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EDITORIAL 1

TECHNICAL ARTICLES AND NOTES / ARTICLES ET NOTES TECHNIQUES

**A synthetic aperture matched field approach to acoustic source localization in a shallow-water environment**
*Garry J. Heard and Ian Schumacher* 3

**Air conducted and body conducted sound produced by own voice**
*Mie Ø. Hansen and Michael R. Stinson* 11

**The benefit of contextual cues for the perception of temporally degraded speech**
*Sharon M. Abel and Robert Najafee* 25

**The elusive decibel: thoughts on sonars and marine mammals**
*David M. F. Chapman and Dale D. Ellis* 29

OTHER FEATURES / AUTRES RUBRIQUES

**Acoustics Week in Canada 1998 Semaine canadienne d'acoustique** 21, 33

**Minutes of CAA Directors' meeting / Compte rendu de la réunion des directeurs de l'ACA** 35

**News / Informations** 39


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Jérémie Voix

# EDITORIAL

Yes, it's me again. I'm still sitting here in the Editor-in-Chief's chair, dealing with the June issue, despite what I said in my last editorial. I guess a good editor is like good ink - instead of disappearing suddenly, it fades away slowly!

The future of the editorship of *Canadian Acoustics* is now known. John Bradley and Trevor Nightingale will produce the September proceedings issue. The new editor, Ramani Ramakrishnan, will take over in time to produce the December issue. He has a lot to learn; please give him your full support.

Finishing my term as editor with this issue, instead of with the last one, has one distinct benefit. I get to go out on a high, rather than on a low, note. The last issue, for completely coincidental reasons, only contained one paper, for the first time in years. This issue contains four! If only we could keep up that pace.

Don Jamieson, coordinator of Acoustics Week in Canada 1998, has asked me to pass on some information about the meeting. Some sessions are still being scheduled, but more than 50 papers have been confirmed on the following nine topics: acoustics in telecommunications, architectural acoustics, developmental psychoacoustics, hearing and aging, musical acoustics, noise control, occupational hearing loss, speech perception, and speech production and disorders. The following plenary sessions are planned:

"Current developments in speech technology", by Professor Li Deng, University of Waterloo, School of Computer and Electrical Engineering;

"The state of the art in hearing aids", Steve Armstrong, Senior Design Engineer, Gennum Corporation, Burlington, ON;

"Ultrasound imaging", Dr. Aaron Fenster, Director, Advanced Imaging Research Group, The John P. Robarts Research Institute, University of Western Ontario.

Further updates on the meeting will be posted on the CAA web page at: http://www.uwo.ca/hhcru/caa/awc98.

Eh oui, c'est encore moi! En dépit de ce que j'avais annoncé dans mon dernier éditorial, j'occupe toujours le poste de rédacteur en chef, ayant coordonné le numéro de juin. Je commence à croire qu'un bon rédacteur, c'est comme de l'encre; plutôt que de disparaître subitement, il ne fait que s'effacer lentement!

Le nouveau comité éditorial de *l'Acoustique Canadienne* est maintenant connu. John Bradley et Trevor Nightingale produiront le numéro de septembre portant sur les actes du congrès. Le nouveau rédacteur, Ramani Ramakrishnan, prendra la relève à temps pour produire le numéro de décembre. Il a beaucoup de pain sur la planche; je vous prie de lui apporter tout l'appui nécessaire.

Terminer mon mandat comme rédacteur avec ce numéro, plutôt qu'avec le précédent, a son avantage. Je peux partir sur une bonne note plutôt que sur une note plus faible. La dernière édition, pour des raisons purement circonstancielles, comptait un seul article, pour la première fois depuis des années. Ce numéro en contient quatre! Si seulement, on pouvait maintenir ce rythme...

Don Jamieson, coordonnateur de la Semaine Canadienne d'Acoustique 1998, m'a demandé de vous transmettre des informations à propos du congrès. Certaines sessions sont encore en préparation, mais plus de 50 résumés ont été acceptés dans les neuf domaines suivants: acoustique dans les télécommunications, acoustique architecturale, psychoacoustique développementale, audition et vieillissement, acoustique musicale, contrôle du bruit, perte auditive due au bruit, perception de la parole ainsi que production et troubles de la parole. Les sessions plénières suivantes sont également planifiées:

"Développements récents en technologie de la parole" par le Professeur Li Deng, Université de Waterloo, École de génie informatique et électrique;

"État de la question sur les aides auditives" par Steve Armstrong, Ingénieur senior de développement, Gennum Corporation, Burlington, ON;

"Imagerie par ultrasons" par Dr Aaron Fenster, Directeur, Groupe de recherche en imagerie avancée, Institut de recherche John P. Robarts, Université Western Ontario.

Les mises à jour du congrès seront disponibles sur la page web de l'ACA à: http://www.uwo.ca/hhcru/caa/awc98.

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*Canadian Acoustics / Acoustique Canadienne 26(2) 3-10 (1998)* **Research article / Article de recherche**

# A SYNTHETIC APERTURE MATCHED FIELD APPROACH TO ACOUSTIC SOURCE LOCALIZATION IN A SHALLOW-WATER ENVIRONMENT

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## ABSTRACT

Matched field processing is a developing technique for localizing underwater acoustic sources and inverting measured acoustic fields for ocean waveguide properties. Considerable successes have been reported using this technique in deep water and, in the recent past, the technique has shown promise for localizing acoustic sources in shallow water. This paper describes an experiment conducted in the shallow water of the Western Bank near Sable Island on the eastern Canadian continental shelf. The experiment involved examining the localization performance of a small vertical array consisting of only four hydrophones and spanning approximately one-third of the water column against a relatively low signal-to-noise ratio source in a weakly range-dependent environment. In this experiment, the source emitted a series of low-frequency tones that were detectable by Fourier analysis of the received time-series. By estimating the phase of the received signal as a function of time, it was possible to estimate the relative change in the target range during an interval. Using a modified Bartlett matched field processor, the phase information was incorporated to estimate the true target range and depth during the same interval. The results of this experiment indicate that matched field localization can be dramatically improved by incorporating auxiliary information, such as the estimated signal phase history. In this case, the performance of the vertical receiving array approximates the expected performance of a planar array with the same vertical dimension as the actual receiver and a horizontal dimension controlled by the duration of the interval during which the phase history is estimated.

## SOMMAIRE

Le traitement de champs appariés est une technique en cours de développement pour localiser des sources acoustiques et inverser les champs acoustiques mesurés afin de déterminer les propriétés de guides d'ondes. On signale que cette technique a eu beaucoup de succès en eau profonde et, dernièrement, elle a eu des résultats prometteurs pour localiser des sources acoustiques en eau peu profonde. Ce rapport décrit une expérience menée lors de l'essai dans les eaux peu profondes du Banc ouest près de l'Île de Sable sur le plateau continental du Canada. Cette expérience consistait à examiner les performances de localisation d'une antenne verticale ne comprenant que quatre hydrophones et couvrant environ le tiers de la colonne d'eau vis à vis d'une cible à rapport signal/bruit relativement faible dans un environment dépendant dans une faible mesure de la distance. Lors de cette expérience, la cible émettait une série de tonalités basse fréquence détectables par analyse de Fourier de la série chronologique reçue. En estimant la phase du signal reçu en fonction du temps, on a pu estimer la variation relative de la distance de la cible pendant un intervalle donné. En utilisant un processeur de champs appariés de Bartlett modifié, on a incorporé l'information de phase pour estimer la distance et la profondeur absolues de la cible pendant le même intervalle. Les résultats de cette expérience indiquent que la localisation par champs appariés peut être grandement améliorée si l'on incorpore de l'information auxiliaire telle que la discordance de phase du signal. Dans ce cas, les performances de l'antenne verticale s'approchent de celles attendues d'un antenne planaire ayant le même dimension verticale que le récepteur en cause et une dimension horizontale déterminée par l'intervalle pendant lequel on estime la discordance de phase.

# 1 INTRODUCTION

The main objective of the work described in this paper was to investigate the use of a matched field processing (MFP) source localization technique with data obtained using a small vertical line array (VLA) receiver in shallow water where source motion is used to improve the resulting estimate of the acoustic source location.

Matched field processing involves optimizing the agreement between measured and modelled acoustic field values, where the modelled values are dependent on parameters such as source location that are initially unknown. A search for the unknown parameters is conducted, guided by the comparison of the measured and modelled fields. Section 2 describes matched field processing basics in more detail.

This paper presents a technique for handling acoustic source motion in MFP localization when it is possible to estimate the relative motion of the source. In our example, the relative phase change of tonal signals during the observation intervals provided the required estimates of relative source motion which was not restricted to constant velocity. Other researchers have considered MFP with moving sources, notably Zala and Ozard [1] who allow for constant source motion during the observation period, and Tantum and Nolte [2] who employ source dynamics in a new processor to improve tracking capabilities over an extended period of observation. Our technique could be extended to include a search over a completely unknown or partially known track, but this would greatly increase the processing demands on the method. In essence, the advantage of using spatially coherent processing techniques over incoherent methods is demonstrated in this work.

Using experimental data, an illustration of the MFP performance enhancement with the inclusion of temporal field information is given. One interpretation of this enhanced performance is that the inclusion of the temporal information allows the VLA to function as a virtual planar array with a vertical dimension equal to the vertical extent of the VLA and a horizontal dimension controlled by the extent of the radial source motion during the integration period. This paper illustrates the gains to be had from spatially coherent processing. It is also possible to achieve improvement by processing across multiple frequencies in a coherent fashion [3].

In the following section, the concepts of MFP and the standard Bartlett processor are introduced. Section 3 describes the field trial and experimental setup. Section 4 provides details of the trial area environment and the propagation model used in the MFP. Section 5 describes the modification of the Bartlett processor to include temporal field information. Section 6 describes how phase tracking was used to include the temporal information in the data obtained from the field trial. Section 7 shows how the MFP performance improves with the inclusion of successively longer data samples. Section 8 shows how the acoustic source can be tracked over an extended period of time. Finally, Section 9 summarizes the results of this work.

# 2 MATCHED FIELD PROCESSING

Matched field processing is the term applied to an acoustic source localization technique, generally attributed to Bucker [4], that attempts to optimize the correlation between a set of measured field values and a replica set generated by assuming a source location and employing an acoustic propagation model. An excellent introduction to the topic was written by Tolstoy [5]. Recently, in addition to localization, MFP has been applied to geoacoustic inversion and also to ocean tom-

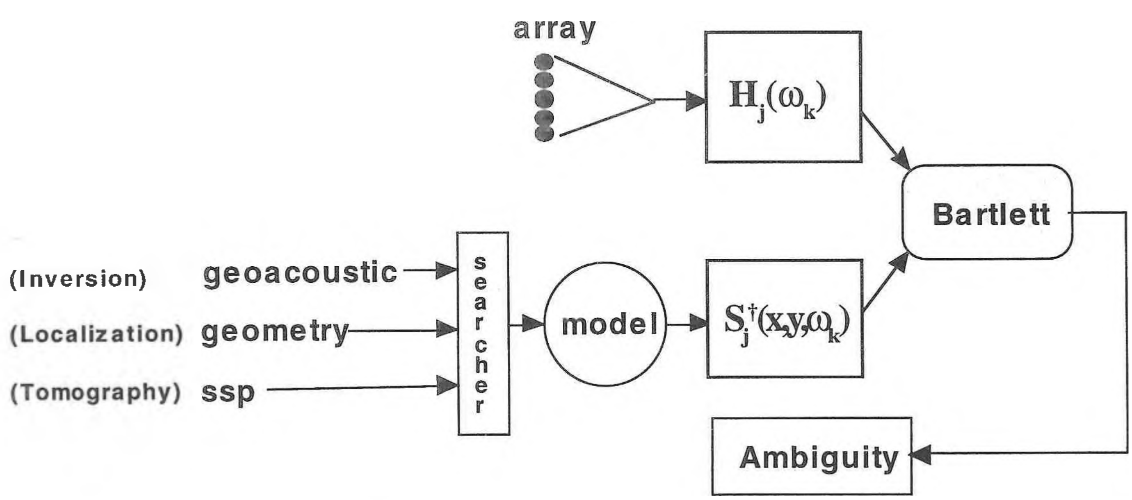


*Figure 1 Block diagram of a straight-forward MFP scheme employing a Bartlett processor. By searching over different parameters MFP can be employed for various purposes, such as: inversion, localization, or tomography.*

ography.

