjective Measures

3.4.1 Output

Of the 24 students who participated in session two, 20 owned a portable audio device. For the 20 who owned a device, it was possible to compare how they actually set the volume of the iPod when they were listening in session two to their subjective questionnaire responses estimating how they thought they usually set the volume.

Importantly, there were discrepancies between their objective measures and their subjective estimations of preferred volume level. Most participants subjectively estimated that they set the volume above halfway, whereas the objective tests of user preferences indicate that many set the volume above halfway when the background noise level was high, but that more students set it less than halfway when the background noise level was lower (Figure 18). A minority of students (N = 4) underestimated their usual volume settings, reporting that they usually set the volume of their own device to be in the second quadrant (25-50% of the volume scale) while objective measures indicated that they set the volume of the test iPod higher. There were no noteworthy gender differences in either the objective or subjective reports of volume settings for these 24 students.

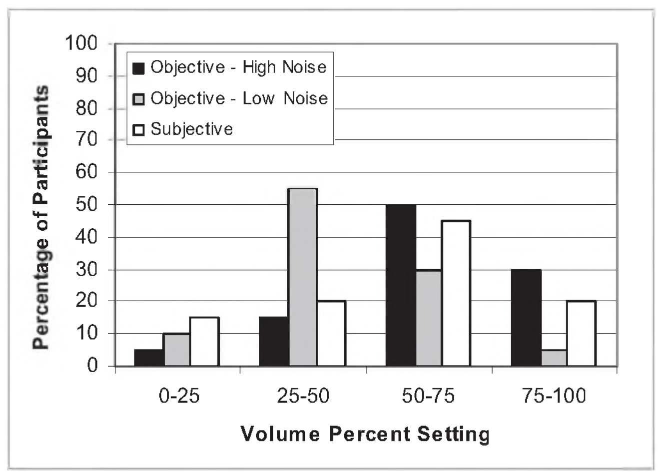

Figure 18. Percentage of participants in the current study who own a portable audio device and who completed both sessions of the current study (N = 20) subjectively estimating their usual volume setting in the survey administered in session one, and objectively adjusting the volume of the iPod in session two in high and low background noise conditions, where volume is divided into four quadrants in terms of percent volume.

Of the 20 device owners for whom both subjective and objective measures of volume setting were obtained, almost half owned more than one portable audio device: 11 owned an MP3 Player, 11 have a CD player and there were 5 iPod owners. It is possible that some of the apparent discrepancy between the subjective and objective measures of how students set the volume might be explained by differences in the specific equipment used by individuals. However, we found no obvious relationship between the objectively measured volume settings and the type of portable audio device and headset owned by the students. Interestingly, 9 students who owned a portable audio device used standard earbuds and 9 used headphones when listening to their devices. Two students used speakers with their device. Only one participant reported using extended-range ear buds that might provide higher actual output than the standard headphones for the same volume control setting.

As seen in the measures of output at predetermined control volume and equalizer settings, the choice of equalizer may also influence the actual output achieved for a given volume control setting. The participants reported adjusting equalizer settings, but not surprisingly they did so less frequently than they adjusted volume settings. About half of the students reported on the survey that they used the equalizer options on their own device, although few students adjusted it during the objective testing.

Even though we did not find significant differences between Electronica and Hip-Hop when measuring output from the test iPod, the majority of the participants reported in the survey that they frequently change volume depending on the genre of music they are playing. Also, the majority of students reported changing the volume of their own device in response to changes in environment, which agrees with the objective output measurements taken in session two in which students increased the volume of the iPod in the high noise conditions. Change in environment was reported to be the most frequent reason for volume adjustments, followed by change in song quality and song loudness. Song genre, environment, and song quality are frequent reasons for use of equalizer settings. Emotions play a role as well, with over half of the students reporting that they adjusted volume and equalizer settings with changes in their mood. They reported listening to their portable audio device mostly when they are bored or when not experiencing extreme emotions.

Thus, the output actually experienced by owners of portable audio devices on a daily basis seems to depend not only on the properties of the device, but also on the settings and the transducer used with the device, and on a variety of non-technical factors. Non-technical factors include stimulus factors such as type of music, personal factors such as mood, as well as environmental factors. Environmental factors range from physical characteristics of the environment (e.g., level of background noise) to social aspects of the environment (e.g., type of activity).

3.4.2 Exposure

The complexity of the factors influencing the actual output experienced by a listener is matched by the complexity of the factors influencing the duration of usage. To determine whether or not the exposure to noise presents a risk to students using portable audio devices, both output and duration must be determined.

It is commonly accepted that 85 dBA for a period of 8 hours is the maximum level of safe exposure (NIOSH). With every 3 dB increase, the duration for safe exposure is halved. The calculations of industrial noise exposure are based on measurements taken in the soundfield rather than measurements taken in the ear canal. The resonances of the outer ear structures effectively amplify the sound, especially in the range from 2 to 4 kHz. Correcting for the head-related transfer function to convert the measurements in the ear canal to corresponding soundfield measurements would yield lower output levels.

In our study, at 100% volume, regardless of type of music or equalizer settings, the sound levels measured in the ear canal exceed 85 dBA, and even after adjusting for the head-related transfer function levels over 85 dBA were found, depending on the equalizer setting and sample of music. Following this logic of using industrial safety limits as a yard stick, guidelines for the safe use of iPods have recently been suggested (Portnuff & Fligor, 2006). The suggested guideline is that there does not need to be a limit on listening time if the volume is set to under 50%. However, for the standard iPod earbuds, a limit of 6 hours per day is suggested if the volume is set at 70%, 1.5 hours if it is set at 80%, 22 minutes if it is set at 90%, and 5 minutes if it is set at 100%. Of course, these guidelines do not take into account differences in output that depend on the equalizer setting and type of music even when the volume is held constant. For example, we examined conditions where extreme differences were observed and discovered that there could be as much as a 27.3 dB difference between the outputs measured for different equalizer settings.

Interestingly, those who set their devices to relatively high levels (at or over 85dBA) during session two are not necessarily more likely to listen to their personal audio devices for longer periods compared to those who preferred lower outputs. About half of the participants who owned a device reported listening to it for up to 2 hours daily for up to 5 days per week. According to the guidelines suggested by Portnuff and Fligor (2006), 2 daily hours of usage would become a problem if the volume were set at 80%, as might be done by about a fifth of our sample.

In addition to noise exposure from portable audio devices, about half of the students were exposed to other sources of high-level noise during commuting, at work or volunteering, or during recreation, but the duration of these exposures is relatively brief and their frequency fairly rare.

For example, about half of the students reported attending nightclubs, bars, and concerts, but the frequency of attendance at such events is usually not more than monthly and few students reported attending more than two concerts a year. Most were exposed to noise during commuting for less than an hour per trip and most of those who worked did so for less than 20 hours per week. Unfortunately, some but not all used ear protection when exposed to potentially hazardous levels of noise. Again, it seems that most university students are not exposed to hazardous levels of noise, but there are a minority who could be at risk.

Overall, within the population of university students, exposure to hazardous doses of noise from the use of portable audio devices or from other noise sources probably does not exceed a "safe" range for the majority of students. Nevertheless, there are individuals whose exposures from these devices, combined with exposures to other noise sources, may pose a risk. To examine the subgroup that seems likely to be at greater risk, we focused on the frequent users of portable audio devices who use their device every day, and on users who prefer outputs at or greater than 85 dBA. We found that frequent users generally seemed to be more cautious about their use of the device and that they expressed more concern about hearing health. The fact that they have begun to experience problems may be prompting them to be more cautious about setting the volume too high.

3.5.3 Evidence of Early Hearing Loss

The estimates of exposures and the reports of concerns about and self-reported symptoms of hearing loss suggest that there may be a subgroup of student owners of portable audio devices who are at risk for noise-induced hearing loss. Nevertheless, there was no audiometric evidence of early signs of hearing loss. Of course, it may take years for the cumulative damage due to noise to cause changes in the audiogram. Thus, the absence of elevated pure-tone thresholds does not prove that the auditory systems of these students have not been damaged by noise. Other kinds of tests such as otoacoustic emissions might provide a more sensitive measure of the health of the outer hair cells which are known to be damaged by noise.

The audiograms were measured on only 24 participants, so the matter of the size of the sample should be considered. In fact, this sample size is not that much smaller than the sample of 60 who were tested in an earlier study that reported a 40% rate of hearing loss in high school and university students (Lees et al., 1985). In their study, the criteria for hearing loss was a 10 dB threshold difference between thresholds at adjacent test frequencies; however, in standard audiometry, test error is considered to be $\pm$ 5 dB, so it is possible that a 10 dB difference could be attributable to test error. In addition, most of the problems they found involved a threshold shift of an average of 20 dB at 6 kHz, not the classic 4 kHz noise notch, and it is common for problems at 6 kHz to result from poor earphone placement. Thus, even though previous studies have reported alarming rates of hearing loss amongst young adults, there does not seem to be a strong enough literature to judge how widespread early noise-induced hearing loss might be or what cohort differences might exist.

Another matter to be considered is whether our self-selected sample might have been biased, especially since they had been given an information session about the dangers of noise after they completed the survey in session one. One possibility is that those who were more worried about their hearing chose to volunteer for session two. Another possibility is that those who were iPod enthusiasts volunteered for session two. The comparison of the subjective measures to the objective measures of iPod use suggest that the participants in session two did respond accurately insofar as their self-reported volume preferences were lower than the settings they selected in session two when the background noise level was high, but their self-reported volume preferences were higher than the setting chosen in session two when the background noise level was low. Furthermore, those who volunteered for session two seemed to be typical of the larger group of 150 students who had completed the survey during session one of the study in terms of their iPod ownership and self-reported use patterns.

4. GENERAL DISCUSSION

Our study confirmed that the majority of undergraduate university students own at least one portable audio device. Therefore, noise-induced hearing loss could be a larger concern now than ever before, especially if the portable audio devices are worn for excessively long durations and/or their wearers are also exposed to other noise sources.

Despite the potential for exposure of these students to hazardous levels of noise, the majority of survey respondents reported that they typically set their device at mid volumes, and only a minority of students reported their work/volunteer places to be loud. The overall 'safe' use of iPods by most undergraduates was confirmed in an experiment to objectively measure the output of an iPod when users adjusted the device to their preferred volume and equalizer setting while listening to music in five different conditions of background ranging from quiet to 70 dB SPL. These findings are consistent with a study of iPod use by 100 graduate students (Filgor & Ives, 2006).

No audiometric signs of early hearing loss were found. Nevertheless, cause for concern is raised by the finding that a third of the participants felt that their hearing had worsened in the past five years, and only 12% of participants ever used hearing protection. Longitudinal studies will be needed to monitor how hearing changes over many years of use of portable audio devices in this cohort.

We also identified subgroups of students who may be at higher risk because of their frequent use of a portable audio device or their preference for setting it to levels exceeding 85 dBA. Thus, our data suggests that although most students tend not to expose themselves to excessive noise (recreational noise, noise at work, or music noise while listening to portable audio devices), there are some individuals who may be at risk. At the same time, those who are at most risk seem to be the least concerned about their hearing. They may simply be not aware that exposure to loud music can result in hearing loss. This assumption was also stated by Chung and colleagues (2005) who conducted an online survey of views on health issues including hearing loss in adolescents and young adults. They concluded that some types of education may be crucial to motivating young people to change their listening habits. In the study by Zogby (2006), respondents indicated that school classes, teen magazines, and TV programs may be effective means for educating young people about hearing. Thus, it seems that, with such widespread use of portable audio devices among young adults, increasing awareness about hearing and early noise-induced hearing loss is essential. Boosting their motivation to protect their hearing may protect this generation from widespread future hearing problems.

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6. APPENDIX A

UTM Portable Audio Device Questionnaire

Basic Demographic Information

1. What is your gender?
   - o Male
   - o Female

2. What is your age?
   - o 5-15
   - o 16-20
   - o 21-25
   - o 26-35
   - o > 35 years

3. What is your marital status?
   - o Married
   - o Life Partners
   - o Single
   - o Divorced
   - o Other

4. What best describes your living situation?
   - o Share off-campus accommodation with peers
   - o Share on-campus accommodation with peers
   - o Living with my family
   - o Living alone
   - o Share an accommodation with strangers

Transportation

5. a. Which means of transportation do you use most often when commuting to UTM?
   - o Bus
   - o Walk
   - o Car
   - o Train
   - o Motorcycle
   - o Other, please specify:___________________

   b. Do you listen to a portable audio device while using this means of transportation?

   Never Occasionally Frequently

   1 2 3 4 5 6 7

6. On average how often do you commute to UTM in a week? (number of round-trips)
   - o 0
   - o 1-2
   - o 3-4
   - o 5-6
   - o > 7 trips

7. On average what is the duration of your trip to UTM (one way)?
   - o 0 – 30 minutes
   - o 30 – 60 minutes
   - o 60 – 90 minutes
   - o 90 – 120 minutes
   - o > 120 minutes

8. a. What is your primary means of transportation on a daily basis when NOT at UTM?
   - o Bus
   - o Walk
   - o Car
   - o Train
   - o Motorcycle
   - o Other, please specify:___________________

   b. How often do you listen to a portable audio device while using this means of transportation?

   Never Occasionally Frequently

   1 2 3 4 5 6 7

Work

9. a. Are you currently employed?

   YES NO

   b. If yes, how many hours per week do you work on average?
   - o 1 – 10 hours
   - o 11 – 20 hours
   - o 21 – 30 hours
   - o 31 – 40 hours
   - o > 41 hours

10. Are you currently volunteering?

    YES NO

11. Which best describes the setting of your most frequent work/volunteer place?
    - o Community (e.g., sport complex, community center, mall, departmental stores)
    - o Residential (e.g., university residence, seniors' homes)
    - o Institutional (e.g., library, school, tutoring centers, day care, medical clinics)
    - o Recreational (e.g., clubs, bars, restaurants, movie theaters)
    - o Office
    - o Industrial settings (e.g., factory, manufacturing, warehouse)

12. Do you consider your primary place of work/volunteering to be…?

Very Quiet		Moderate		Very Loud		
1	2	3	4	5	6	7

13. Do you listen to a portable audio device at your place of work/volunteering?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

Hearing History

14. a. Did a family member have a hearing loss before old age?

YES NO

b. If yes, what is their relationship to you?

- o Father
- o Mother
- o Brother
- o Sister
- o Aunt
- o Uncle
- o Grandmother
- o Grandfather
- o Cousin
- o Other

c. If yes, do you know what type it is?

- o Age-related
- o Noise-induced
- o Present at birth
- o Other (i.e., due to illness, drug side effect)
- o Don't know

15. Do you wear a hearing aid?

YES NO

16. How often do you get ear infections?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

17. Do you get colds?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

18. Do you have allergies (e.g., hay fever)?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

19. Do you take any medication on a regular basis that makes your ears ring?

YES NO

20. Do you have ringing in your ears?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

21. After being exposed to loud sound, do you experience ringing in the ears?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

22. Do you experience the feeling of cotton in your ears after exposure to loud sound?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

23. Do you turn up the TV more than you used to?

YES NO

24. How much do you enjoy listening to loud music?

Not at all		Somewhat		Very Much		
1	2	3	4	5	6	7

25. How do you rate your hearing compared to five years ago?

Much Worse		Same		Much Better		
1	2	3	4	5	6	7

26. a. Do you think you have a hearing loss?

YES NO

b. If so, what do you think is the cause?

- o Age-related
- o Noise-induced
- o Present at birth
- o Other (i.e., due to illness, drug side effect)
- o Don't know

<u>When NOT wearing any portable audio devices…</u>

27. I hear the doorbell…

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

28. I can have a phone conversation with…

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

29. I can hear someone speaking in a whisper with

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

30. I can carry on a conversation with one other person when in a quiet place (e.g., library) with

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

31. I can carry on a conversation with one other person when in a noisy place (e.g., party) with

Difficulty		Some difficulty			No difficulty	
1	2	3	4	5	6	7

32. I feel that difficulty with my hearing limits or hampers my personal or social life…

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

33. Difficulty with my hearing upsets me…

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

Recreational Activities

34. What genre(s) of music do you listen to on a regular basis? (check all that apply)
- ○ Hip-Hop
- ○ Jazz
- ○ Pop
- ○ Rock
- ○ Alternative
- ○ Punk
- ○ Reggae
- ○ Other, please specify:___________________

35. How many hours a day do you listen to music?
____________average hours per day

36. a. Do you go to nightclubs?
YES NO

b. If yes, how often?

Daily	Weekly		Monthly		Annually	
1	2	3	4	5	6	7

37. a. Do you go to bars?
YES NO

b. If yes, how often?

Daily	Weekly		Monthly		Annually	
1	2	3	4	5	6	7

38. a. How many concerts do you attend in a year?
_____________concerts

b. If yes, which type do you typically attend?
- ○ Hip-Hop
- ○ Jazz
- ○ Pop
- ○ Rock
- ○ Alternative
- ○ Punk
- ○ Reggae
- ○ Other, please specify:___________________

39. How often do you attend professional sporting events in one year?

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

40. How often do you go to the cinema?

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

41. a. Do you play a musical instrument?
YES NO

b. If yes, what kind(s):________________________

42. How many years of formal music training have you had?
______years of training

43. a. Are you a member of a music band?
YES NO

b. If yes, what genre of music do you play?
- ○ Hip-Hop
- ○ Jazz
- ○ Pop
- ○ Rock
- ○ Alternative
- ○ Punk
- ○ Reggae
- ○ Other, please specify:___________________

44. Did/do you have any of the following noisy hobbies? (Check all that apply)
- ○ Activities involving firearms
- ○ Carpentry
- ○ Ski/Sea-dooing
- ○ Motorcycling
- ○ Go-karting
- ○ None
- ○ Other, please specify: ____________________

45. What other past times do you engage in on a regular basis? (Check all that apply)
- ○ Reading
- ○ Watching Television
- ○ Playing computer/video games
- ○ Does not apply

46. Do you participate in any physical activities (i.e., exercise, sports)?
YES NO

47. a. Do you use hearing protection (e.g., earplugs)?
YES NO

b. If yes, in what situations do you most use hearing protection (Check all that apply)?

- o Attending concerts
- o Playing music
- o Sleeping
- o Studying
- o Operating machinery
- o Other, please specify: _____________

Portable Audio Devices

48. Do you have a portable audio device?

YES NO

IF NO THEN STOP HERE
IF YES THEN PLEASE GO ON

49. If so, what type of portable audio device do you own currently?

- o MP3 Player (Generic)
- o IPod
- o CD Player
- o Portable Cassette Player
- o Mini Disc
- o Cell Phone
- o PSP
- o Other, please specify:__________________

50. Currently, what type of output device do you use most to listen to this portable audio device?

- o Speakers (Portable)
- o Earbuds
- o Headphones
- o Car Stereo System
- o Home Stereo System
- o Other, please specify:___________________

51. Do you use the equalizer settings on your portable audio device?

YES NO

52. On a scale from 0-100 what level would you typically have the volume of your audio device set at while using headphones? (Please mark on scale.)

0 25 50 75 100

Actual Value:__________

53. On a scale from 0-100 what level would you typically have the volume of your audio device set at while using speakers? (Please mark on scale.)

0 25 50 75 100

Actual Value:__________

54. How often do you use your portable device?

________________ days per week

55. If yes, do you ever wear a portable audio device during any recreational activities?

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

56. In which situation do you use your portable audio device the most?

- o Walking
- o Studying
- o Leisure
- o Working
- o Working Out
- o Commuting
- o Other, please specify:_____________

57. When listening to a portable audio device, how long do you wear the headset in a single session?

_________ hours on average (use fractions if < 1 hour)

58. When in the following emotions/moods, rate how often you use your portable audio device

a. Happy

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

b. Sad

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

c. Angry

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

d. Upset

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

e. Frustrated

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

f. Bored

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

g. Anxious

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

h. Excited

Never		Occasionally			Frequently	
1	2	3	4	5	6	7

i. Depressed

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

j. Tired

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

k. Hungry

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

59. a. If you use ear buds, do you share or use other people's ear buds?

YES NO

b. If so how often?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

When wearing my portable audio device the way I wear it most often….

60. I hear the doorbell…

Never		Most of the time		Always		
1	2	3	4	5	6	7

61. I can have a phone conversation with…

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

62. I can hear someone speaking in a whisper…

Never		Most of the time		Always		
1	2	3	4	5	6	7

63. I can carry on a conversation with one other person when in a noisy place (e.g., party) …

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

64. I can carry on a conversation with one other person when in a quiet place (e.g., library) …

Difficulty		Some difficulty		No difficulty		
1	2	3	4	5	6	7

65. I feel that difficulty with my hearing limits or hampers my personal or social life…

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

66. Difficulty with my hearing upsets me…

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

67. I most often adjust the volume because of a…

a. Change in environment

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

b. Change in song genre

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

c. Change in song quality

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

d. Change in song loudness

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

e. Change in my mood

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

PLEASE COMPLETE THE FOLLOWING QUESTIONS ONLY IF YOU USE THE EQUALIZER SETTINGS.