Figure 1 shows a block diagram of a typical MFP scheme. Field samples are measured by hydrophones in an array and a data sample taken over a short time interval, hereafter referred to as a 'snapshot', is Fourier transformed and a cross-spectral matrix (CSM) for one or more frequency bins is created. The measured CSM is then correlated with a set of steering vectors generated by assuming a suitable set of inputs to an acoustic propagation model. In this case, the suitable model inputs are the sound-speed profile, geoacoustic properties of the bottom, the bathymetry, and a guessed location for the source. In Figure 1 the correlation is carried out by a Bartlett correlator, to be described later. It is common to call the correlation function, the processor. Usually, the assumed source location (or whatever parameters are unknown) are varied in a systematic fashion by the searcher block and the correlation is computed for each input state. The correlation is normalized so that a maximum of unity is obtained when the measured and modelled fields are proportional and zero is obtained when the fields are completely uncorrelated. In this fashion, the maxima of the correlation are interpreted as occurring at best estimates of the unknown parameter values.

Rarely is the environment sufficiently well known in MFP problems that significant correlation results from the input of first-estimate values for the source-receiver geometry and geoacoustic parameters. Usually, it is necessary to perform a search over at least some parameter intervals, particularly for the receiving hydrophone locations, in order to obtain a significant correlation at the true source location. Collins and Kuperman [6] call this combined search for source position and environmental parameters 'focalization'. In the case of the trial data analyzed here, the model parameters were known sufficiently well that an indication of the true source location was obtained directly from the correlation values with estimated environmental inputs; however, the correlation at the true location was not the largest value obtained, and so, without *a priori* knowledge we would have misidentified the source location. In order to assess just the effect of the inclusion of the temporal information, we froze the environmental parameters and searched only over the location parameters.

The Bartlett processor is the most often used matched field processor and is a simple normalized correlation of the measured and modelled acoustic fields. With an extension for an incoherent sum over multiple frequencies, $\omega_k$, we can write the Bartlett processor as

$$A(x,y) = \frac{\sum_{k=1}^{K} \left| \sum_{j=1}^{N} H_j(\omega_k) S_j^{\dagger}(x,y,\omega_k) \right|^2}{\sum_{k=1}^{K} \sum_{j=1}^{N} \left| H_j(\omega_k) \right|^2 \sum_{k=1}^{K} \sum_{j=1}^{N} \left| S_j(x,y,\omega_k) \right|^2}, \qquad \textbf{Eq. 1}$$

where $H_j(\omega_k)$ is the field measured on hydrophone $j$ at frequency $k$, $S_j(x,y,\omega_k)$ is the modelled field for hydrophone $j$ and frequency $k$ with the source at location $(x,y)$, and $\dagger$ denotes the complex conjugate. $H_j$ may be obtained from each hydrophone by Fourier transformation of a data snapshot of a length determined empirically and partially dictated by the source's range-rate and the range-dependence of the environment. The $S_j$ are obtained directly from a model run. This processor will be modified in Section 5 to include multiple snapshots from a moving source.

## 3 THE FIELD TRIAL

In July 1994, Defence Research Establishment Atlantic (DREA) conducted a shallow-water MFP experiment on the Western Bank area of the Scotian Shelf. The MFP experiment employed the DREA OMEGA Vertical Line Array (VLA) [7] as a receiver. At the time, only 4 stations of this modular array were available for use. These four hydrophone stations were located at depths 43.8, 51.0, 58.2, and 72.5 metres providing an aperture of just 28.7 m, which represented less than one-third of the nominal 95 m water depth. The array was assumed to be vertical during the data collection. The VLA was not outfitted with tilt sensors, but the arrival times of impulsive acoustic signals were consistent with less than one degree of tilt.

The matched field experiment involved towing the DREA Moving Coil Projector (MCP) [8] at a nominal depth of 30.5 m along a straight path approximately 12 km in length past the location of the VLA. The MCP simultaneously emitted three tones at 17, 21.5, and 30 Hz. The received signal-to-noise ratio (SNR) for these tones varied during the trial from 6 to 9 dB in a 1-Hz wide band. In MFP studies currently published, SNR values are typically much higher than our current levels. This is not to say that MFP requires a high SNR, it just means that the ever present effects of mismatch are more easily overcome with a strong signal. In the current work, the available SNR was more than sufficient.

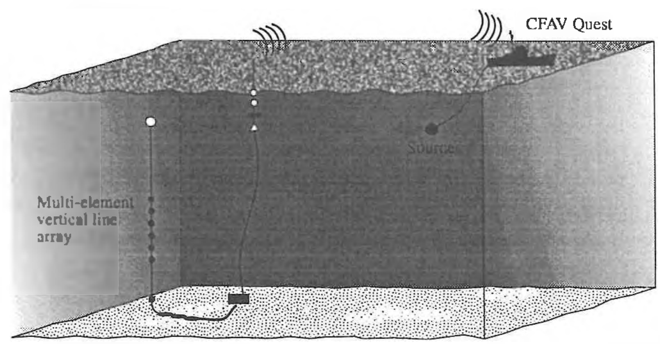


*Figure 2 Experiment setup for the VLA matched field experiment.*

Figure 2 shows the experimental setup for the MFP experiment. Note that the location of the VLA is not exactly the same as the location of the surface telemetry float. The horizontal distance between the VLA and the surface float is as much as several hundred metres (~2 cables) and the orientation and range change from deployment to deployment and at the mercy of current, wind, and wave conditions. This uncertainty in the receiver position was of considerable importance to the MFP analysis. In contrast, the position of the acoustic source is relatively accurately known due to the measurement of the cable scope, MCP depth, and global positioning system (GPS) navigation. Since the receiver positions are less accurately known than the source position, we decided to initially localize the receiver rather than the source (as is more commonly done). The bottom hydrophone location was used as a reference position. Later, when a reasonably accurate receiver position had been estimated, we reversed the procedure and localized the source.

## 4 THE ENVIRONMENT

The trial site chosen for this experiment was near the edge of the continental shelf. A bathymetric survey was conducted and it was determined that along the tow direction, which was almost parallel to the shelf edge, the mean bottom slope locally was approximately 0.08°; however, the variance in the depth is large compared to the mean and range-independent modelling was deemed appropriate only for a 4-km long portion of the tow to the east of the array position. Thomson's finite-difference parabolic equation model (PE) [9] was used to model the range-independent portion of the track. To the west of the array, an adiabatic normal mode approximation was used in the modelling to reduce the effects of mismatch in the bathymetry by including a smoothed version of the estimated bathymetry. This time, the KRAKEN model of Porter [10] was used, because the model is well suited to the adiabatic approximation and the numerical techniques that were employed. Run times were also significantly less with this approach rather than the alternative of recomputing the PE field estimate a number of times. Perpendicular to the tow path the bathymetry exhibits a mean slope about twice that along the tow-path direction. Although this cross-slope will have some effect on the propagation, no account of it was included in the modelling since the impact was expected to be small.

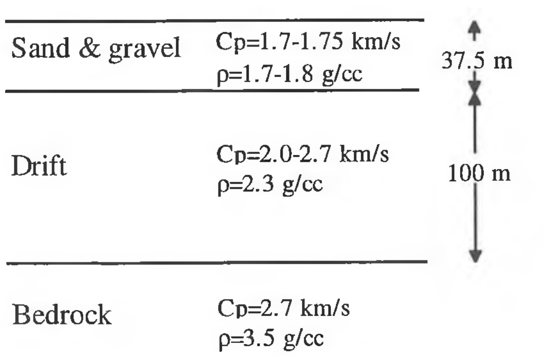


*Figure 3 Geoacoustic model used.*

Sound-speed profiles were collected with the aid of expendable bathythermographs (XBT) throughout the duration of the field trial. Thermal data were converted to sound speed by assuming a constant 35 parts per million salinity concentration and using Medwin's [11] empirical formula. Over a three-day period, eleven XBT profiles were acquired in an 11 km by 24 km area enclosing the MFP tow-path. These profiles showed considerable similarity in the first 100 m of depth (note that the maximum depths varied from 80 to 250 m, depending on location) and so an average sound-speed profile was used for the MFP localization. In retrospect, using the average profile may have contributed to the mismatch, since there is an indication of an east-west variation in the sound-speed profile that probably should have been included in the optimization. Typically, we obtained maximum correlations between the measured and modelled fields in the range 0.6 to 0.8 indicating a probable match between the model environment and the actual environment, but still with some room for improvement. The extent of the mismatch produced by employing the averaged profile has not been examined, but it could be a significant factor in the remaining error.

The geoacoustic properties of the area were taken mostly from a report by Osler [12] that summarizes the information available for the experiment site. Since this work was done, some additional estimates of the sub-surface structure have become available and it is known that some of the geoacoustic data was misinterpreted. Specifically, the layer of Scotian Shelf Drift (sand, clay and silt, pebbles, cobbles and boulders) included beneath the uppermost sediment layer of relatively hard sand and gravel is absent in the region where the experiment was carried out. Seismic soundings in the area have shown that the sand lies directly over bedrock [13] and that there is no recognizable layer of Drift. Fortunately, the highly reflective sand layer appears to have reduced the importance of the error in our geoacoustic model and good localization results were obtained. Figure 3 shows the geoacoustic model used. Absorption parameters are not critical, and not terribly well defined; we used 0.46 dB/λ for the sand, 0.3 dB/λ for the Drift, and 0.08 dB/λ for the bedrock [12]. Modelling has also shown that the acoustic fields in the water column are not strongly dependent on the shear properties, so shear values were not generally included in the modelling. When shear was included we used 260 m/s for the sand and Drift [14], and 800 m/s for the bedrock [12].

## 5 ENHANCED PROCESSOR

The Bartlett processor, Eq. 1, was modified to include the effect of source motion over an extended period of time. Since Eq. 1 uses snapshots of data to estimate $H_j(\omega_k)$, an obvious extension of the processor is to combine a number of snapshots into a single estimate of the correlation. One way of combining multiple data snapshots is given below,

$$A(x,y)=\frac{\sum_{k=1}^{K}\left|\sum_{u=1}^{U}\sum_{j=1}^{N}H_j(\omega_k,t_u)S_j^{\dagger}\left(x+r_x(t_u),y+r_y(t_u),\omega_k\right)\right|^2}{\sum_{k=1}^{K}\sum_{u=1}^{U}\sum_{j=1}^{N}\left|H_j(\omega_k,t_u)\right|^2\sum_{k=1}^{K}\sum_{u=1}^{U}\sum_{j=1}^{N}\left|S_j\left(x+r_x(t_u),y+r_y(t_u),\omega_k\right)\right|^2},$$

**Eq. 2**

where $H_j(\omega_k,t_u)$ represents the spectral estimate for hydrophone $j$, at frequency $\omega_k$, using the $u^{th}$ data snapshot, and $S_j(x+r_x(t_u),y+r_y(t_u),\omega_k)$ represents the modelled field estimate at frequency $\omega_k$ for an assumed source position $(x+r_x(t_u),y+r_y(t_u))$. The position $(x,y)$ is the assumed source position at the time of the first snapshot, and $r_x$ and $r_y$ represent the relative motion of the source during the total interval of the U snapshots. Coordinates $x$ and $y$ may represent range and depth (as they do in this application), azimuth and bearing, or any other coordinates as required by the application.

Eq. 2 combines the information from the individual snapshots semi-coherently. For each snapshot, data from each hydrophone are combined in a spatially coherent sense, but contributions from different frequencies are summed incoherently. Variations of Eq. 2 have been tried, but this particular semi-coherent approach appears, qualitatively, to provide a useful combination of robustness and spatial resolution.

Direct application of Eq. 2 requires breaking an interval of data into U snapshots. In general, these snapshots may be randomly selected during the interval; however, the simplest approach is to order the snapshots chronologically. The snapshots may also overlap in time. The spacing and length of the snapshots are in part dictated by the behaviour of the source. The total length of the interval depends on the frequency stability of the emitted tones, or upon the available knowledge of the signal dynamics. Our approach has been empirical with only rough guidelines for snapshot length and overlap based on frequency and spatial resolution concerns. In our application of Eq. 2 to the trial data we have used a slightly modified form that will be described in the next section.

Successful use of Eq. 2 requires an accurate estimate of the relative motions. People immediately ask; why use Eq. 2 if you have the relative motion estimates? The answer to this question is that Eq. 2 provides an estimate of, for example absolute range and depth, when you only know the relative change in the source range and depth. With additional assumptions about source behaviour, Eq. 2 also provides depth estimates when no prior information was available. The next section shows how the relative motions were obtained for the field trial data previously discussed.

## 6 PHASE TRACKING

Application of Eq. 2 to the current experiment was carried out by making a reasonable assumption about the vertical source motion and by using phase tracking to estimate the relative radial source motion. Since the acoustic source was towed at nearly constant speed and weather conditions were favourable, it was assumed that the depth variations of the acoustic source during the tow were negligible: this implies that $r_y(t_u)\to 0$ for all $t_u$. An estimate of $r_x$ was obtained by employing an analysis to estimate the phase of the received sinusoidal signals and interpreting the temporal change in phase to be due entirely to a change in the source-receiver separation. The remainder of this section describes in more detail the process of determining $r_x$.

The first step in the process of determining $r_x$ is to heterodyne the received time-series by the known source frequency of interest. Note that in this case we knew the projector frequencies, but in general we have found that good results are obtained even when the signal frequency is initially unknown and an estimate from a spectral analysis is used. Once the signal of interest has been moved to baseband, a relatively narrow-bandwidth low-pass filter operation removes unwanted noise. The filter can be almost any linear, steep roll-off filter with parameters chosen by examining the signal spectrum near baseband. Optimal filters can also be designed based on models of the source motion [15], but in general this is not necessary. After filtering, the signal is decimated to reduce the number of samples and ease subsequent handling. The amount of decimation is controlled by the retained bandwidth of the signal, the magnitude of the derivative of the heterodyned and filtered signal with respect to time, and by the final interval at which estimates of $r_x$ are required.

The second step in the process is to recover the phase, $\phi(t)$, of the received signal. Several methods of performing this operation are available, but our preferred method involves a numerical integration of the heterodyned, filtered, and decimated signal $g(t)=a(t)\exp(i\,\phi(t))$, where $a$ represents the possibly time-dependent amplitude. Given the form of $g(t)$, it is easy to see that the change in the phase from time 0 to time $t$ is given by

$$\phi(t)=\int_0^t \mathrm{Im}\left[\frac{g'(\zeta)}{g(\zeta)}\right]d\zeta, \qquad \textbf{Eq. 3}$$

where the prime denotes the derivative with respect to time. The advantage of Eq. 3 is that the resulting phase does not need to be unwrapped as $\phi \in \Re$. Other methods usually result in the phase estimate being defined on the interval $(-\pi, \pi]$ and require the additional step of unwrapping the phase estimates. Once $\phi(t)$ has been determined, then $r_x(t)$ is given by

$$r_x(t) = \frac{c}{\omega}\phi(t), \qquad \textbf{Eq. 4}$$

where $\omega$ is the radial frequency of interest and $c$ is approximated by the phase speed of the dominant mode or ray arrival. The discrete time equivalents of Eqs. 3 & 4 are easily obtained by approximating the derivative of $g(t)$ by a first order difference. In this work, $r_x(t)$ was estimated from each of the three tonals and an average was formed for final use.

The use of the filter suggests a way to apply Eq. 2 in this particular application. The output of the filter is itself an approximation of $H_j(\omega_k, t_u)$ with the harmonic time dependence removed by the shift to baseband. This implies that we can interpret $u$ as being only 1 sample long (although the filter will likely have an integration time of several seconds to a minute) and form the numerator of Eq. 2 by taking the dot product of the filtered data with the model output directly, without the need to add the harmonic dependence, $e^{i\omega t}$, into the model predictions (remember that the field samples do not occur at the same time). This technique saves computation time and works quite well in practice.

The discussion to this point has assumed that we are receiving a direct-path signal from the acoustic source. In practice, we usually receive a signal composed of several multi-path arrivals. The presence of the multi-path results in a bias in the estimated phase and can result in severe errors in the estimate of radial motion. The bias term is small if the direct path amplitude dominates; unfortunately this can't be counted on in practice. In our current application with a VLA in shallow water, a simple solution to the multi-path bias was to sum the signals from each hydrophone in the array thereby effectively beamforming horizontally. Alternatively, one can think of this as mode filtering. In the present case, we favour mode 0 by simply summing the hydrophone signals. Our mapping of phase-change to distance-change (Eq. 4) should therefore use the phase speed of mode 0 for c. Although we have only used the mode 0 signals in this work, the potential exists to estimate the change in distance using the contributions from each mode.

Simulation of the multi-path bias showed that it could be a considerable problem in our current application. Using the beamformed signal, rather than the hydrophone signals directly, indicated that a dramatic improvement in the radial range estimates should be possible. On comparing the radial range estimates from GPS positions with those from the phase-tracked beamformed signal, an excellent agreement was obtained.

## 7 SYNTHETIC APERTURE

With phase-tracking accomplished for each tonal of interest,

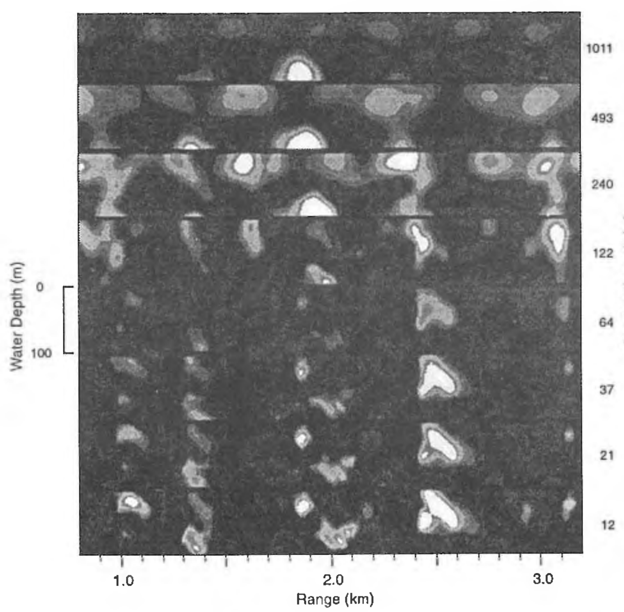


*Figure 4 Array localization versus aperture length using known source location. Shortest aperture (12 m) is at bottom, longest aperture (1011 m) at top. Each ambiguity strip represents 2.4 km by 100 m. True array location estimated to be at 2 km range (±360 m) and 72.5 m depth.*

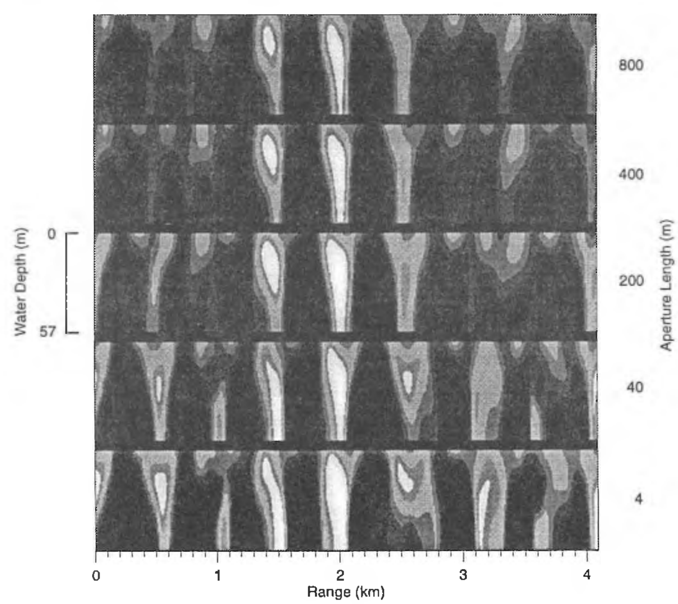


*Figure 5 Source localization vs. aperture length using estimated array position. Shortest aperture (4 m) at bottom, longest aperture (800 m) at top. Each ambiguity strip represents a region 4.1 km by 57 m. True source location at 2 km range and 30 m depth.*

we can now proceed to apply Eq. 2 and obtain a synthetic aperture matched field result. The aperture is equivalent to the distance travelled by the source during the interval of observation. Figure 4 shows the ambiguity surfaces obtained for 8 different aperture lengths in the region where range-independent modelling was found to be adequate. Each strip in the figure represents the depths from 0—100 m and a 2.4 km range interval. In this figure, and in all the following figures, lighter regions indicate higher correlation values. The shortest aperture is at the bottom of the figure and represents a source motion of only 12 m. This result is equivalent to a single snapshot estimate (approx. 6 seconds integration time) and the result fails to accurately localize the receiving array (in fact, we were unable to localize the source for any choice of processing parameters without employing phase tracking). For the data segment processed, the true array position was estimated to be at a range of (2±0.2) km and a depth of (72.5±1) m (relative to a point on the surface directly above the MCP at the start time of the data segment). As aperture length is increased, the number of false localizations decreases. For aperture lengths exceeding 122 m a useful localization is obtained.

Figure 5 shows the result of using the estimated location of the receiving array and applying the synthetic aperture technique to localize the source. In this figure, the aperture lengths vary from 4 to 800 m with the shortest aperture at the bottom and the longest at the top. Each strip represents a region just over 4 km long and water depths 0–57 m. The true source location is accurately localized for each aperture tried, but an improvement in the results is seen for apertures up to 200 m. For the longer apertures we see a reduction in the number and amplitude of false localizations, but we also note a reduction in the correlation value at the true location. This limitation is most likely the result of cumulative mismatch.

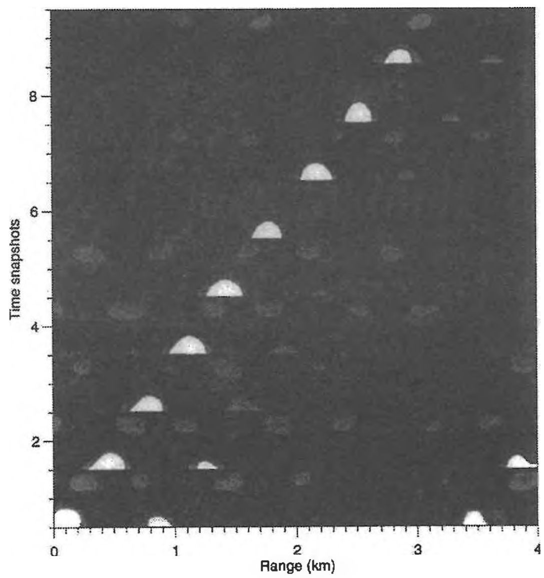


*Figure 6 Tracking the array position as a function of time using known source location. Aperture length is 300 m.*

## 8 TRACKING

Target tracking can be accomplished by sequentially processing segments of data. Sequentially processing intervals of data is not an optimal way of performing tracking [2], but it does illustrate that the synthetic aperture technique can be successfully applied at any starting point in the data. Figure 6 shows

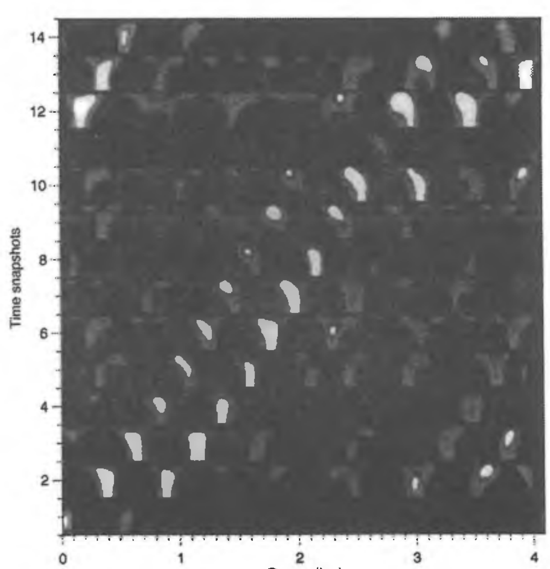


*Figure 7 Tracking the source location using an estimated array location. Aperture length is 200 m.*

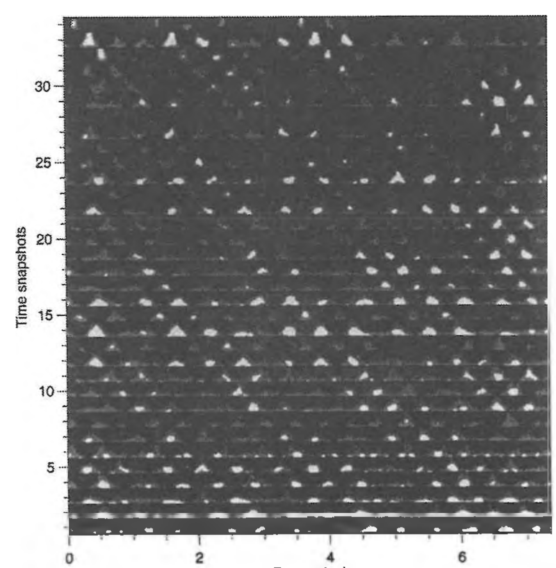


*Figure 8 Source track in the range-dependent region using the assumed array location . Note the repetition of ambiguity at intervals in range.*

the result of employing a 300 m aperture length to localize the receiving array. In this figure the array location is unambiguously tracked as the source is towed away from the receiver.