68. How often do you adjust your equalizer settings?

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

69. I most often change equalizer settings because of a

a. Change in environment

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

b. Change in song genre

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

c. Change in song quality

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

d. Change in song loudness

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

e. Change in my mood

Never		Occasionally		Frequently		
1	2	3	4	5	6	7

ANSWER THE FOLLOWING QUESTION ONLY IF YOU OWN AN iPod

70. How often do you change the following equalizer settings?

a. Acoustic
Never Occasionally Frequently
1 2 3 4 5 6 7

b. Bass Booster
Never Occasionally Frequently
1 2 3 4 5 6 7

c. Bass Reducer
Never Occasionally Frequently
1 2 3 4 5 6 7

d. Classical
Never Occasionally Frequently
1 2 3 4 5 6 7

e. Dance
Never Occasionally Frequently
1 2 3 4 5 6 7

f. Deep
Never Occasionally Frequently
1 2 3 4 5 6 7

g. Electronic
Never Occasionally Frequently
1 2 3 4 5 6 7

h. Flat
Never Occasionally Frequently
1 2 3 4 5 6 7

i. Hip Hop
Never Occasionally Frequently
1 2 3 4 5 6 7

j. Jazz
Never Occasionally Frequently
1 2 3 4 5 6 7

k. Latin
Never Occasionally Frequently
1 2 3 4 5 6 7

l. Loudness
Never Occasionally Frequently
1 2 3 4 5 6 7

m. Lounge
Never Occasionally Frequently
1 2 3 4 5 6 7

n. Piano
Never Occasionally Frequently
1 2 3 4 5 6 7

o. Pop
Never Occasionally Frequently
1 2 3 4 5 6 7

p. R&B
Never Occasionally Frequently
1 2 3 4 5 6 7

q. Rock
Never Occasionally Frequently
1 2 3 4 5 6 7

r. Small Speakers
Never Occasionally Frequently
1 2 3 4 5 6 7

s. Spoken Word
Never Occasionally Frequently
1 2 3 4 5 6 7

t. Treble Booster
Never Occasionally Frequently
1 2 3 4 5 6 7

u. Treble Reducer
Never Occasionally Frequently
1 2 3 4 5 6 7

v. Vocal Booster
Never Occasionally Frequently
1 2 3 4 5 6 7

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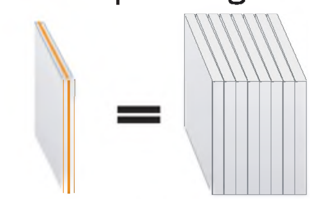

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Effect On Noise Emissions From Varying Distance Between Heat-Sink Fin To Cooling Fan Blade Tip

Colin Novak, Helen Ule and Robert Gaspar
Dept. of Mechanical, Automotive and Materials Engineering, University of Windsor, 401 Sunset Avenue, Windsor, Ontario

ABSTRACT

The challenge to deliver performance improvements in computer graphic cards has surpassed the ability of finned, passive, cooling devices to dissipate the heat generated by next generation graphics processing units (GPU). The dissipation rates required by these latest GPU designs can only be delivered by more complicated thermal management systems which often require forced air cooling of finned heat sinks. The concurrent challenge to the industry is to provide this cooling while minimizing the noise generated by these cooling fans. One of the fundamental mechanisms for the generation of fan noise is the dynamic force fluctuations on the fan blade and how these fluctuations interact with fixed irregularities such as adjacent cooling fins. This study investigates the effect on the acoustic emissions resulting from the variation of the distance between the fan blade tips and the heat sink fins. A discussion and comparison of the measured results will be presented using both traditional analysis techniques as well as psychoacoustic or sound quality metrics. It was found that a minimum distance between the blade and adjacent obstructions is desired in order to minimize excessive noise levels. The minimization of the noise emissions also had a desirable effect on the sound quality analysis.

SOMMAIRE

Le défi de constamment améliorer la performance des cartes vidéo ont surpassé la capacité d'appareils de refroidissement passif consistant d'ailerons pour disperser la chaleur produite par les processeurs graphiques (GPU) courantes. La dissipation exigée par ces GPU modernes peut être livré seulement par les systèmes de gestion thermiques plus compliqués qui exige souvent l'utilisation du refroidissement à air forcé de passer des ailerons. Le défi simultané qui presse l'industrie c'est de fournir ce refroidissement en minimisant le bruit produit par les ventilateurs utilisés. Un des mécanismes fondamentaux pour la génération de bruit de ventilateur est la variation de forces dynamiques sur la lame du ventilateur et comment ces variations interactent avec les irrégularités fixes telles que des ailerons adjacents. Cette étude examine l'effet sur les émissions acoustiques qui résultent de la variation de la distance entre les pointes des lames du ventilateur et les ailerons de dissipateur thermique. Une discussion et une comparaison des résultats mesurés seront présentées utilisant les techniques d'analyse traditionnelles de même que des métrques psychoacoustique, ou de qualité du son. Il a été trouvé qu'une distance minimum entre la lame et les obstructions adjacentes est désirée afin de minimiser le niveau de bruit. La minimisation des émissions de bruit avait aussi un effet désirable sur l'analyse de la qualité du son.

1. INTRODUCTION

There have been marked improvements in the performance of computer graphic cards which obviates the need for cooling fans that can dissipate the increased rate of heat generation in the graphic processing chip. The use of a high-performance cooling fan also has the negative effect of the introduction of a new source of noise. This further complicates the performance and compliance requirements faced by the GPU card manufacturer.

As a first rule of thumb, it is preferable to use large, slowly rotating fans as opposed to smaller but faster rotating fans. The slower the fan speed, the less acoustic emission will result. However, care must be taken to maintain required cooling capacities, thus the careful balancing act between fan size and speed. To further complicate the issue, the available space due to form factor restrictions within the computer chassis is often limited. The space limitation, thus, prevents the implementation of large, low speed, high flow rate fans. Consequently, there is often little choice but to use smaller, higher speed fans operating in order too provide the required cooling capacity. The need then is to control the increased noise emissions.

Two fundamental mechanisms associated with the operation of a cooling fan cause the generation of noise. Both of these are sources of dynamic force variations between the surface of the fan blade and the immediate surrounding air.

The first of these is random generated noise produced at the fan inlet. If the inlet flow is turbulent, a force results which is dependant on the influence of the lifting force to the angle of attack [1]. Acoustic models which predict the generated sound power of this force can be readily found in aeroacoustic literature. The resulting amplitude of this random noise generation is directly dependant on the level of inlet turbulence. As such, the inlet path must be kept as unobstructed as possible to minimize the creation of any turbulence.

The second source of noise generation is caused by spatially fixed irregularities which can produce a wake in either the inlet or outlet flow. In the case of computer cooling solutions, these irregularities can be the result of either the presence of heat sink fins or an improperly designed shroud. The resulting dynamic forces can contribute to strong tones which are usually found at the blade passage frequency.

To lessen the impact of these mechanisms of noise generation, one can either modify the blades of the fan or lessen the impact of the irregularities on the flow. The first can be achieved through the implementation of airfoil shaped fan blades or the use of a variable depth volute. In this study, the latter of the two fixes was used. Here, the effect of the distance between the fan blade tips and the heat sink fins of a relatively simple video card cooling solution has on the acoustic emissions is experimentally investigated.

2.0 ACOUSTICAL MEASURES

A brief discussion of the fundamental acoustic parameters used in this investigation is presented in this section.

To measure the effectiveness of either of the above approaches, traditional acoustic measuring techniques and metrics are used to quantify the acoustic emissions. These include the determination of the emitted sound pressure level at a given distance as well as the radiated sound power level. Sound pressure level will be presented as A-weighted decibels, or dBA.

While A-weighting is the traditional analytical approach and serves well to quantify the amplitude of acoustic emissions, it offers no insight to the perceived quality of the sound produced by the graphic card cooling fans. Given this, the acoustic product evaluation of computer graphic card cooling noise should include sound quality or psychoacoustic analysis. Sound quality, as a product attribute, can significantly affect the acceptability or desirability to the consumer. Therefore, in order to truly determine the full acoustic impact that an active cooling solution will have on the end user, the measurement of the applicable psychoacoustic metrics is warranted.

For this investigation, measurements of loudness will be presented. While somewhat similar to the A-weighting scale discussed above, loudness is an ISO standardized metric which is a more detailed representation of how loud a source is perceived as opposed to a simply reported sound pressure level. The human body, head and outer ear act as spatial and spectral filters on an acoustic signal. The inner ear also imparts nonlinear characteristics on this signal which are not incorporated in a simple sound pressure level measurement. Compensation for the effect of temporal processing and audiological masking effects of sounds across the frequency range can be realized through the application of the calculated loudness metric in an acoustic measurement. The units of loudness are given as sones.

Also determined in this study is prominent tone (PR). This psychoacoustic metric gives an objective measure of the prominence of a tonal component of a measured sound. This metric is computed according to ANSI S1.13-1995. The prominent tone is defined as the ratio of the power of the critical (frequency) band centred on the tone under investigation to the mean power of the two adjacent critical bands. A tone is said to be prominent if its PR exceeds 7 dB and is usually reported along with the frequency at which it is located. For this investigation, only the presence of prominence will be indicated.

The acoustic testing was done in compliance to ECMA-74, "Measurement of Airborne Noise Emitted by Information Technology and Telecommunication Equipment" [2] and ISO 7779, "Measurement of Airborne Noise Emitted by Information Technology and Telecommunication Equipment" [3]. To further comply with these acoustic measurement standards, ISO 3745, "Determination of Sound Power Levels of Noise Sources" [4], was followed for the measurement and calculation of the sound power level

As part of the investigation, an uncertainty analysis was also performed. It was found that the measurement procedure produced an uncertainty level of less than 1 dB for the sound pressure level based results.

3.0 THE EXPERIMENT

The fan-sink design used has a 53 mm diameter fan and operates in a manner where the intake air is drawn in through and across the cooling fins located at the end and one side of the sink and is expelled out through the top of the fan. For this design, the GPU and memory modules are located beneath the cooling fins. The sink plate and fins are constructed of copper and have a polycarbonate shroud on the top side to direct the air flow through the fins. Figure 1 is a photo of the assembled fan-sink design.

In order to investigate the effect of varying distance between the fan and fins, a movable fin design was devised. This fin module was radiused to the contour of the impeller and was designed to be positioned at six different distances, from the tip of the fan blades. The distances used were 1 mm, 4 mm, 7 mm, 10 mm, 13 mm and 16 mm. Figures 2 and 3 illustrate the disassembled fan mount and movable fin module. The fan was operated at each of the six distances at three different speeds, each representing minimum, medium and maximum operational speed. Under normal operating conditions, the air mover speed would be controlled and varied based on the thermal load.