Figure 7 shows the result of tracking the source using the estimated receiver location. The source is ambiguously tracked from 0.5 to 3.2 km in the 14 snapshots shown. The increased ambiguity in the figure (relative to the array track shown earlier) is undoubtedly the result of increased mismatch due to using the estimated receiver location. In this situation where there is considerable ambiguity, more sophisticated tracking algorithms would likely improve the localization. Wilmut, Ozard and Yeremy [16] discuss one technique that should be applicable.

Figure 8 shows the result of tracking the source along a 7 km path in the range-dependent region west of the receiving array. Construction of this figure required using an estimated receiver location and the results show the effects of increased mismatch due to both the receiver position uncertainty and the more complicated geometry. The true source location begins in the lowest strip at just over 7-km range and ends in the top strip at approximately 0.5-km range.

## 9 CONCLUSIONS

By including temporal data in MFP localization, enhanced performance is realized through the use of spatially coherent processing. In this particular application, the results indicate that even a very modest VLA can be used to accurately localize a moderate level acoustic source provided that it is possible to estimate accurately the relative source motion, either through other measurements, by dynamic motion modelling, or by searching for the motion. The source motion itself can be non-uniform. The results were best where range-independent modelling was appropriate, but useful results were also obtained in an area of weak range-dependence.

As we have discovered since completing this study, significant mismatch in the geoacoustics and in the structure of the sound-speed profile were included in our modelling. Refinement of the model inputs or focalization would likely result in further enhancement of the localization performance; however, the excellent results indicate that at least for the low-grazing angles involved our assumed environment was sufficiently close enough to the real world.

The technique of phase-tracking tonal signals, while not always applicable, does when it can be applied result in an estimate of the relative change in source position. The use of suitable models or assumptions of source motion and MFP techniques can then be used to obtain absolute source range and depth estimates. The applicability of the technique has been empirically demonstrated and future work may now consider the SNR, range-rate limitations, multi-path environment, and other factors affecting the applicability of the method.

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# AIR CONDUCTED AND BODY CONDUCTED SOUND PRODUCED BY OWN VOICE

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## ABSTRACT

When we speak, sound reaches our ears both through the air, from mouth to ear, and through our body, as vibrations. The ratio between the air borne and body conducted sound has been studied in a pilot experiment where the air borne sound was eliminated by isolating the ear with a large attenuation box. The ratio was found to lie between -15 dB to -7 dB, below 1 kHz, comparable with theoretical estimations. This work is part of a broader study of the occlusion effect and the results provide important input data for modelling the sound pressure change between an open and an occluded ear canal.

## SOMMAIRE

Quand on parle, le son atteint nos oreilles à la fois à travers l'air, de notre bouche à l'oreille, et à travers notre corps comme vibrations. Le rapport entre le son se propageant dans l'air et le son se propageant dans le corps a été étudié dans un projet-pilote où le son se propageant par l'air fut éliminé en isolant l'oreille avec une grande boîte d'atténuation. Il fut découvert que ce rapport se trouve entre de -15 dB à -7 dB, au-dessous de 1 kHz, comparable aux évaluations théoriques. Ce travail fait partie d'une étude plus large sur l'effet d'occlusion et les résultats fournissent des données d'entrée importantes pour la modelisation du changement de la pression sonore entre un canal auditif ouvert et un qui est bloqué.

*On a work term from the Dept. of Acoustic Technology, Technical University of Denmark

## 1. INTRODUCTION

People using earplugs are aware that the sound of their own voice is unnaturally loud. In the following text, the term 'occlusion effect' (OE) refers to the difference in sound pressure level between closed and open ear canal while the person is speaking. A tight earmold can give an occlusion effect up to 30 dB. The occlusion effect has typically a maximum at 100-500 Hz and drops to 0 dB at 1-2 kHz, but can vary much between subjects.

The hearing organ detects both environmental sounds and the sounds we produce with our own body. The total sound pressure in the ear canal created by our own voice is thought to be mainly a sum of two components: An air conducted signal propagates from mouth through the air to the ear while a body conducted component propagates as vibrations from the larynx through the neck and skull to the soft cartilage of the ear canal, where it radiates into the canal. At low frequencies, the body conducted sound waves will radiate out of the unoccluded ear so that the open canal sound pressure will be nearly the same as the air borne sound. In the closed ear, however, the body conducted sound will be reflected by the earmold (or earplug) and an enhanced sound pressure from this component will arise in the closed ear canal.

Although the occlusion effect has been measured by various authors, additional knowledge of the air conducted and body conducted components is required to help understand and to model the occlusion effect. The availability of a model to describe the occlusion effect would be a very helpful tool for developing a method to eliminate the occlusion effect. Such a method would benefit hearing aid users, for example. An independent measure that is particularly pertinent is the relative magnitude of the body conducted sound to the air conducted component. This ratio is the object of interest in this paper.

One of the more comprehensive studies on bone conducted sound is that of Schroeter and Poesselt[1] (1986). They modelled the contributions from the middle ear and the ear canal walls to the total sound pressure in the occluded ear canal. However, they considered only the sound produced by a bone vibrator placed on the skull and not by one's own voice. The literature provides little real ear data on body conducted sound produced by one's own voice. The most relevant work is that of Békésy[2] in 1949. Békésy tried to eliminate the air conducted sound by the mean of large tubes placed on the ears. These tubes attenuated the air conducted sound by 30 dB between 150 Hz and 4 kHz, without causing an occlusion effect. The difference in loudness between open ears and covered ears of short words, using the subject's own vocalization, was measured by comparison of the words heard to the loudness of a 1 kHz tone produced by a bone-conductor on the forehead. Békésy found that the body conducted component was 0 - 10 dB lower than the air conducted component, depending on the sound produced.

This experiment needs some comment. First of all, the ratio was only measured indirectly, no real ear pressures being determined. Assuming that the loudness measurements are done using standard methods, tissue and bone act as a lowpass filter and the voice will then sound deeper when only the body conducted sound is heard. The difference in loudness between open and covered ear is therefore not necessarily a measure for the loudness at the same frequencies. Secondly, a reference microphone was not used to assure that the subjects spoke with the same level in the two situations. The present author[3] has performed an experiment showing that people automatically change their speech level when the ear is occluded. When the ear is covered and there is no occlusion effect, it must be anticipated that subjects still raise their own voice by a few dB. This phenomenon means that the ratio is smaller than that measured by Békésy.

It was therefore decided to perform an experiment in order to get an estimate of the ratio between air and body conducted voice sounds, making use of the probe tube technology that has developed since the time of Békésy.

## 2. SOUND CONDUCTION MECHANISMS

An overview of the mechanisms involved in the generation of the air and body conduction components will be presented here. The relevant anatomy is shown schematically in Fig. 1.

### 2.1. Speech Production

Speech contains voiced (vowels and voiced consonants) and unvoiced (consonants) sounds. Voiced sounds are primarily created by periodic oscillations of the vocal folds. Unvoiced sounds are aperiodic and generated by oral constriction. Oral constriction is also called occlusion in speech production terminology and should not be confused with the occlusion effect in the ear canal[4].

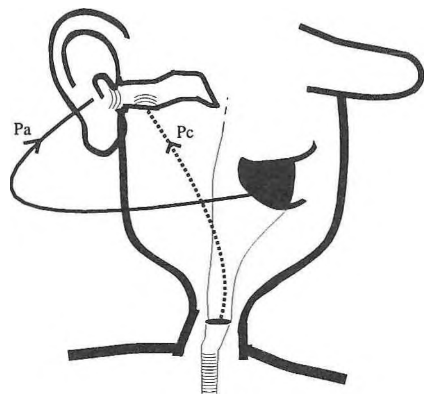



**Figure 1.** Sketch of a human head showing the air conduction ($P_a$) and body conduction ($P_c$) pathways.

The vocal folds (which form the glottis) are set into vibration by air flow from the lungs up through the vocal tract. The opening and closing of the vocal folds acts like a saw tooth generator. The time function of the velocity of volume passing through the glottis (the area between the vocal folds) is a train of approximately triangular pulses. Transformed into the frequency domain, these pulses give a line spectrum that decreases by approximately 12 dB per octave, as seen in Fig. 2. This spectrum is the glottal source spectrum. The fundamental frequency of a voice is determined by the strength and the length of the vocal folds. On average, the fundamental frequency is 125 Hz for men and 200 Hz for women.

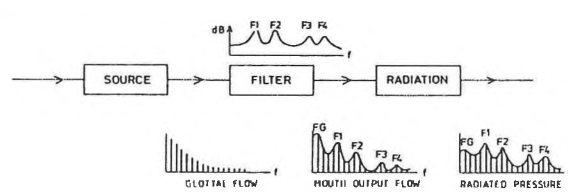



**Figure 2.** Vowel shaping.

The sound produced in the glottis is modified by the vocal tract and by the oral and nasal cavities and depends on the geometry of these structures. The shape of the oral cavity is controlled by the position of the tongue. Sound is radiated from the mouth, being controlled by lip rounding and opening of the mouth. There is also some radiation of sound from the nose. The air conduction pathway is completed by propagation of sound from mouth (and nose) around the head to the ear; this propagation is controlled by diffraction. A simple model of vowel shaping is illustrated in Fig. 2. The first line in the final spectrum represents the fundamental frequency, the next lines are the harmonics and the peaks of the envelope are the formants of the vowel. The formants are resonances in the acoustic system consisting of the vocal tract and the shaped oral cavities. Changing the shape of the oral cavity with the tongue, jaw and lip positioning leads to different vowel sounds. The three most distinct vowels with most extreme tongue positions are the cardinal vowels /i/, /a/ and /u/.

Continuous speech can be measured as a long-term average spectrum, over 60 seconds, for example. The radiated spectrum is language independent[5].

## 2.2. Ear Canal Anatomy and Acoustics

The average ear canal is about 23.5 mm long and 7 mm in diameter[6]. The outer part of the ear canal is soft tissue and the inner part is surrounded by the temporal bone. The soft part of the ear canal is surrounded by connecting tissue of the cartilage type which contains elastic fibers and makes it very flexible. The cartilage is covered by a 0.5-1.0 mm thick skin layer that holds hair cells. In the bony part of the ear canal the skin layer is thinner, approximately 0.2 mm and this skin is continuous with the skin covering the eardrum.

Seen from above, the ear canal is S-shaped. The ear canal terminates the eardrum, forming an angle of approximately $55^{\circ}$ with the inferior wall. Oliveira's[7] scannings show that the first bend is located in the cartilaginous part while the second bend is surrounded partly by cartilage and partly by bone, implying that the soft tissue reaches 1/3-1/2 into the ear canal. A standard hearing aid earmold ends between the first and second bend. On average it reaches 1/3 into the ear canal. The ear canal shape and dimensions do differ significantly from person to person.

Despite these complexities, some simple acoustical models can be used. If the largest dimension of the ear canal is much less than a quarter of the wavelength, then the ear canal can be treated as a simple cavity. This simplification is valid for frequencies below about 1 kHz. At higher frequencies, up to 6 kHz or so, the ear canal can be approximated as a uniform cylindrical tube[8,9]; the influence of higher modes is negligible[10] at these frequencies. At even higher frequencies, the individual ear canal geometry has an effect. The surfaces of the ear canal can be treated as rigid regarding air borne sound transmission because the dilatation impedance in bone and cartilage is much larger than air[10]. The eardrum and middle ear impedance can be calculated using a lumped-element model such as that of Zwislocki[11].

## 2.3. Hypotheses of Body Conduction

The body conduction pathway extends between the larynx and the ear canal. The vocal folds are located within the larynx, in the front part of the neck, just behind the thyroid cartilage. Attached to this cartilage is the thyroid muscle, running up to the hyoid bone, and muscles from the hyoid bone are directly connected to the bone around the external ear, the temporal bone. It is anticipated that the sound in the vocal tract can couple mechanically to some or all of these structures.

Vibrations from the speech organ and the mouth cavity are transmitted, not only to the outer ear canal, but to the middle ear and probably to the inner ear as well[18]. All three pathways will contribute to the perception of the body conducted sound. In this paper, though, we are only concerned with the sound pressure level in the ear canal for which the dominant source is believed to be the vibrations of the ear canal walls. We neglect, therefore, the middle ear and inner ear pathways in the modelling.

The basic requirement for producing a sound in the ear canal is that a volume change takes place, for example a (tiny) deformation of the ear canal wall. The source of body conducted sound can be the vibrations of the vocal cords and the subsequent air pressure changes in the oral cavity, exciting the skull. Békésy[2] demonstrated that one's own voice sets the skull into vertical vibration. Another way to create body conducted sound is to clench the teeth and waves will be transmitted via the bone. The result is that the cartilagineous part of the ear canal vibrates whether it has been excited by skull vibrations or directly via the tissue from the vocal cords to the ear canal.

The cartilagineous wall in the canal not only vibrates during speech but deforms when the jaw moves. Lowering the jaw increases the cross-section of the soft ear canal[7], mostly because the ventral wall is moved outwards. Hence, jaw movements cause a volume change in the ear canal, but do not create excess sound in the speech frequencies because the jaw movements are slow: It takes about 0.4 seconds (2.5 Hz) to lower the jaw during speech[12].

However, the jaw does influence the vibration of the cartilage through its inertia. Franke et al.[13] (1952) measured the ear canal sound pressure when the lower jaw vibrates relative to the skull. They placed a bone conductor on top of the skull, producing a vertical oscillation of the skull. The ear canal was closed with a rubber ear plug and the sound pressure in the ear canal was measured with open and closed mouth positions. The sound pressure in the occluded ear canal increased by 8 dB at 200 Hz. When the ear was open the increase was only 2-2.5 dB at 200 Hz. Howell et al.[14] (1988) also measured the occluded ear canal pressure on 4 subjects with the jaw closed and open. They found that the phase did not change significantly. These measurements indicate that the sound pressure level created in the ear canal by a vibrating jaw is higher when the jaw is lowered. The upper part of the jaw (the condyle) is located closer to the ear canal when the mouth is closed than when it is open. This observation supports the hypothesis that the inertia of the jaw contributes to the damping of vibrations of the cartilage.

# 3. ESTIMATING $P_C/P_A$

During speech with the ear canal open, the sound pressure in the canal is a sum of an air borne component $P_A$ and a component $P_C$ passed through the body (see Fig. 1). Assuming that these components are uncorrelated, the total root-mean-square pressure is $(P_A^2 + P_C^2)^{1/2}$. In the following, we consider the effects of blocking the air borne path, both in producing the occlusion effect and as a means for establishing the relative magnitude of the two components.

## 3.1. The Occlusion Effect

The occlusion effect $OE$ is calculated as the difference in sound pressure level between the closed ear and open ear conditions, when the canal is blocked with an earplug or hearing aid mold. With the ear closed, the air conducted component is reduced by the attenuation of the mold giving a sound pressure $\alpha P_A$, where $\alpha$ is the earmold attenuation factor. On the other hand, closing the ear canal tends to increase the sound pressure due to the body conducted component, at least at lower frequencies. The acoustic load presented by the closed canal is higher than the load for the open canal and, with an effective volume velocity source, higher sound levels are anticipated. With the canal blocked, the sound pressure due to body conducted sound is $\beta P_C$, where $\beta$ is an occlusion gain factor. The occlusion effect can then be calculated as

$$OE^2 = \frac{P_{\text{closed}}^2}{P_{\text{open}}^2} = \frac{\alpha^2 P_A^2 + \beta^2 P_C^2}{P_A^2 + P_C^2} \tag{1}$$

The values of $\alpha$ and $\beta$ depend on the depth of the earmold. Consequently, the $OE$ also depends on the insertion depth, in agreement with clinical experiences. In the present study, only standard length earmolds were used, i.e., insertion depths about 1/3 of the total ear canal length.

Some measurements of the occlusion effect are shown in Fig. 3. For these, Wimmer[15] and Thorup[16] both inserted the probe tube through a hole in the earmold, avoiding leakage. May and Dillon[17] placed the probe tube between the ear canal wall and earmold.

The $\beta$ factor has not been measured previously. As seen from Eq. (1), it is a quantity that can be derived once measurements of both $OE$ and $P_C/P_A$ are available. It is possible, though, to estimate $\beta$ theoretically if assumptions are made about how the body conduction is taking place. A model, currently under development[18], to predict the occlusion gain factor will be used in this paper to provide rough estimates for $\beta$. In this model, an effective volume velocity source in the ear canal wall represents the radiation of the body conducted sound into the canal. The ear canal is terminated using a middle ear model (modified Zwislocki

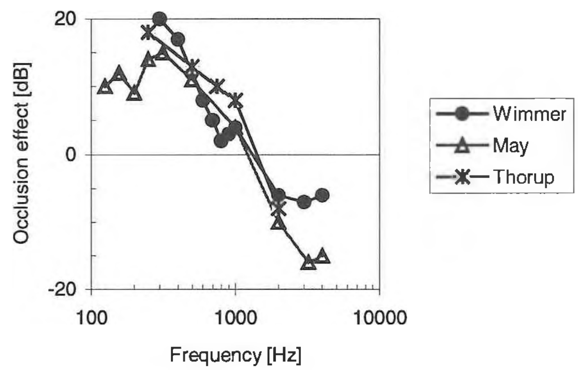



**Figure 3.** Occlusion effects, own speech (References 15, 16 and 17).

model). Initial calculations indicate that the value of $\beta$ is much greater than unity. The sound pressure $P_C$ is typically of the same order of magnitude[2] as $P_A$ (though generally smaller) and the factor $\alpha$ is less than unity, so that $(\beta P_C)^2 \gg (\alpha P_A)^2$ should be a good approximation. With this assumption,

$$OE^2 \cong \frac{\beta^2 (P_C/P_A)^2}{1+(P_C/P_A)^2} \Leftrightarrow \tag{2}$$

$$(P_C/P_A)^2 \cong \frac{OE^2}{\beta^2 - OE^2}$$

This result allows us to obtain estimates of the ratio $P_C/P_A$ from published values of $OE$.

The air conducted and body conducted sound have been assumed to be uncorrelated. However, $P_C$ and $P_A$ are likely highly correlated. It has not been possible, though, to find any reports on measurement of the phase created by vocalization, although some data on phase differences of signals traveling from one ear to the other and sound velocity in the skull has been reported. The phase difference could not be measured in this pilot study because the required two probe measurements were not done simultaneously.

## 3.2. Procedure to Measure $P_C/P_A$

Our method of determining the ratio $P_C/P_A$ is similar to that used by Békésy[2] except for two main differences. First, the ear canal sound pressures are measured directly, using probe microphones. And second, an attenuation box, rather than large tubes, is used to reduce the air conduction component. With this technique, described later, the air borne component can be reduced by a substantial factor $\gamma$, without causing an occlusion effect. Then, the pressure with the attenuation box in place $P_{\text{boxed}}$ is related to the pressure without the box $P_{\text{open}}$ through

$$\frac{P_{\text{boxed}}^2}{P_{\text{open}}^2} = \frac{\gamma^2 P_A^2 + P_C^2}{P_A^2 + P_C^2} \tag{3}$$

Now, if $\gamma$ can be made sufficiently small that $(\gamma P_A)^2 \ll P_C^2$, then we have

$$(P_C / P_A)^2 \cong \frac{P_{\text{boxed}}^2}{P_{\text{open}}^2 - P_{\text{boxed}}^2} \tag{4}$$

so a measurement of $P_{\text{boxed}}$ and $P_{\text{open}}$ yields the desired ratio $P_C / P_A$.

# 4. MEASUREMENT METHOD

This section describes the set up and equipment used in the pilot study. The three main issues are the design of the box, repeatability of the reference microphone and cushion pressure.

## 4.1. Set Up and Calibration

The test set up is sketched in Fig. 4. The subjects were seated in a chair in an anechoic chamber. The ear canal pressure in the test ear was measured with a probe microphone. At the same time, a reference signal was picked up near the opposite ear with a ½" Brüel & Kjaer microphone. The probe microphone signal was amplified with a Stanford Research System SR640 amplifier. The reference signal was amplified using a Brüel & Kjaer 2610 measuring amplifier. Both signals were lowpass-filtered with a Stanford Research System SR640 amplifier at 9 kHz in order to avoid aliasing. The signals were recorded onto DAT-tape with a sampling frequency of 44.1 kHz. In the data conversion procedure, the signals were down sampled to 22,050 Hz. The supervisor could check the signals via an external loudspeaker and a scope (not shown on the figure).

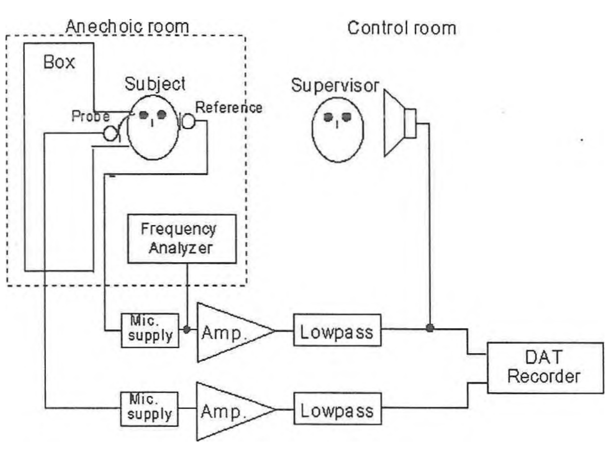



**Figure 4.** Test set up

Figure 4 shows the subject attached to the attenuation box. In the free field situation, the box was taken out of the anechoic room. As only one probe microphone was available it was necessary to have the subject repeat the same sound twice. Continuous speech (10 Harvard sentences) and the cardinal vowels /i/, /a/ and /u/ were used. The challenge here was to maintain identical conditions (formant frequencies and levels) for the two cases, with and without the attenuation box in place. The subject could look at the monitor of a simultaneous frequency analyzer, where the fundamental frequency of the first pronounced vowel was stored and the subject then had to hit the same fundamental frequency again. A successful reproduction required a few tries every time.

The system was calibrated using a Brüel & Kjaer calibrator Type 4231. The calibrator tone (94.0 dB ± 0.2 dB at 1 kHz) was fed into a 1/2" microphone, followed by a 70 Hz highpass filter and a 9.0 kHz lowpass filter and into measuring amplifier, with an amplification of 30 dB. Digital signals were then calibrated by calculating the rms-value of the calibration signal in the time domain and setting this value equal to 94.0 dB.

## 4.2. Attenuation Box Design

A critical element of the experimental apparatus was the attenuation box. This box provides sound isolation so that the air conduction path to the test ear is attenuated, leaving the body conducted signal as the dominant signal in the ear canal. The volume of the box is large (over 0.25 $m^3$) so that there will be no occlusion effect in the test ear.

The box was made of 2 cm thick plywood panels, with height of 104 cm, width of 54 cm and depth of 54 cm. The outside and inside of the box were covered with 10.2 cm (4") fiberglass in order to minimize reflections. A hole was cut in the front panel of the plywood box and a rectangular Plexiglas tube installed. This tube was 15.5 cm x 15.5 cm and extended out 10.5 cm from the box. A removable front plate, also made of Plexiglas, was made to fit to the end of the tube. A doughnut cushion was attached to this front plate. For an experiment, the cushion was placed over the subject's ear and the plate attached to the head with Velcro straps. It was then easy to place the probe microphone in the ear canal and buckle the plate to the box. It was assured that the plate was attached tightly to the box.

The cushion was fit around the pinna, so that the effect of the pinna on the radiated sound was essentially the same as for an uncovered ear. The effect of the box on the radiation from the ear canal is insignificant at frequencies below about 1.3 kHz.

Measurement of the attenuation provided by the box was not trivial and could not be made with a head-and-torso simulator because it could not be attached to the box. In order to get as close as possible to the experimental situation, where the subject would be vocalizing, a small loudspeaker was placed close to the mouth and chin of the subject. The speaker was placed either at a $0^o$ or at a $45^o$ angle to the subject's nose and the speaker center was raised to mouth height. A white noise test signal was used, giving 74 dB SPL at the position of the subject. The probe tube was inserted into the subject's ear canal to a depth corresponding to a standard earmold made especially for that subject (about 8 mm + 3 mm). The sound pressure was measured first without the attenuation box. This simulated free field situation was performed in the empty chamber, anechoic above 200 Hz. The box was then placed in the chamber and the test ear attached to the box. The box attenuation was calculated as the difference in sound pressure level between the free field condition and the boxed ear condition.

The box attenuation was about 5 dB different for the two speaker positions with the $45^o$ position providing the larger attenuation. Averaging the $45^o$ and $0^o$ results, we find that the box attenuates sound by **more than 25 dB** for frequencies between 250 Hz and 3 kHz.