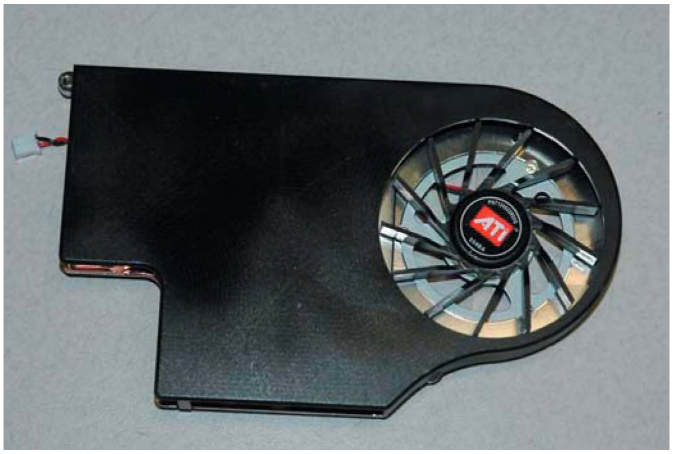

Figure 1. Assembled Video Card Fan-Sink Cooling Module.

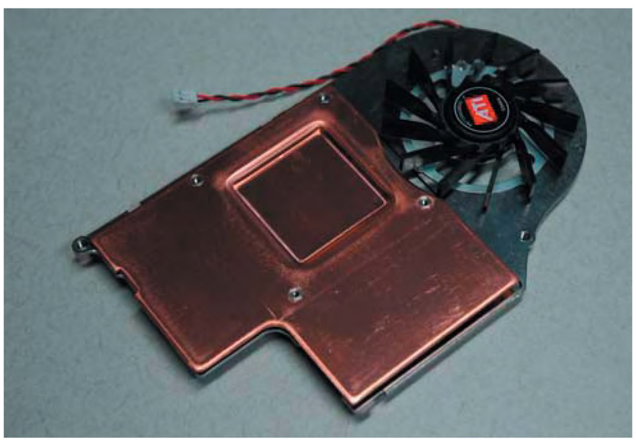

Figure 2. Disassembled Fan and Mounting Tray.

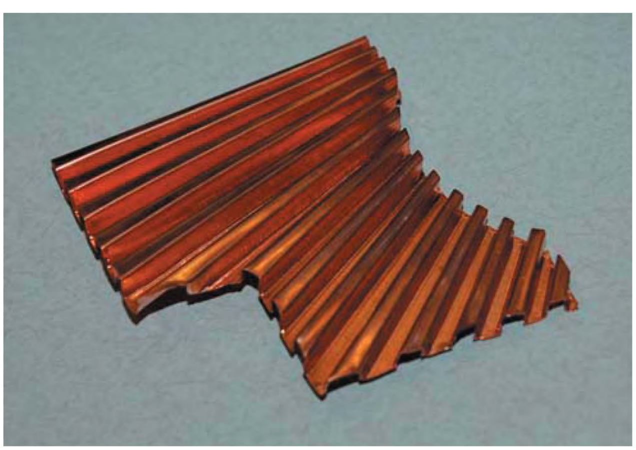

Figure 3. Movable Fin Module.

The acoustic measurements were conducted inside a hemi-anechoic room with an ambient noise level of approximately 17 dBA in the frequency range of interest. The determination of the sound power emission for each of the fan-fin distance variations involved the acquisition of ten free field sound pressure level measurements around the noise source in a specified hemi-spherical pattern which has a radius of 1 metre. The sound power level, reported as A-weighted Bells (BA) to comply with ECMA-74, is then calculated by taking a logarithmic average of these ten sound pressure levels and adjusting this to account for the surface area of the test sphere as well as atmospheric conditions and floor reflections.

Acquisition of the sound pressure levels and noise data used for the analysis of the psychoacoustic metrics were measured using a binaural head manikin at a distance of 0.5 metres. The manikin, which is shown in Figure 4, is a representation of a human head and torso with microphones mounted inside the artificial ear canals. The use of a binaural head for the acquisition data to be used for sound quality analysis is essential since the data is most representative of what would be perceived by an actual person.

Figure 4. Binaural Head Manikin used Acoustic Data Acquisition.

4.0 RESULTS AND DISCUSSION

Figure 5 illustrates the measured sound pressure level results of each fan-fin distance for each of the three measured speeds. An obvious pattern exists for each of the speeds. The sound pressure level is greatest for the 1 mm and 4 mm spacing between the fan tip and cooling fin. However, after the spacing is increased to 7 mm, the sound pressure level exhibits a rapid decline where it remains somewhat constant for subsequent larger spacings of 7 mm, 10 mm and 13 mm for this fan. For the 16 mm distance between the fan and fin another rapid decline in sound pressure level is realized. Inspection of Figures 6 and 7 for sound power level and Loudness respectively show very similar trends in the measured data.

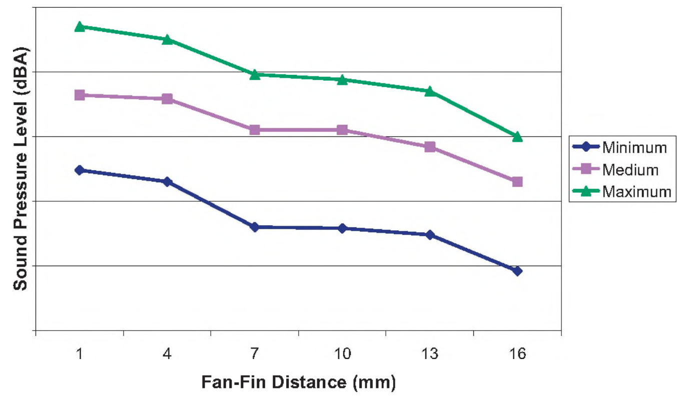

Figure 5. Measured Sound Pressure Level variation with distance.

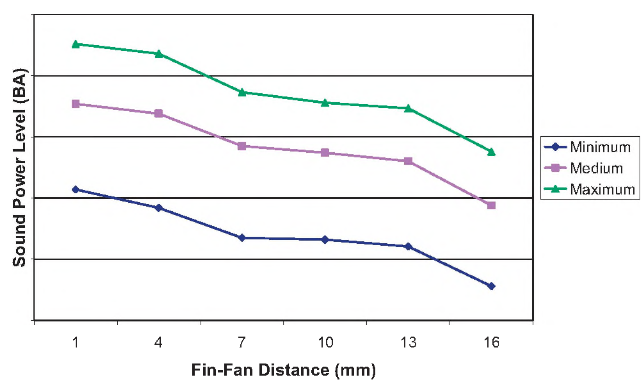

Figure 6. Calculated Sound Power Level variation with distance.

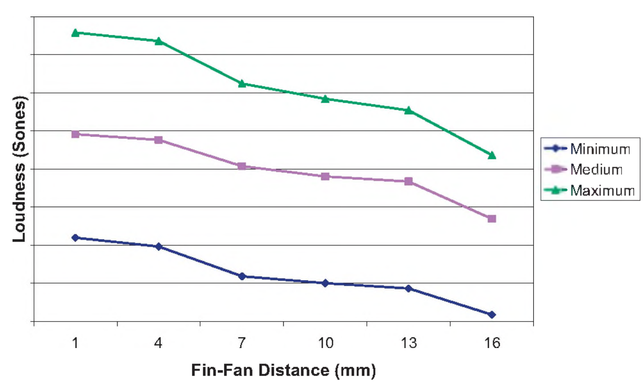

Figure 7. Calculated Loudness variation with distance.

For very close clearance spacing between the blade tip and cooling fin, it appears that a prominent mechanism of noise generation is present. This would account for the excess noise levels at the 1 mm and 4 mm clearance distances. Given that the length of cooling fins present on the intake side, it is a fairly safe assumption that the flow at the immediate exit would not be highly turbulent due to the flow straightening effect of the enclosed cooling fin channels. The noise generation would then be surmised to be the result of the presence of the spatially fixed irregularities in the very close proximity of the blade tips. This is further reinforced by the presence of prominent tones at the two closest distances as is shown in Table 1. What is not shown is that while some prominence was also detected at the 7 mm and 10 mm distances, the amplitudes of the tone were not as strong as the 1 mm and 4 mm cases. While this mechanism for noise generation can also be due to an interaction between the exit shroud and exhaust flow, this remained a constant for the entire study and therefore would play no role in the varying noise emissions. It should also be noted that the type of noise generation demonstrated here can at times also be controlled by other means. One way to accomplish this is through redesign of the fan blade. Examples of this include both the curvature of the blade as well an appropriate cross sectional profile. The latter of these two is more effective for larger diameter fans.

Table 1. Presence of Prominent Tone (Yes/No)

Distance (mm)	Minimum	Medium	Maximum
1	Yes	Yes	Yes
4	Yes	Yes	Yes
7	No	Yes	No
10	No	Yes	Yes
13	No	No	No
16	No	No	No

For the distances of 7 mm, 10 mm and 13 mm, an obvious reduction in noise emission is observed. Further, it is seen that the changes in noise level caused by the change in spacing are not very different from each other. It is suggested that the noise is created by a combination of both noise generation mechanisms. Inspection of Table 1 demonstrates a continued presence of prominent tones at the 7 mm and 10 mm distances. Therefore, the interaction of the cooling fins with the dynamic forces at the blade tips is still present. It should also be noted that, as the distance is increased, the flow straightening effect of the cooling fins is lessened. In other words, there is an increase in noise generation due to an increasing level of turbulence. From this it can be surmised that as the effect on the prominent tones is decreased with an increase in distance, this is offset by the noise generation due to increasing turbulence.

Once the cooling fins have been located at a distance of 15 mm from the fan, another drop in acoustic emissions is realized. It is suggested that at this distance, the effect of the fins as a fixed irregularity has been minimized and that the noise generation is now mostly due to the presence of turbulent flow.

5.0 SUMMARY AND CONCLUSIONS

Through the use of acoustic measurement and analysis techniques, it has been demonstrated that the generation and presence of acoustic emissions, for the particular cooling solution design considered, is the result of two mechanisms of noise generation. The first of these is due to the presence of turbulent flow and the second is from the creation of annoying prominent tones created by the presence of fixed irregularities in near proximity to the moving fan blades. From the presented results, it has been shown that a fine balancing act is necessary to minimize the noise emissions caused by these two mechanisms.