### 4.3. Pressure from the Cushion

The subject had to lean against the cushion to avoid leakage between the head and the cushion, but there is a risk that the mechanical motion of the bones and tissue will be different than if the head was not attached to the cushion. The effect of the cushion pressure was measured with the subject sitting in the anechoic room. The front plate was detached from the box and strapped on the subject's head. The sound pressure in the ear canal was measured while the subject read aloud with the front plate either very loose, but still without leak, or very tight. The difference in sound pressure between loose and tight attachment is a measure of the effect of cushion pressure. If the ear canal is open, the difference is 2.5 dB or less below 2 kHz. This is comparable to the error of repeatability so cushion pressure effects may be ignored.

### 4.4. Reference Microphone

The reference microphone was positioned at the non-boxed ear at the top of the pinna and 1 cm out so it did not touch the skin. The effect of moving the reference microphone a few centimeters upwards, downwards and sideways was checked using a KEMAR mannikin in an empty anechoic room. A speaker was positioned 3 cm in front of the mouth of the mannikin with the centerline in mouth height. Displacement of the microphone 2.5 cm downwards and 1 cm backwards gave the largest measured difference of +1.25 dB at f < 2 kHz.

The disturbance of the box was also measured, using the same set up. The signals picked up by the reference microphone in the free field conditions were compared to those with the box in place. At frequencies below 1 kHz, the disturbance was of the order of measurement error.

## 5. RESULTS

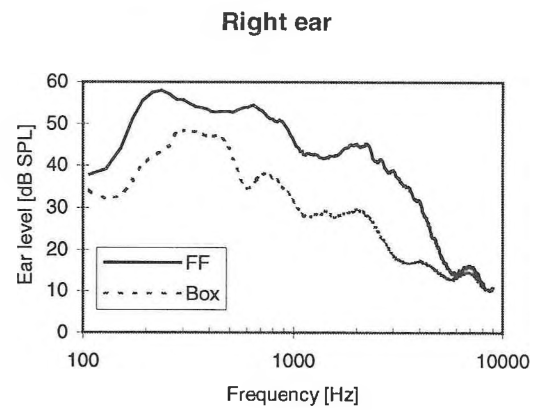
### 5.1. Continuous Speech

The long-term power spectrum of continuous speech was calculated with the Welch method:

$$P_{xx}(\omega) = \frac{1}{K}\sum_{i=1}^{K}\left|\sum_{n=0}^{M-1} x^i(n)w(n)e^{-j\omega n}\right|^2 ; i = 1,2,\ldots,K \quad (5)$$

where $K$ is the number of windows, $M$ is the window length (512 samples) and $w$ is a Hanning window. An overlap of 50% was used. The optimal overlap is 75% but to save process time the overlap was reduced.

Spectra for the probe microphone recordings are shown in Fig. 5 for both left and right ears, comparing spectra with the attenuation box in place and with no box. Spectra are calibrated to correspond to dB SPL and have been corrected for the probe microphone frequency characteristics.



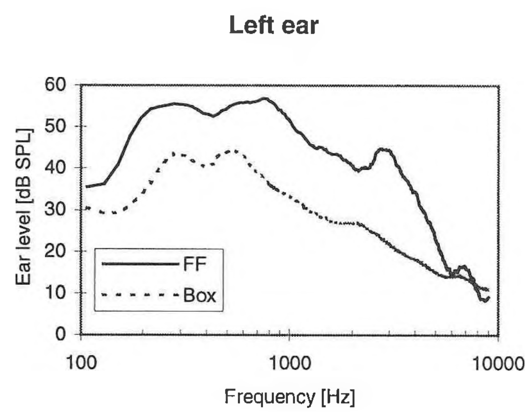




**Figure 5.** Real ear spectra using continuous speech. FF: free ear, Box: boxed ear. The spectra in each panel have not been normalised to account for differences in speech levels.

The measurements obtained for boxed ear and free field conditions have not been measured simultaneously so we must consider that the subject may have spoken with a different rms-level in the two situations. Therefore, it is necessary to normalize the boxed ear sound pressure level with respect to the free ear level, using the difference in sound power of the reference microphone signals. The normalization factor used is the power ratio,

$$\text{ratio} = \frac{1/2\pi \sum S_{\text{boxed}}(n)}{1/2\pi \sum S_{\text{freefield}}(n)} \tag{6}$$

where $S_{\text{boxed}}$ is the spectrum measured by the reference microphone when one ear is boxed and the other open, $S_{\text{freefield}}$ is the reference spectrum when both ears are open (and the anechoic room empty, the simulated free field condition), and $n$ is the frequency index.

With the above normalization, the ratio between the sound pressure in the open ear and in the boxed ear was calculated using Eq. (4). The ratio $P_C/P_A$ is shown in Fig. 6 for the left and right ear of one subject.

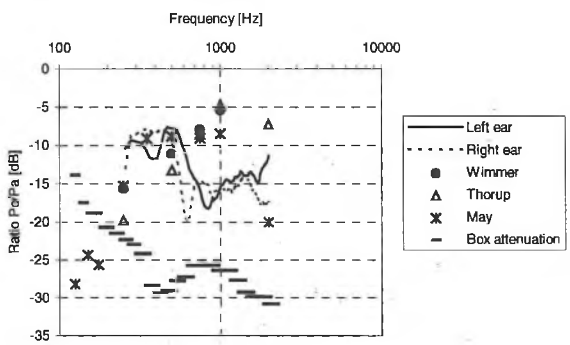



**Figure 6.** Ratio of body conducted and air conducted sound using continuous speech (1 subject). Measured ratios are compared with estimated values derived from literature data. Speech 1/3 octave smoothed.

Included on this figure is the measured box attenuation. Between 250 Hz and 2 kHz, the box attenuation is at least 6 dB below the pressure ratio. The error due to neglect of the box is then less than 1.3 dB over this frequency range. Below 200 Hz, the measured $P_C/P_A$ is unreliable.

Also shown on Fig. 6 are pressure ratios estimated from occlusion effect data. The estimates make use of a model[18] for computing the function $\beta$ and use Eq. (2) to transform the occlusion effect data of Fig. 3 to predicted ratios $P_C/P_A$. The Wimmer and Thorup data was measured using very tightly fitted earmolds, but the data from May and Dillon was measured with standard earmolds and effect of leakage[19] between the ear canal wall and the earmold had to be accounted for in the calculations. The estimates are consistent with our measurements. In the range 300-600 Hz, the measured data lie in the same range as Békésy's values[2] for loudness (-10 dB to 0 dB). Above 700 Hz the measured ratio is about 5 dB lower than measured by Békésy. In the Békésy experiments, short words were used and these sounds will have most of their energy at the fundamentals of the vowels, e.g. up to about 700 Hz.

The spectra for the right and left ears are similar but not exactly the same. This is due to the anatomical asymmetry that is particularly important for body conducted sound transmission.

## 5.2. Vowels

The ratio $P_C/P_A$ was also measured for the three cardinal vowels /a/, /i/ and /u/. Each vowel was repeated three times. The subject was trained to reproduce the fundamental frequency within 30 Hz precision. The levels were also reproduced. It is known that the speech spectrum changes when a voice is softer or louder than normal speech level[20]. But in the pilot experiment a soft, normal and a loud level vowel did not give a consistent change in the ratio of boxed ear and free ear level. This indicates that the ratio of body to air conducted sound is level independent as long as the sounds are produced in the same way, i.e., whispering might not give the same result.

The differences between the boxed ear and the free field ear levels for the left ear are shown in Fig. 7; evaluations were performed using specific harmonics of the spectra. The differences correspond well to that for continuous speech. The higher harmonics of the /a/ were well reproduced and the ratio corresponds well with the ratio found for continuous speech.

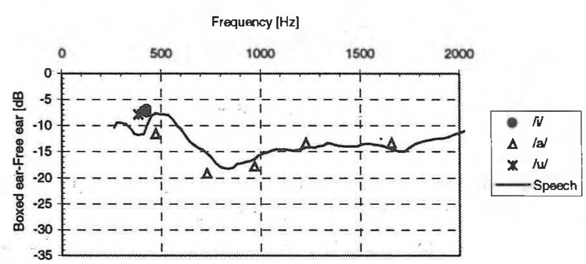



**Figure 7.** Ratio, for specific harmonics, of boxed ear to free ear levels produced by speech and vowels. Left ear. Note, the linear frequency axis. Speech 1/3 octave smoothed.

## 6. DISCUSSION

Several factors affect the accuracy of the speech spectra being evaluated. The reference signal was measured with an uncertainty of 2 dB. Considering all sources of uncertainty, the measured pressure ratio $P_C/P_A$ is believed to be accurate to within 5 dB.

The measurements indicate that a significant fraction of the sound pressure in the ear canal is due to body conduction. To show that the sound pressures generated through body conduction are physically reasonable, the required motion of the canal walls can be estimated using the same ear canal model that was used to estimate $\beta$ earlier. The body source is treated as a piston in the ear canal wall. The volume velocity entering the canal is $q = P_C / Z_C$, where $P_C$ is the pressure generated in the canal and $Z_C$ is the acoustic impedance presented by the canal. The corresponding piston displacement for angular frequency $\omega$ has an amplitude

$$x = q/(A\omega) \quad (7)$$

where the effective piston area is $A$. The result of this calculation is shown in Fig. 8. The displacement at 300 Hz is approximately 0.05 μm. This is much less than the 20% change in diameter of the ear canal that was observed by Oliveira[7] during opening of the mouth. It must be concluded that the measured sound pressures correspond to cartilage displacements that are physically reasonable. The decrease of the displacement with frequency is consistent with the known glottal source spectrum and Eq. (7). It is known that the impedance of soft tissue increases with frequency[21] up to about 15 kHz where it becomes constant.

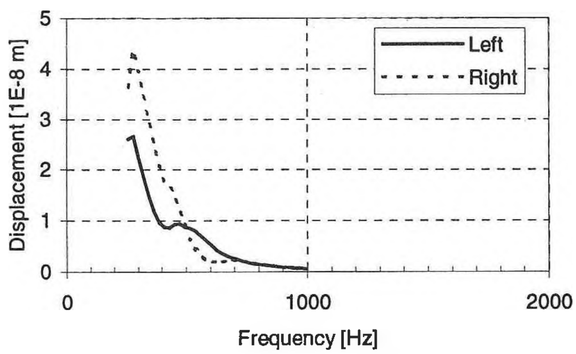



**Figure 8.** Displacement of the ear canal wall for body conducted sound due to own voice. The curves are valid above 250 Hz; at lower frequencies there was inadequate box attenuation. Note the linear frequency axis.

There has been considerable study of the transmission of air borne sound around the head, head-related transfer functions (HRTF). The body conduction path requires body-related transfer functions (BRTF). Sound transmission through muscles and soft tissue at audio frequencies has not been given as much attention. Some measurements of the point skull impedance have been made for bone-anchored hearing aids, e.g., Håkansson and Carlsson[22], but here the bone is directly stimulated and the results cannot be applied directly to speech production. Ishizaka *et al.*[23] measured the mechanical impedance on the cheek and neck in order to estimate the impedance of the vocal tract walls. Unfortunately they only measured up to 160 Hz. The impedance of the skin and soft tissue has been measured as a point impedance in most cases on the finger or the arm[21]. These data, although necessary and useful, are not sufficient in themselves to lead to the development of a model for body conduction of sound.

There are several body conduction routes that need to be addressed in developing a model. For example, consider Fig. 9 in which the open ear spectrum for a vowel sound is compared to the spectrum obtained with the ear closed with the attenuation box. The boxed ear spectrum shows the same resonant structure as the free ear spectrum and the formant (F1 and F2) are somewhat represented in the boxed ear signal too. The formants are an effect of the acoustic of the vocal tract length and the shape of the oral cavities. This comparison indicates that the body conducted sound is produced, not only by vibration of the vocal cords and attached structures, but through pressure changes in the vocal tract and oral cavity.