From the competing nature of these two affects, a simple solution derived from the location of the cooling fins with respect to the fan blade is not present. While it is obvious that a decrease in sound level is realized with increasing distance between the cooling fins and fan blades, a definitive decision of optimum distance can not be made in the absence of consideration of the required thermal performance of the heat removal system. Given this, other design considerations must also be considered for the control of noise. These include careful design of the fan rotor as well as the shroud used for flow directionality. However, the importance of clearance distance as one of the design considerations has been incontrovertibly demonstrated in this study.

ACKNOWLEDGEMENTS

Acknowledgement and thanks is extended to the Ontario Centres of Excellence (OCE) for their financial assistance, which in part made this research possible.

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4. International Standard ISO 3745: Acoustics-Determination of Sound Power Levels of Noise Sources-Precision Methods for Anechoic and Semi-Anechoic Rooms, 1st Edition, 1977.

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Histoire de l'acoustique sous-marine de Pierre Juhel Coéd. Adapt/Vuibert ; ISBN: 2-909680-68-1; 192 p ; 2005; 25 euros

La communauté scientifique en acoustique sous-marine étant relativement restreinte, les livres en français dans cette spécialité sont donc peu nombreux. Puisque l'auditoire de ce livre est limité, on le lit avec d'autant plus de plaisir.

Le contenu est séparé en quatre chapitres. Le premier chapitre, intitulé « Histoire des techniques de production et de réception des sons dans la mer », est la pièce de résistance. Le chapitre couvre les systèmes principaux qui ont permis à l'acoustique sous-marine de se développer, des premiers sondeurs acoustiques aux sonars actifs, en passant par les développements majeurs en transducteurs piézo-électriques. La période couverte commence à l'Antiquité, et se termine vers les années 60; le matériel récent est relativement mince. L'histoire des sondeurs est particulièrement complète, et inclut une chronique descriptive de la division des camps sur le choix de la fréquence des premiers sondeurs – les fréquences audibles ou les ultrasons – provenant des différents instruments disponibles. L'évolution des cloches sous-marines est aussi bien présentée, ainsi que l'usage conjoint de la source sous-marine avec le signal radio pour la navigation maritime.

Le deuxième chapitre « Les aléas de la propagation des sons dans l'océan » est une présentation des conquêtes scientifiques qui ont permis la dépendance croissante de la communauté scientifique sur l'acoustique sous-marine, tels les principes de propagation du son dans l'eau, incluant la détermination de la vitesse du son, l'absorption marine et l'équation des ondes. Le chapitre précédent portant surtout sur les sources acoustiques et les systèmes sonar, ce chapitre concerne les développements physiques et mathématiques qui sont à la base des systèmes sonar. Le traitement, par contre, est descriptif et non mathématique, la seule équation présentée étant celle des ondes.

Le troisième chapitre « Des outils indispensables pour la connaissance et l'exploitation de l'océan » décrit pour l'amateur certaines applications de l'acoustique sous-marine: la sismique et l'océanographie, incluant la cartographie, la tomographie, et la détection halieutique. Finalement, le dernier chapitre « L'océan n'est pas dans le monde du silence » décrit l'inquiétude croissante de l'impact de la pollution acoustique sur l'environnement. Ce dernier chapitre semble plutôt hors contexte avec sa touche un peu politique. L'auteur discute du problème, mais ne mentionne pas les efforts déployés par la communauté pour trouver des solutions à ce problème.

Le livre de Pierre Juhel est très intéressant. Le style est aisé et le texte est écrit en fonction du public scientifique. Il est donc facile à lire et plein de détails intéressants qui rendent le texte aussi attrayant au lecteur amateur qu'au lecteur avisé.

Ma principale critique est que le titre est mal choisi : il devrait s'intituler « Histoire française de l'acoustique sous-marine ». Le texte contient un préjugé favorable omniprésent pour les inventions et les applications françaises, et les autres pays sont souvent représentés comme des concurrents. Par exemple, l'auteur passe plus de temps à expliquer pourquoi la France n'a pas développé son propre réseau de cloches sous-marines qu'à discuter la croissance rapide de cette technologie dans les autres pays. Cette attitude s'amplifie au Chapitre 3, où les exemples de systèmes, manufacturiers ou utilisateurs, sont en grande majorité d'origine française.

Certains pardonneront certes l'auteur en raison de sa recherche approfondie du sujet. Il est clair que l'auteur a une expérience variée en acoustique sous-marine et qu'il a pratiqué dans trois domaines différents. Le livre est féru de photos et de diagrammes datant d'aussi loin que les années 1800. Le premier chapitre se concentre sur les appareils et les systèmes, et les diagrammes qui accompagnent les descriptions rendent le texte très intéressant.

Ce premier chapitre est vraiment la partie la plus recherchée de l'ouvrage. Bien que l'auteur n'ait pas travaillé dans le contexte militaire, il décrit bien l'évolution des premiers sonars pour la détection de sous-marins. Ce qui n'est peut-être pas mis en évidence est la profondeur de l'impact des recherches militaires sur l'évolution de ce champ de recherche. Ceci est surtout tangible pour les années suivant la Deuxième guerre mondiale. L'évolution des sous-marins durant la guerre froide et leur baisse de niveau acoustique a déclenché d'énormes progrès en sonar actif, traitement de signal, modèles de propagation acoustique, et développements de sources et récepteurs à réseaux. Cette période et ces développements ne sont pas traités dans cet ouvrage. Les chapitres 2 et 3 sont peut-être trop généraux pour ceux qui travaillent dans le domaine de l'acoustique sous-marine, et pour cette raison j'aurais personnellement préféré que l'auteur approfondisse le premier chapitre au détriment des chapitres subséquents.

Le livre sera plus apprécié de l'ingénieur et de l'expérimentaliste que du théoricien, mais tous y trouveront quelque chose à apprécier. Je le recommande donc aux lecteurs français.

Par Francine Desharnais
DRDC Atlantic, Darmouth

Acoustics and Psychoacoustics, Third Edition
D.M. Howard, J. Angus, Pages 411
Focal Press 2006 – ISBN 13: 978-0-24-051995-1: $49.95 USD

Acoustics and Psychoacoustics is an easy to read text which is claimed to be targeted to a rather large professional demographic with little previous background in acoustics. The book is suitable for both students, especially those taking music, as well as those who simply require a practical understanding of fundamental acoustics, musical sound generation and the human perception of sound. The book is divided into 7 chapters of which the contents of these are: 1) Introduction to Sound; 2) Introduction to Hearing; 3) Notes and Harmony; 4) Acoustic Model for Musical Instruments; 5) Hearing Timbre and Deceiving the Ear; 6) Hearing Music in Different Environments; and 7) Processing Sound Electronically. Also included are five appendices and an audio compact disk.

Chapter 1 introduces the reader to the fundamental concepts of acoustics, its generation, calculation methods and theory of propagation. The descriptions given in each subsection are written in a very simple and easy to comprehend manner which makes it particularly suitable for those without a technical background. By the same token, the brevity results in a lack of detail given for each subtopic area. In fact, the topics covered in this first chapter can very often entail the contents of an entire textbook on acoustics.

Chapter 2 familiarizes the reader to the functioning of the human ear and does so with just the right amount of detail. The manner in which the authors explain the concepts of critical bands is done very well and is a section which I believe all engineers working in acoustics should read. Accompanying this section is a demonstration on the enclosed CD which somewhat reinforces the idea of critical bands but more interestingly it is a very good example of the fundamental concept of modulation which could nicely lead into a discussion of the psychoacoustic metrics of roughness and fluctuation strength. Discussion is also given to the ideas of loudness, mechanisms of hearing loss and human perception.

Chapter 3 seems to be intended for musicians and/or students of music. An attempt is made to relate the physical phenomenon of acoustics to ideas and concepts, familiar to the musician. While somewhat unique in its presentation, this chapter may lose the interest of the reader from other backgrounds including engineers and those with medical interests as suggested by the rear cover of the text. It is this alienation that I believe would leave many readers looking for more.

Much of what was said of chapter 3 can also be applied to chapter 4. Very interesting descriptions of how various musical instruments produce sound are given in technical detail. While this information can be an interesting read for some, a lack of general applicability is apparent. An exception to this would be the descriptions of the generation of sounds through the mechanisms of speaking and the singing voice. As was expressed of the descriptions of human hearing from chapter 2, this is something that all acousticians should have at least a peripheral understanding of.

Chapter 5 really begins to explain the appropriateness of haveing "psychoacoustics" in the book's title. A very thorough discussion with applications on hearing timbre is given. The material presented can be very closely related to discussions on engineering sound quality, although again, a musician's perspective is taken in this book. The technical content is reinforced with several figures including spectrograms of measured signals. The downside of this is the fact that this book has no colour content which results in a loss of impact in the use of these figures. This chapter has perhaps the richest content of psychoacoustics and includes several important concepts including time and frequency masking of sound, however, the chapter finishes with illustrative examples which again would be relevant only to musicians leaving everyone else unsatisfied.

Chapter 6 attempts to describe the concepts of architectural acoustics by using examples of listening environments. The manner in which the material is presented is well suited to the needs of an acoustical engineer. Very good details of the ideas of sound propagation, absorption, reflection and decay are given. The fundamentals of room acoustics are also given in an easy to understand, yet still detailed enough, manner. Some information on sound isolation is also given but not in very great detail which leaves the reader again wanting more.

The final chapter in the textbook deals with the electronic processing of sound including filtering, equalization and artificial reverberation. The information here is directed more to recording and music technology engineers. However, the sections are very brief and do not provide much more information than what the target audience would already have. The chapter also does not lend well to the title of the text of Acoustics and Psychoacoustics and might have been better reserved for a book on audio and multimedia engineering.

This book struggles to find a target audience. Even though it claims to be applicable to students of music, engineering, communications and even ear surgeons, it at times tries to do too much at once. The content is for the most part easy to follow and I think should be summed up to be a textbook for student musicians. The book, however, does provide enough peripheral material to the professor or acoustical engineer who would like to round out his or her collection of acoustical textbooks particularly if they are looking for material presented that is different than a conventional acoustics textbook.