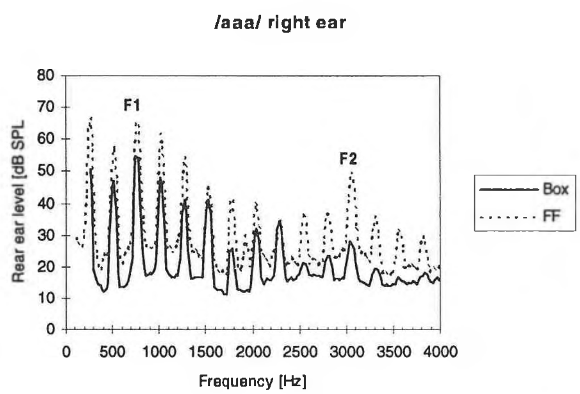



**Figure 9.** Spectrum of /a/ measured in the ear canal on the free ear and boxed ear. Note, the linear frequency axis.

## 7. CONCLUSION

The ear canal sound pressure produced by own voice has been investigated. A pilot study of the ratio between body conducted and air conducted sound was performed. The pilot study provides new information on sound produced by one's own voice. Previous studies have concentrated on bone conductor stimuli.

The measured spectra for vowel sounds indicate that body conducted sound originates from pressure changes in the oral cavity, vocal tract and vibrations of the vocal folds. The observed real ear levels due to vibrations of the soft tissue in the ear canal have been shown to be plausible, but the transmission mechanisms need further investigation.

The measurements here provide important input data for the modelling of the sound transmission in the ear canal and the occlusion effect. This modelling will be the main object of interest in the proceeding of the Ph.D. project[18]. The pilot experiment described in this paper has lead to further study by Carsten Bremmelgaard[24] in a master degree project. In this work, the ratio $P_C / P_A$ is being measured in nearly the same way as in the pilot experiment, using 10 subjects (21 to 33 years).

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*Canadian Acoustics / Acoustique Canadienne 26(2) 25-28 (1998)* **Research article / Article de recherche**

# THE BENEFIT OF CONTEXTUAL CUES FOR THE PERCEPTION OF TEMPORALLY DEGRADED SPEECH

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## ABSTRACT

The effect of contextual cues on the understanding of distorted speech was studied. Two groups of subjects, aged 20-25 and 50-59 years participated. Three lists of 50 sentences that were continuous or interrupted at rates of 8/s and 2/s were presented to each ear. In half the sentences within list, the final word was highly predicable from the context, and in half, poorly predicted. Final word recognition scores decreased with interruption, more so for the 2/s rate. Context was compensatory. Middle-aged subjects did not perform more poorly. Factors possibly accounting for this outcome are discussed.

## SOMMAIRE

L'effet d'indices contextuels sur la compréhension de la parole dégradée a été étudié. Deux groupes de sujets, âgés respectivement de 20-25 et 50-59 ans, ont participé à cette étude. Trois listes de 50 phrases ont été présentées, à chacune des oreilles, de façon continue ou interrompue avec un taux de 8/s ou 2/s. Pour la moitié des phrases de chacune des listes, le dernier mot était hautement prévisible sur la base du contexte et, dans la seconde moitié, peu prévisible. La reconnaissance du mot final décroît avec l'interruption, et ce davantage pour le taux de 2/s. Le contexte s'est avéré compensatoire. Les sujets plus âgés ne performent pas moins bien que les plus jeunes. Les facteurs pouvant potentiellement expliqués ces resultats sont discutés.

## 1.0 INTRODUCTION

Published results from a number of studies support the conclusion that, with aging, there is a slowing of information processing ability (for reviews, see Birren and Schaie, 1977; Fitzgibbons and Gordon-Salant, 1996; Divenyi and Haupt, 1997). This effect is clearly apparent in the perception of degraded speech (Bergman, 1980; Wingfield, 1996). Poorer speech understanding observed in older listeners could be due to a variety of underlying factors, including deficits in the ability to use linguistic cues, memory, selective attention, reaction time, and frequency selectivity (e.g., Working Group on Speech Understanding, 1988; Hutchinson, 1989; Gordon-Salant and Fitzgibbons, 1997). Several experiments have confirmed that the difference limens for duration, frequency and intensity, as well as the interaural time of arrival difference required to perceive laterality increase with aging (Herman et al., 1977; Robin and Royer, 1989; Abel et al., 1990; Schneider et al., 1994; Fitzgibbons and Gordon-Salant, 1996; He et al., 1998).

In a benchmark experiment conducted by Bergman and co-workers in the mid-1970s (Bergman et al., 1976), the understanding of distorted speech was compared in listeners ranging in age from 20-79 years, screened for hearing loss. The effects of reverberation of the listening environment, a competing speaker, and simultaneous presentation of the two parts of two-part words to the right and left ears, as well as low and high pass filtering, time compression, and periodic interruption of the speech materials, were studied in relation to the undistorted condition. Decrements due to aging were observed to some degree under all conditions of distortion. However, the greatest changes in speech understanding occurred when time-processing abilities were stressed, that is, under conditions of reverberant listening, compression and interruption. The understanding of undistorted sentences decreased by about 15% over the age range studied, while periodic interruption resulted in a decrement of about 65% in the number of words correctly recognized. In the case of speech interruption, changes were evident as early as the fourth decade.

A number of studies have demonstrated that the age-related decrement in processing distorted speech is independent of any concurrent presbycusis (Sticht and Gray, 1969; Dubno et al., 1984; Gordon-Salant and Fitzgibbons, 1993). In the study by Sticht and Gray (1969), subjects under and over the age of 65 years, subdivided into groups with either normal or mild to moderate bilateral sensorineural hearing loss, were tested with time compressed speech. Younger and older subjects in each of the two hearing categories were matched in their ability to understand undistorted speech. Hearing loss, aging and percent compression of the speech materials were all statistically significant factors. Importantly, the effects of age and hearing loss were independent.

## 2.0 RATIONALE

The present experiment was conducted to explore whether contextual cues might be used to counteract the effect of speech degradation. Miller and colleagues (Miller and Licklider, 1950; Miller et al., 1951) showed that, in young listeners, word recognition scores improved with familiarity of the test items. Scores decreased as the number of possible alternatives increased. More recently, Hutchinson (1989) compared subjects in their 20s and 60s on their ability to understand words in sentences with either good or poor contextual cues, presented against a babble noise background. Both groups were aided, almost equally, by the availability of contextual cues. Older subjects were more severely affected by the absence of context, particularly as the speech to noise ratio decreased.

## 3.0 EXPERIMENTAL DESIGN

In the present study, speech degradation was accomplished by means of periodic interruption. This method was chosen because of the demonstration that it provides the earliest indicator of the age effect (Bergman, 1983). Electronic interruption of speech has previously been investigated in detail by Miller and Licklider (1950) for young normal listeners. In their study, monosyllabic words were interrupted at regularly spaced intervals ranging from 0.1/s to 10,000/s, with an on-off fraction (duty cycle) of 0.5. Word recognition improved from 50% at 0.1/s to near 100% at 100/s. Given a constant interruption rate, performance improved as the speech on time increased.

The study by Bergman et al. (1976) utilized an interruption rate of 8/s. In later work, Gordon-Salant and Fitzibbons (1993) were unable to show an aging effect with an interruption of 12.5/s. In the present experiment, two values were chosen, 8/s and 2/s, to maximize the likelihood that time-processing abilities would be stressed. Performance under these two conditions was compared with continuous undistorted speech. Speech materials were presented in quiet to preclude possible confounding by age-related differences in the perception of acoustic cues with masking (Hutchinson, 1989). Right and left ears were tested independently. The nontest ear was fitted with a sound attenuating ear plug. Previous research by Johnson et al. (1979) showed an age-related decline in dichotic memory for spoken digits presented to the left ear but not the right ear. This difference was attributed to hemispheric differences in the encoding of language.

## 4.0 METHODS AND MATERIALS

### 4.1 Subjects

Two groups of eight subjects, aged 20-25 years (1 male and 7 females) and 50-59 years (2 males and 6 females) participated. All had normal hearing thresholds of less than 10 dB HL in both ears at 2 kHz. This frequency has been shown to be a good predictor of speech perception in cases of mild hearing loss (Abel, 1993).

### 4.2 Apparatus

The apparatus has been described previously (Abel et al., 1990). Each subject was tested in a sound proof IAC booth with ambient noise levels less than the maximum recommended for headphone testing (ANSI, 1991). The SPIN test (Bilger et al., 1984) used for the experiment was commercially available on audio cassette. Lists of sentences pre-recorded by a male American speaker in quiet, were presented to the subject monaurally over a TDH-39 matched headset by means of a Nakamichi Bx-125 Cassette Deck. A sustained 1-kHz tone at the start of the tape, recorded at a level equivalent to the rms value of the speech, allowed calibration to a comfortable listening level of 75 dB SPL. The interuption rate of the speech (2/s or 8/s) was controlled by means of a Hewlett-Packard 3325A Synthesizer/Function Generator. The duty cycle was fixed at 0.5. The rise/decay time of speech segments was 10 ms.

### 4.3 Procedure

Eight alternative lists were available. Each comprised 50 sentences. In half the sentences, the final word was highly predictable from the preceding context (e.g., THE SLEEPY CHILD TOOK A <u>NAP</u>), and in half, the final word was poorly predictable (e.g., THE CLASS SHOULD CONSIDER THE <u>FLOOD</u>). Each subject was presented six lists, three to the right ear and three to the left, with either no interruption (i.e., continuous speech) or with interruption rates of 2/s and 8/s, respectively. The continuous condition was always given first to maximize practice. The subject's task was to write down the final word in each sentence. Guessing was encouraged, in case of uncertainty. In half the subjects within each group, the right ear was tested first, and in half, the left. The order of the two interruption rates and choice of list for each of the six listening conditions was counter-balanced across subjects in the group. No list was presented twice within subject to control familiarity.

## 5.0 RESULTS

The results of the experiment are presented in Table 1. Mean correct high and low context final word recognition scores, each based on 25 sentences, are shown for each of the continuous and interrupted conditions by group and ear. A nested analysis of variance with repeated measures on ear, context, and interruption rate indicated that context, interruption rate and their interaction were statistically significant ($p<0.001$). There was no effect of age or ear. Post hoc pairwise comparisons using Fisher's LSD method (Daniel, 1983) confirmed that for the high context sentences, there was no difference between speech presented continuously or with an interruption rate of 8/s. Both conditions gave significantly higher scores than the 2/s condition. In the case of the low context sentences, the three levels of interruption were significantly different from each other. For both the 8/s and 2/s rates, scores were significantly higher with high compared with poor contextual cues ($p<0.05$).

## 6.0 DISCUSSION

The results of the experiment indicated that interrupted speech understanding in subjects aged 50-59 yrs was no different than that of subjects aged 20-25 yrs. This is in contrast to Bergman's (1976) finding that the effect of degrading speech in the time domain was evident by the fourth decade. It is possible that the negative outcome was due to the difference in speech materials and task in the two studies, or the small sample size and possible lack of sensitivity of the test in the present study. In contrast, the significance of rate of interruption and contextual cues were clearly apparent. For the two groups combined, the availability of context significantly improved the outcome by 26% in the case of the 8/s interruption rate and by 19% for the the 2/s interruption rate. Collapsed across the two levels of context, mean percent correct scores, 78% and 38% for 8/s and 2/s respectively, were comparable to the results reported by Miller and Licklider (1950).

Findings of Wingfield and coworkers (see Wingfield, 1996) may shed light on the nature of the context effect. Wingfield et al. (1985) presented subjects, aged 18-22 yrs and 65-73 yrs, with normal sentences, syntactic word strings (i.e., word strings without meaning but with syntactic form), and random word strings. The percentage of words correctly recognized decreased, as the presentation rate increased from 275 to 425 words/min. Both groups achieved at least 80% correct, if given normal or syntactic strings but performance deteriorated, particularly for the older group, when syntax was eliminated. Thus, when time processing ability was stressed, both groups performed well, as long as syntactic cues were available. In the present study, language structure was always preserved, regardless of the interruption rate. Both groups gained from the linguistic cues provided by context.