Colin Novak, Ph. D., P. Eng.
University of Windsor, Windsor, ON, Canada
novak1@uwindsor.ca

NEWS / INFORMATIONS

CONFERENCES

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 stevenb@aciacoustical.com

2007

13-17 March. Spring Meeting of the Acoustical Society of Japan., Tokyo, Japan. Web: www.asj.gr.jp/index-en.html

15-17 March. AES 30th International Conference on Intelligent Audio Environments. Saariselka, Finland. Web: www.aes.fi/aes30

19-22 March. Meeting of the German Acoustical Society (DAGA2007). Stuttgart, Germany. Web: www.daga2007.de

09-12 April. International Congress of Ultrasonics (2007 ICU). Vienna, Austria. Web: www.icultrasonics.org

10-12 April. 4th International Conference on Bio-Acoustics. Loughboro, UK. Web: www.ioa.org.uk

17-20 April. IEEE International Congress on Acoustics, Speech, and Signal Processing (IEEE ICASSP 2007). Honolulu, HI, USA. Web: http://www.icassp2007.org

24-25 April. Institute of Acoustics (UK) Spring Conference. Cambridge, UK. Web: www.iap.org.uk/viewupcoming.asp

16-20 May: IEEE International Conference on Acoustics, Speech, and Signal Processing (IEEE ICASSP 2007). Honolulu, HI, USA. Web: www.icassp2007.org

01-03 June. 2nd International Symposium on Advanced Technology of Vibration and Sound (VS Tech 2007). Lanzhou, China. Web: www.jsme.or.jp/dmc/meeting/vstech2007.pdf

03-07 June. 11th International Conference on Hand-Arm Vibration. Bologna, Italy. Web: www.associazioneitalianadiacustica.it/HAV2007/index.htm

04-06 June. Japan-China Joint Conference on Acoustics 2007. Sendai, Japan. Web: www.asj.gr.jp/eng/index.html

04-08 June: 153rd Meeting of the Acoustical Society of America. Salt Lake City, Utah, USA. Web: www.asa.aip.org

18-21 June. Oceans07 Conference. Aberdeen, Scotland. Web: www.oceans07ieeeaberdeen.org

25-29 June. 2nd International Conference on Underwater Acoustic Measurements: Technologies and Results. Heraklion, Crete, Greece. Web: www.uam2007.gr

02-06 July. 8th International Conference on Theoretical and Computational Acoustics. Heraklion, Crete, Greece. Web: www.iacm.forth.gr/~ictca07

03-07 July. 1st European Forum on Effective Solutions for Managing Occupational Noise Risks. Lille, France. Web: www.noiseatwork.eu

9-12 July: 14th International Congress on Sound and Vibration (ICSV14). Cairns, Australia. Email: n.kessissoglou@unsw.edu.au

26-29 August: Inter-noise 2007. Istanbul, Turkey. Web: www.internoise2007.org.tr

27-31 August: Interspeech 2007. E-mail: conf@isca-speech.org

CONFÉRENCES

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 stevenb@aciacoustical.com

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27-31 août: Interspeech 2007. E-mail: conf@isca-speech.org

2-7 September 19th International Congress on Acoustics (ICA2007), Madrid Spain. (SEA, Serrano 144, 28006 Madrid, Spain; Web:www.ica2007madrid.org

9-12 September: ICA2007 Satellite Symposium on Musical Acoustics (ISMA2007). Barcelona, Spain. Web: www.isma2007.org

9-12 September: ICA2007 Satellite Symposium on Room Acoustics (ISMA2007). Sevilla, Spain. Web: www,isra2007.org

17-19 September. 3rd International Symposium on Fan Noise. Lyon, France. Web: www.fannoise2007.org

19-21 September. Autumn Meeting of the Acoustical Society of Japan. Kofu, Japan. Web: www.asj.gr.jp/index-en.html

24-28 September. XIX Session of the Russian Acoustical Society. Nizhny Novgorod, Russia. Web: www.akin.ru

22-24 October. Noise-Con 2007. Reno, Nevada, USA. Web: www.inceusa.org/nc07/index.asp

November 27 - December 02: 154th Meeting of the Acoustical Society of America. New Orleans, LA, USA. Web: www.asa.aip.org

2008

29 June - 04 July: Joint Meeting of European Acoustical Association, Acoustical Society of America, and Acoustical Society of France. Paris, France. Web: www.sfa.asso.fr/en/index.htm

7-10 July: 18th International Symposium on Nonlinear Acoustics (ISNA18). Stockholm, Sweden. E-mail: benflo@mech.kth.se

27-30 July. Noise-Con 2008. Dearborn, MI, USA.

28 July - 1 August: 9th International Congress on Noise as a Public Health Problem. Mashantucket, Pequot Tribal Nation, (CT, USA). Web: www.icben.org

22-26 September: Interspeech 2008 - 10th ICSLP, Brisbane, Austrailia. Web: wwwinterspeech2008.org

01-05 November. IEEE International Ultrasonic Symposium. Beijing, China. Web: www.ieee-uffa.org/ulmain.asp?page=symposia

2010

23-27 August: International Confress on Acoustics 2010. Sydney, Australia. Web: www.acoustics.asn.au

2-7 septembre 19e Congrès international sur l'acoustique (ICA2007), Madrid Spain. (SEA, Serrano 144, 28006 Madrid, Spain; Web: www.ica2007madrid.org

9-12 septembre: ICA2007 Satellite Symposium sur Musical Acoustics (ISMA2007). Barcelona, Spain. Web: www.isma2007.org

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2010

23-27 août: International Confress sur Acoustics 2010. Sydney, Australia. Web: www.acoustics.asn.au

NEWS

We want to hear from you! If you have any news items related to the Canadian Acoustical Association, please send them. Job promotions, recognition of service, interesting projects, recent research, etc. are what make this section interesting.

EXCERPTS FROM "*SCANNING THE JOURNALS*", IN *ECHOS*, ASA

Two papers presented at the 2005 **Australian Acoustical Society** conference are reprinted in the December issue of *Acoustics Australia*. A paper entitled "Learning Acoustics through the Boundary Element Method: An Inexpensive Graphical Interface and Associated Tutorials" was awarded the President's Prize for the best technical paper. The Boundary Element Method (BEM), the paper points out, is particularly useful for analyzing sound radiation and acoustic scattering problems. The other paper "Acoustic Systems in Biology: From Insects to Elephants" discusses the physical principles in the sound production and hearing of a variety of creatures. The dominant frequencies used for communication by a large range of air-breathing animals is nearly proportional to the body mass raised to the -0.4 power.

Tunable nanoresonators constructed from telescoping multiwalled carbon nanotubes (MWNTs) are described in the 2 June issue of *Physical Review Letters*. Such resonators, with their low masses, low force constants, and high resonant frequencies, are capable of weighing single bacteria, detecting single spins in magnetic resonance systems, and even probing quantum mechanics in macroscopic systems. In the device, a specially prepared MWNT is suspended between a stationary contact and a mobile piezocontrolled electrode. Varying the length of the nanotube beam through the controlled telescoping of the inner nanobtube core from the outer nanotube shell tunes its resonant frequency.

Higher sound clarity is obtained in **classrooms** when sound diffusers are applied to rear walls and ceilings rather than side walls, according to a paper in the *Proceedings of WESPAC IX*. However, absorbers increase sound clarity even more effectively with smaller area in comparison with diffusers.

A protein associated with a disorder that causes **deafness** and blindness may hold a key to one of the foremost mysteries of hearing, according to a paper in the June 28 issue of *Journal of Neuroscience*. Scientists have identified protocadherin-15 as a likely player in the conversion of sound into electrical signals. The findings will not only provide insight into how hearing takes place at the molecular level, but may also help us explain why some people temporarily lose their hearing after exposure to loud noise but regain it a day or two later. The protein is referred to as the "tip-link antigen" (TLA) because it induces the production of special antibodies which bind to the protein at the stereocilia tips in the cochlea. Using mass spectrometry, the researchers analyzed the makeup of the TLA and found two peptide sequences that match up to key segments of the protein protocadherin-15 in humans, mice and chickens.

For any **touch, sound, or image** to be perceived, our senses have to activate neurons at the center of the brain, in the thalamus. According to a paper in the 16 June issue of *Science*, the weak neurons along the thalamocortical pathway may find strength through amplification within cortical layer 4. Researchers were able to measure excitatory electrical activity generated in a single cortical neuron in a mouse cortex by a single thalamic neuron.

Several interesting effects are reported when **cavitation bubbles** are generated inside water drops in microgravity, according to a paper in the 1 September issue of *Physical Review Letters* (**97**, 094502). Toroidally collapsing bubbles generate two liquid jets escaping from the drop, and the "splash jet" discloses a remarkable broadening. Shock waves induce a strong form of secondary cavitation due to the particular shock wave confinement, which offers a way to estimate integral shock wave energies in isolated volumes. Bubble lifetimes in drops are shorter than in extended volumes in remarkable agreement with herein derived corrective terms for the Rayleigh-Plesset equation. These observations, made on board an aircraft flying in parabolic arcs to create near-weightless conditions, would be difficult, if not impossible, in normal gravity.

A mathematical system for organizing the 12 **tones of the Western scale** that makes use of a topological structure called an orbifold is described in the 7 July issue of *Science*. Chords are points in the topological space, and the segments connecting them indicate how chords progress. The work described in this paper is part of an ambitious project to characterize musical composition in great generality by means of mathematical music theory.

Language is largely symbolic, but how we say something can be as important as what we say, according to an article in the 21 July issue of *Science Now Daily News*. Twenty four college students were asked to describe a dot moving across a screen. The students were told to use one of two sentences: "It is going up" or "It is going down." The team found that when students described the dots going up, the pitch of their voice was, on average, 6 hertz higher than that of those describing the dot going down. The same thing happened when another 24 students read the sentences from a computer screen, indicating people change the sound of their voice according to directional information contained within words. Listeners readily caught these cues.

To a person who suffers from **hyperacusis**, even the sound of their own voice can be intolerable, according to an article in the 15 July issue of *New Scientist*. Nobody knows exactly what causes hyperacusis, but it can be brought on by head injuries, exposure to extremely loud sounds, Lyme disease or autism. It has also been linked with tinnitus. The main question puzzling researchers is whether it results from structural damage to the ear or flaws in the way the brain interprets sound signals. The most successful treatment is a therapy using gradual desensitization with "pink noise," sound in which the amplitude decreases with increasing frequency.

Two-way communication over gas pipelines using multicarrier **modulated sound waves** is the topic of a paper in the July issue of *Acoustical Science and Technology*. Conventional acoustic communication technology is limited by the effect of reverberant signals, but the use of multi-carrier frequencies which change cyclically avoids this. Using this method, a transmission rate of 3840 bps was achieved.

The Canadian Acoustical Association
L'Association Canadienne d'Acoustique

PRIZE ANNOUNCEMENT • ANNONCE DE PRIX

A number of prizes and subsidies are offered annually by The Canadian Acoustical Association. Applicants can obtain full eligibility conditions, deadlines, application forms, past recipients, and the names of the individual prize coordinators on the CAA Website (http://www.caa-aca.ca). • Plusieurs prix et subventions sont décernés à chaque année par l'Association Canadienne d'Acoustique. Les candidats peuvent se procurer de plus amples renseignements sur les conditions d'éligibilités, les échéances, les formulaires de demande, les récipiendaires des années passées ainsi que le nom des coordonnateurs des prix en consultant le site Internet de l'ACA (http://www.caa-aca.ca).

Next deadline: Shaw, Bell, Fessenden, Eckel and Hétu Prizes: **30 April 2007**
Prochaine échéance: Prix Shaw, Bell, Fessenden, Eckel et Hétu: **30 Avril 2007**

EDGAR AND MILLICENT SHAW POSTDOCTORAL PRIZE IN ACOUSTICS • PRIX POST-DOCTORAL EDGAR AND MILLICENT SHAW EN ACOUSTIQUE

$3,000 for full-time postdoctoral research training in an established setting other than the one in which the Ph.D. was earned. The research topic must be related to some area of acoustics, psychoacoustics, speech communication or noise. • $3,000 pour une formation recherche à temps complet au niveau postdoctoral dans un établissement reconnu autre que celui où le candidat a reçu son doctorat. Le thème de recherche doit être relié à un domaine de l'acoustique, de la psycho-acoustique, de la communication verbale ou du bruit.

ALEXANDER GRAHAM BELL GRADUATE STUDENT PRIZE IN SPEECH COMMUNICATION AND HEARING • PRIX ÉTUDIANT ALEXANDRE GRAHAM BELL EN COMMUNICATION VERBALE ET AUDITION

$800 for a graduate student enrolled at a Canadian institution and conducting research in the field of speech communication or hearing • $800 à un(e) étudiant(e) inscrit(e) au 2e ou 3e cycle universitaire dans une institution canadienne et menant un projet de recherche en communication verbale ou en audition.

FESSENDEN GRADUATE STUDENT PRIZE IN UNDERWATER ACOUSTICS • PRIX ÉTUDIANT FESSENDEN EN ACOUSTIQUE SOUS-MARINE

$500 for a graduate student enrolled at a Canadian institution and conducting research in underwater acoustics or in a branch of science closely connected to underwater acoustics. • $500 à un(e) étudiant(e) inscrit(e) au 2e ou 3e cycle universitaire dans une institution canadienne et menant un projet de recherche en acoustique sous-marine ou dans une discipline reliée à l'acoustique sous-marine.

ECKEL GRADUATE STUDENT PRIZE IN NOISE CONTROL • PRIX ÉTUDIANT ECKEL EN CONTRÔLE DU BRUIT

$500 for a graduate student enrolled at a Canadian institution and conducting research related to the advancement of the practice of noise control. • $500 à un(e) étudiant(e) inscrit(e) au 2e ou 3e cycle universitaire dans une institution canadienne et menant un projet de recherche relié à l'avancement des pratiques en contrôle du bruit.

RAYMOND HÉTU UNDERGRADUATE PRIZE IN ACOUSTICS • PRIX ÉTUDIANT RAYMOND HÉTU EN ACOUSTIQUE

One book in acoustics of a maximum value of $150 and a one-year subscription to *Canadian Acoustics* for an undergraduate student enrolled at a Canadian institution and having completed, during the year of application, a project in any field of acoustics or vibration. • Un livre sur l'acoustique d'un montant maximal de 150 $ et un abonnement d'un an à la revue *Acoustique Canadienne* à un(e) étudiant(e) inscrit(e) dans un programme au 1er cycle universitaire dans une institution canadienne et qui a réalisé, durant l'année de la demande, un projet dans le domaine de l'acoustique ou des vibrations.

CANADA-WIDE SCIENCE FAIR AWARD • PRIX EXPO-SCIENCES PANCANADIENNE

$400 and a one-year subscription to *Canadian Acoustics* for the best project related to acoustics at the Fair by a high-school student • $400 et un abonnement d'un an à la revue *Acoustique Canadienne* pour le meilleur projet relié à l'acoustique à l'Expo-sciences par un(e) étudiant(e) du secondaire.

DIRECTORS' AWARDS • PRIX DES DIRECTEURS

One $500 award for the best refereed research, review or tutorial paper published in *Canadian Acoustics* by a student member and one $500 award for the best paper by an individual member • $500 pour le meilleur article de recherche, de recensement des travaux ou d'exposé didactique arbitré publié dans l'*Acoustique Canadienne* par un membre étudiant et $500 pour le meilleur article par un membre individuel.

STUDENT PRESENTATION AWARDS • PRIX POUR COMMUNICATIONS ÉTUDIANTES

Three $500 awards for the best student oral presentations at the Annual Symposium of The Canadian Acoustical Association. • Trois prix de $500 pour les meilleures communications orales étudiant(e)s au Symposium Annuel de l'Association Canadienne d'Acoustique.

STUDENT TRAVEL SUBSIDIES • SUBVENTIONS POUR FRAIS DE DÉPLACEMENT POUR ÉTUDIANTS

Travel subsidies are available to assist student members who are presenting a paper during the Annual Symposium of The Canadian Acoustical Association if they live at least 150 km from the conference venue. • Des subventions pour frais de déplacement sont disponibles pour aider les membres étudiants à venir présenter leurs travaux lors du Symposium Annuel de l'Association Canadienne d'Acoustique, s'ils demeurent à au moins 150 km du lieu du congrès.

UNDERWATER ACOUSTICS AND SIGNAL PROCESSING STUDENT TRAVEL SUBSIDIES • SUBVENTIONS POUR FRAIS DE DÉPLACEMENT POUR ÉTUDIANTS EN ACOUSTIQUE SOUS-MARINE ET TRAITEMENT DU SIGNAL

One $500 or two $250 awards to assist students traveling to national or international conferences to give oral or poster presentations on underwater acoustics and/or signal processing. • Une bourse de $500 ou deux de $250 pour aider les étudiant(e)s à se rendre à un congrès national ou international pour y présenter une communication orale ou une affiche dans le domaine de l'acoustique sous-marine ou du traitement du signal.

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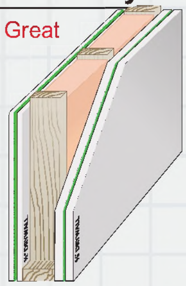

Soundboard

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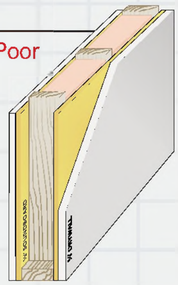
46 STC Rating
32 OITC



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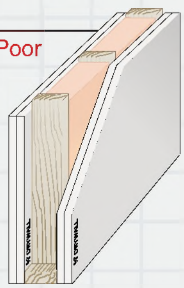

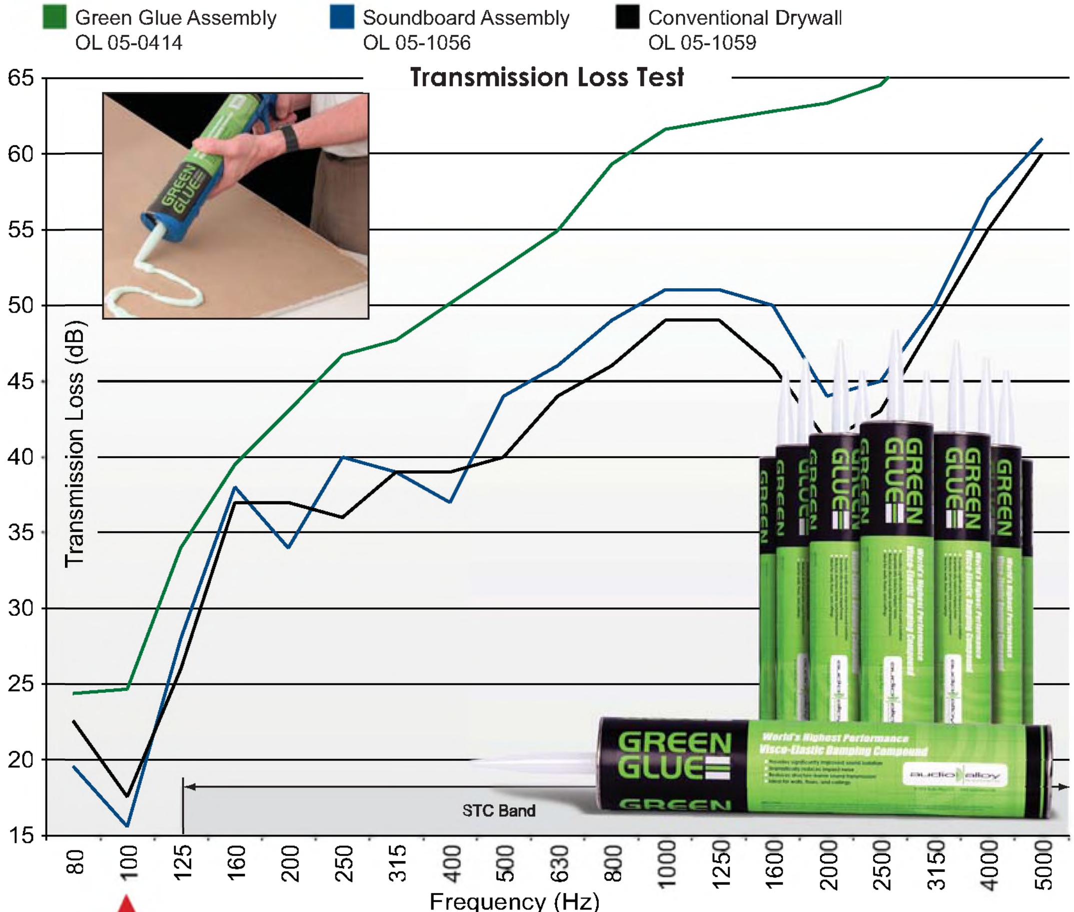

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Canadian Acoustical Association Annual Conference

2007

CONCORDIA UNIVERSITY
Faculty of Engineering and Computer Science, Montreal, Canada
October 9-12, 2007

FIRST ANNOUNCEMENT

The Canadian Acoustical Association (CAA) Annual conference attracts researchers in various fields of Acoustical Sciences and Engineering, and Auditory Perception. The theme for the 2007 conference will be **AEROACOUSTICS,** befitting the reputation of Montreal as the "Aerospace Capital of the World". There will be three days of parallel sessions on diverse areas of acoustics, including some special sessions on emerging topics as well as an interesting array of exhibits showcasing acoustical applications and products.

Venue of the Conference
The conference will be held in the attractive new Engineering and Visual Arts Complex of the Concordia University in downtown Montreal. CAA conferences are always an opportunity to inform ourselves on the state of acoustical research, new developments, emerging topics. It is also an opportunity to meet our friends and to make new ones over a coffee during the conference, or over a drink after the sessions. The participants can enjoy Montreal filled with colors of Fall in the backdrop of comfortable October weather. In addition to the technical sessions, you are invited to attend the industrial exhibits of acoustic products, and a banquet.

Goals
- To provide a national and international forum for communicating most recent advances in the field of acoustics and auditory perception.
- To provide an opportunity for information exchange among delegates from industry, research labs and academia.

Special Sessions
Several special sessions will be organized that will include invited and contributed papers in the areas of Aeroacoustics, Building Acoustics, Musical Acoustics, Audiology/Bioacoustics, Noise Control, Speech Communication, Instrumentation & Signal Processing in Acoustics, Acoustic Standards and Environmental Acoustics, Thermoacoustics, Electroacoustics, Photo Acoustics, Micro/Nano Acoustics. If you are interested in organizing a special session, please contact K. Siddiqui, the Technical Chair (technical-chair@caa-aca.ca).

Exhibits & Sponsors
During the conference there will be an interesting array of exhibits of acoustical products and services. The exhibit area will also be the central coffee break area. Please contact M. Packirisamy, the exhibit coordinator (exhibits@caa-aca.ca) for exhibitor information and sponsorship of various aspects of this conference.

Student Participation
Student members enrolled in Canadian universities may also enter into a competition for the best student presentation award. The student presenters may also get travel subsidy if they submit the request before 15th September 2007. This subsidy is subject to the availability of funds.

Important Dates
22nd May 2007 - Submission date for 2 page paper
3rd July 2007 - Notice of acceptance
15th Sept. 2007 - Early registration deadline
9th to 12th Oct. 2007 - Conference dates

Selected papers will be published in a special issue of the journal *Canadian Acoustics.*

Organizing Committee
Conference Chair: R. Bhat conference@caa-aca.ca
Technical Chair : K. Siddiqui technical-chair@caa-aca.ca
Exhibits Chair: M. Packirisamy exhibits@caa-aca.ca
Publications Chair: S. Narayanswamy conf-web@caa-aca.ca
Logistics Chair: W. Xie logistics-technical@caa-aca.ca
Treasurer: Z. Chen conf-treasurer@caa-aca.ca

Conference URL: http://users.encs.concordia.ca/~caa-2007

Association canadienne d'acoustique
Congrès annuel

2007

UNIVERSITÉ CONCORDIA
Faculté de génie et d'informatique
Montréal, Canada
9-12 octobre 2007

PREMIÈRE ANNONCE

L'Association canadienne d'acoustique (ACA) regroupe des chercheurs dans différents domaines des sciences et du génie acoustiques et de la perception auditive. Le thème du Congrès 2007 porte sur l'AÉROACOUSTIQUE en harmonie avec la réputation de Montréal comme « capitale mondiale de l'aérospatiale ». Parallèlement au Congrès se tiendront trois journées sur divers sujets touchant l'acoustique, dont des séances spéciales sur les domaines émergents et des expositions sur les applications et les produits acoustiques.

Lieu du Congrès

Le Congrès se déroulera au centre-ville de Montréal, dans le très élégant pavillon intégré Génie, informatique et arts visuels de l'Université Concordia. Les congrès de l'ACA permettent aux participants de se tenir informés de l'état des recherches en acoustique (percées, domaines émergents, etc.). C'est aussi une occasion de retrouver de vieux amis ou de se faire de nouvelles connaissances autour d'un café entre deux séances de travail ou le soir, après le Congrès. Les participants pourront admirer les couleurs automnales de Montréal sous la température encore clémente du mois d'octobre. En plus d'assister à des ateliers techniques, ils pourront visiter les expositions de l'industrie sur les produits acoustiques et assister à un banquet.

Ojectifs

- Offrir une tribune nationale et internationale pour partager les plus récentes avancées dans le domaine de l'acoustique et de la perception auditive.
- Permettre l'échange d'information entre les représentants de l'industrie, des laboratoires de recherche et du milieu universitaire.

Séances spéciales

Des séances spéciales donneront lieu à des communications sollicitées par le comité d'organisation ou proposées par les membres. Elles porteront notamment sur l'aéroacoustique, l'acoustique des bâtiments, l'acoustique musicale, l'audiologie/bioacoustique, le contrôle acoustique, la communication verbale, l'instrumentation et le traitement des signaux en acoustique, les normes acoustiques et l'acoustique environnementale, la thermoacoustique, l'électroacoustique, la photoacoustique, la microacoustique et la nanoacoustique. Pour organiser une séance spéciale, prière de contacter M. K. Siddiqui, président du Comité technique (technical-chair@caa-aca.ca).

Expositions et commandites

Des expositions sur les produits et services acoustiques se dérouleront pendant le Congrès. L'aire d'exposition tiendra également lieu d'espace-détente pendant les pauses. Pour des renseignements sur les expositions et les possibilités de commandite, veuillez contacter M. Packirisamy, coordonnateur des expositions (exhibits@caa-aca.ca).

Participation étudiante

Les étudiants et étudiantes inscrits à une université canadienne peuvent participer au concours de la meilleure communication étudiante. Ils sont admissibles à une subvention de déplacement, à condition de soumettre leur demande à cet effet avant le 15 septembre 2007, sous réserve de disponibilité des fonds.

Dates importantes

8 jun 2007 - 250 mots sommaire
22 jun 2007 - Avis d'acceptation
16 juillet 2007 - 2 page communications
15 septembre 2007- Date limite pour le tarif de préinscription
9 au 12 octobre 2007 – Congrès

Comité organisateur

Président du congrès : R. Bhat, conference@caa-aca.ca
Préseident-Comité technique :: K. Siddiqui, technical-chair@caa-aca.ca
Président – Comité des expositions : M. Packirisamy, exhibits@caa-aca.ca
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URL du Congrès : http://users.encs.concordia.ca/~caa-2007

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.

Figures/Tableaux: De petites tailles. Les insérer dans le texte dans le haut ou dans le bas de la page. Les nommer "Figure 1, 2, 3,..." Légende en 9 pt à simple (12 pt) interligne. Laisser un espace de 0.5" entre le texte.

Largeur Des Traits: La largeur des traits sur les schémas technique doivent être au minimum de 0.5 pt pour permettre une bonne reproduction.

Photographies: Soumettre la photographie originale sur papier glacé, noir et blanc.

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Pagination: Au crayon pâle, au bas de chaque page.
Tirés-à-part: Ils peuvent être commandés au moment de l'acceptation du manuscrit.

The Canadian Acoustical Association
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Application for Membership

CAA membership is open to all individuals who have an interest in acoustics. Annual dues total $60.00 for individual members and $20.00 for Student members. This includes a subscription to ***Canadian Acoustics***, the Association's journal, which is published 4 times/year. New membership applications received before August 31 will be applied to the current year and include that year's back issues of ***Canadian Acoustics***, if available. New membership applications received after August 31 will be applied to the next year.

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Subscriptions to ***Canadian Acoustics*** are available to companies and institutions at the institutional subscription price of $60.00. Many companies and institutions prefer to be a Sustaining Subscriber, paying $300.00 per year, in order to assist CAA financially. A list of Sustaining Subscribers is published in each issue of ***Canadian Acoustics***. Subscriptions for the current calendar year are due by January 31. New subscriptions received before August 31 will be applied to the current year and include that year's back issues of ***Canadian Acoustics***, if available.

Please note that electronic forms can be downloaded from the CAA Website at **caa-aca.ca**

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8. Speech Sciences
9. Underwater Acoustics
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The Canadian Acoustical Association

Formulaire d'adhésion

L'adhésion à l'ACA est ouverte à tous ceux qui s'intéressent à l'acoustique. La cotisation annuelle est de 60.00$ pour les membres individuels, et de 20.00$ pour les étudiants. Tous les membres reçoivent ***l'Acoustique Canadienne***, la revue de l'association. Les nouveaux abonnements reçus avant le 31 août s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de ***l'Acoustique Canadienne*** de cette année. Les nouveaux abonnements reçus après le 31 août s'appliquent à l'année suivante.

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Les abonnements pour la revue ***Acoustique Canadienne*** sont disponibles pour les compagnies et autres établissements au coût annuel de 60.00$. Des compagnies et établissements préfèrent souvent la cotisation de membre bienfaiteur, de 300.00$ par année, pour assister financièrement l'ACA. La liste des membres bienfaiteurs est publiée dans chaque issue de la revue ***Acoustique Canadienne***. Les nouveaux abonnements reçus avant le 31 août s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de ***l'Acoustique Canadienne*** de cette année. Les nouveaux abonnements reçus après le 31 août s'appliquent à l'année suivante.

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M. André L'Espérance - (418) 686-0993
contact@softdb.com - Sillery, QC

Solutions Acoustiques Inc.
Sylvain Larivière - (514) 793-3767
sylvainlariviere@solutionsacoustiques.com - Laval, QC

SounDivide Inc.
C.W. Roy Bakker - (416) 208-3040,
roy.bakker@SounDivide.com - Mississauga

Spaarg Engineering Ltd.
Dr. Robert Gaspar - (519) 972-0677
gasparr@kelcom.igs.net - Windsor, ON

State of the Art Acoustik Inc.
Dr. C. Fortier - (613) 745-2003
sota@sota.ca - Ottawa, ON

Swallow Acoustic Consultants Ltd.
Mr. John Swallow - (905) 271-7888
jswallow@jsal.ca - Mississauga, ON

Tacet Engineering Ltd.
Dr. M.P. Sacks - (416) 782-0298
mal.sacks@tacet.ca - Toronto, ON

Valcoustics Canada Ltd.
Dr. Al Lightstone - (905) 764-5223
solutions@valcoustics.com - Richmond Hill, ON

Vibro-Acoustics
Mr. Tim Charlton - (800) 565-8401
tcharlton@vibro-acoustics.com - Scarborough

Wakefield Acoustics Ltd.
Mr. Clair Wakefield - (250) 370-9302
nonoise@shaw.ca - Victoria

Water & Earth Science Assoc. (WESA)
Dejan Zivkovic, M.Sc. - (905) 639-5789 x151
dzivkovic@wesa.ca - Burlington, ON

West Caldwell Calibration Labs
Mr. Stanley Christopher - (905) 595-1107
info@wccl.com - Brampton, ON

Wilrep Ltd.
Mr. Don Wilkinson - (905) 625-8944
info@wilrep.com - Mississauga, ON