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**Table 1**

Effects of age, interruption rate, ear and contextual cues on word recognition.

| | | Context/Ear | | | |
|---|---|---|---|---|---|
| Age | Interruption | High | | Low | |
| (yrs) | Rate | Left | Right | Left | Right |
| 20 - 25 | Cont. | 24.9 (0.4) | 25.0 (0.0) | 23.9 (0.8) | 24.4 (1.2) |
| | 8/s | 23.5 (1.5) | 23.6 (2.1) | 16.3 (3.0) | 17.0 (2.1) |
| | 2/s | 11.8 (5.1) | 13.9 (5.2) | 7.6 (2.5) | 7.5 (3.0) |
| 50 - 59 | Cont. | 24.9 (0.4) | 25.0 (0.0) | 23.8 (1.2) | 23.1 (1.4) |
| | 8/s | 21.1 (4.2) | 22.6 (2.6) | 15.3 (3.5) | 16.4 (2.1) |
| | 2/s | 10.9 (4.4) | 10.9 (3.4) | 5.5 (2.9) | 6.9 (2.6) |

Mean raw score (1 S.D.), 25 items

*Canadian Acoustics / Acoustique Canadienne 26(2) 29-31 (1998)* **Technical note / Note technique**

# THE ELUSIVE DECIBEL: THOUGHTS ON SONARS AND MARINE MAMMALS

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## INTRODUCTION

A few years ago, there was considerable controversy over the effects of a proposed global acoustic experiment designed to measure the temperature of the world's oceans[1]. The focus of concern was the possible effect of the acoustic signals on whales and other marine life. There is continued interest in the effects of underwater sound on marine animals, according to a recent news item in *The Economist*[2] based on related scientific correspondence in *Nature*[3]. The thesis is that loud signals from experimental sonars harm marine mammals, or at least harass them enough to unacceptably alter their behaviour patterns. In the various discussions of this important issue that can be found in the press and on the internet, one often sees questionable comparisons being made, such as the acoustic output of a naval sonar being compared with the noise from a jet aircraft. Some misunderstandings between professionals in different fields can be traced to the multiple uses of the term "decibel". Acoustical terms can be confusing, even for experts. It is not at all surprising that well-intentioned articles sometimes fail to present situations clearly. By definition, the decibel is a relative unit, not an absolute unit with a physical dimension; unless the standard of comparison is cited, the term "decibel" is to all intents and purposes useless. The confusion is not helped by the use of the decibel to specify distinctly different physical quantities, or the same physical quantity with different reference levels. Some reporters—and even some scientists—are getting their "apple" decibels mixed up with their "orange" decibels, as it were.

The decibel (abbreviated dB) is simply a numerical scale used to compare the values of like quantities, usually power or intensity. Acousticians introduced the decibel to devise a compressed scale to represent the large dynamic range of sounds experienced by people from day to day, and also to acknowledge that humans—and presumably other animals—perceive loudness increases in a logarithmic, not linear, fashion. An intensity ratio of 10 translates into a level difference of 10 decibels[4]; a ratio of 100 translates into a level difference of 20 dB; 1000 into 30 dB; and so on. (The term "level" usually implies a decibel scale.) In a uniform acoustic medium, the magnitude of the acoustic intensity is proportional to the square of the pressure for a freely-propagating sound wave. Accordingly, the level difference


in decibels associated with two sound pressure values (measured in the same medium) is determined by calculating the ratio of the pressures, squaring this number, taking the logarithm (base 10), and multiplying by 10.[5] If one chooses a standard reference pressure value, then sound pressure levels can be specified in decibels relative to that reference, but this should be stated along with the number, for clarity[6].

The following is a typical erroneous statement found in the press, on radio, on television, and on internet discussion groups. Referring to an experimental sonar source that produces very loud low-frequency sound, *The Economist* wrote: "It has a maximum output of 230 decibels, compared with 100 decibels for a jumbo jet." Regardless of the author's intention, the implication is that a whale would experience an auditory effect from the sonar that would be substantially greater than that of a person exposed to the jet aircraft. However, this type of comparison is misleading for at least three reasons: (1) the reference sound pressures used in underwater acoustics and in-air acoustics are not the same; (2) it compares a source level with a received level; and (3) there is no obvious connection between an annoying or harmful sound level for a human in air and an annoying or harmful sound level for a marine animal in water. In the remainder of this note, we will expand on these topics somewhat, attempt to correct the mistaken impression, and try to direct attention to the real issue at the heart of the controversy.

## 1. STANDARD REFERENCE SOUND PRESSURES IN AIR AND IN WATER

The standard reference pressures used in underwater acoustics and in-air acoustics are not the same. In water, acousticians use a standard reference sound pressure of 1 micropascal (i.e. $10^{-6}$ newtons per square metre), abbreviated μPa. In air, acousticians use a higher standard reference sound pressure of 20 μPa. The in-air standard was chosen so that the threshold of hearing for a person with normal hearing would correspond to 0 dB at a frequency of 1000 Hz. Adopting different standards for air and water inevitably leads to a confusing consequence: the same sound pressure that acousticians label 0 decibels in air would be labelled 26 decibels in water. Presumably, both factions of acousticians had equally-good reasons for proposing their respective standards, and this dichotomy is now entrenched in an ANSI standard[4], which is unlikely to change. Accordingly, the following dictum should always be observed, especially when dealing with cross-disciplinary issues: *It is essential that sound levels stated in decibels include the reference pressure.*

## 2. SOURCE LEVEL AND RECEIVED LEVEL

The erroneous statement compares a *source* level with a *received* level. In underwater acoustics, a source level usually represents the sound level at a distance of one metre from the source, while a received level is the sound level at the listener's actual position, which could be considerably more distant with a correspondingly reduced sound level. In an unbounded uniform medium, loudness decreases rapidly with increasing source-receiver distance, 6 dB less per doubling of distance. For example, *The Economist* (and even *Nature*), in referring to the 230 dB sonar source level, neglected to mention the reference distance of 1 metre. In contrast, the 100 dB number that *The Economist* associated with a jumbo jet is not a source level at all, but is typical of a received noise level measured during jet airplane take-off, averaged over several microphones situated several hundred to some thousands of metres from the runway[7]. *It is incorrect to compare a source level at 1 metre with a received noise level at an unspecified (and probably much larger) distance.*

Combining these two remarks, the output of the sonar source should have been written as 230 dB re 1 μPa at 1 m, while the jumbo jet noise level should have been written as 100 dB re 20 μPa. The inclusion of the reference values shows that these are not like quantities, and that the numbers are not directly comparable. *The Encyclopedia of Acoustics*[8] offers 120 dB re 20 μPa as a typical noise level associated with jet aircraft take-off measured at 500 m distance (although there is sure to be a wide variation about this number, depending on the type of aircraft, etc.). With the assumption of spherical spreading, referencing this level back to 1 metre distance adds 54 dB. Switching to the 1 μPa standard reference adds another 26 dB. Accordingly, the source level of a large jet looks more like 120 + 54 + 26 = 200 dB re 1 μPa at 1 m, compared with 230 dB re 1 μPa at 1 m for the sonar. Both of these are loud sources, but now at least the comparison is sensible. The ratio of sound pressures is around 32, rather than over 3 million, as some commenters would have you believe!

There are other minor issues that could be discussed. The signal from the sonar source is narrowband, and the concentration of all the signal at one frequency may be particularly troublesome for an animal who has a cavity that resonates at that frequency. On the other hand, the jet noise is broadband, and the acoustic signal was probably passed through a filter that approximately matches the sensitivity of the human ear before the measurement was made, so this measurement would be meaningless for an animal with a different hearing sensitivity curve. Much more could be said about these issues, but the principal reason for raising them is to underscore the message that the sonar / jet plane comparison has little validity.

## 3. WHAT HURTS?

There is no clear connection between a harmful sound level for a human in air and that for an animal in water. All creatures have evolved and adapted to their respective environments and there is no reason why human hearing characteristics should apply to any other animal, including whales. If a given sound pressure hurts a human, would the same sound pressure level in water hurt a whale (or a fish, or a shrimp)? Is the threshold of pain higher? Is it lower? Particularly when comparing acoustic effects in media of widely different impedance, is acoustic pressure the relevant acoustic quantity, or is it acoustic intensity?[9] In the end, it is the answers to these and related questions that really matter, not juggling decibels. To properly answer these questions and to determine the "community" noise standards for marine animals, scientific research is necessary—just as it was for humans. Some of this work has already been done, and an excellent review[10] of the state of knowledge up to 1995 is a good starting point for acousticians and biologists interested in deepening their understanding. A single example cannot represent the whole range of species under consideration, but is typical: The response threshold (determined through behavioural studies) of a Beluga at 1000 Hz is just over 100 dB re 1 μPa. Of course, this says nothing about the Beluga's threshold of pain, and says nothing about what sound level would unacceptably alter its behaviour. *It is unwise to assume that the auditory experience of any animal would be the same as that of a human exposed to the same sound level.*

## CONCLUSION

As sonar engineers, marine biologists, and environmentally conscious citizens continue to discuss these important issues, we should at least agree to use the same acoustical units to convey our points of view, to avoid confusion and misrepresentation. Some sensible acousticians have advocated abandoning the use of the decibel—which is partly to blame for our woes—in favour of good old SI (i.e., metric) units for sound pressure, acoustic intensity, power, etc. Until that happy day dawns, let us include reference values with our decibels, so we don't end up with fruit salad dBs. Ultimately, what is important is to determine what underwater sound levels are harmful to marine life. We must develop mitigation measures to allow underwater acoustic systems to be operated while ensuring the protection of the marine environment with due diligence.


## ACKNOWLEDGEMENT

The authors thank Harold M. Merklinger for his helpful comments on the manuscript.


## REFERENCES


[1] Whitlow W.L. Au *et al.*, "Acoustic effects of the ATOC signal (75 Hz, 195 dB) on dolphins and whales", J. Acoust. Soc. Am. **101**, 2973–2977 (1997).

[2] "Quiet, please. Whales navigating", *The Economist*, 1998 March 7, page 85.

[3] R. Frantzis, "Does acoustic testing strand whales?", *Nature* **392**, 1998 March 5, page 29.

[4] In fact, this defines 1 bel, named after Alexander Graham Bell. The bel turned out to be too large for practical purposes and the decibel—which is 1/10 of a bel—is the preferred unit. Also, one decibel is about the smallest incremental change of sound pressure level a person can sense.

[5] Mathematically, this is equivalent to taking the logarithm of the pressure ratio and multiplying by 20, but knowing when to multiply by 10 or 20 in such calculations is an endless source of confusion to the neophyte, so we advocate the definition in the main text.

[6] *American National Standard Preferred Reference Quantities for Acoustical Levels*, ANSI S1.8-1969, page 8.

[7] Malcolm J. Crocker, editor, *The Encyclopedia of Acoustics* (John Wiley and Sons, Inc., New York, 1997), page 1095.

[8] Malcolm J. Crocker, editor, *The Encyclopedia of Acoustics* (John Wiley and Sons, Inc., New York, 1997), page 11.

[9] The suggestion that acoustic intensity has more bearing than sound pressure in this context has been seriously proposed by some acousticians; however, the available evidence gives the nod to sound pressure, not intensity.

[10] W. John Richardson *et al.*, *Marine Mammals and Noise* (Academic Press, New York, 1995).


***Erratum*** ***Software review / Revue de logiciel***

# WinSAl-V and Speechlab
## Media Enterprise - Ingolf Franke, Manager

In the review of this software published in Canadian Acoustics 26(1) 15, 24 (1998), incorrect information was published regarding obtaining the software.

The software packages are available on separate CDs. They can be ordered from Media Enterprise, Gottbillstrasse 34 a, D-54292 Trier, Germany; e-mail: office@media-enterprise.de. Price: DM 499 for WinSAL-V, DM 130 for Speechlab (student discount available).

# APPEL DE COMMUNICATIONS

**Semaine canadienne d'acoustique 1998**
**L'hôtel Westin, London, Ontario**
**28-30 octobre, 1998**

Cette année, tous les domaines de l'acoustique et des vibrations seront offertes à la Semaine canadienne d'acoustique. Des sessions techniques portant sur les thèmes sont déjà planifiées. En voici la liste:

Acoustique Architecturale
Acoustique Musicale
Acoustique Sous-marine
L'âge et l'audition
Contrôle du Bruit
Contrôle du Bruit en Milieu de Travail
Contrôle du Bruit Industriel
Contrôle du Vibration
Psycho-acoustique developmentale
Parole
Contrôle du Bruit de l'Aéroport et des Aéroplanes

Les présentations soumises seront réparties dans les sessions précédentes ou dans d'autres sessions. Les soumis d'autres thèmes des sessions techniques sont également encouragés.

Pour soumettre une demande de session techniques:

- Veuillez contacter Dr. Cheesman avant le 5 juin, 1998, à l'adresse ci-dessus.

Pour soumettre une présentation:

- Envoyez un résumé d'un maximum de 250 mots au responsable technique **avant le 5 juin 1998**. Cette échéance doit être scrupuleusement respectée. Les résumés doivent être préparés en suivant les instructions incluses dans ***Acoustique canadienne*** et à **http://www.uwo.ca/hhcru/caa/awc98/instr.html** .

- Une notification d'acceptation du résumé sera envoyée aux auteurs avant le 15 juin 1998 avec un formulaire d'inscription et les instructions pour préparer le sommaire .

- Un sommaire, préparé suivant les instructions, doit être envoyé au responsable technique **avant le 15 août 1998**. Cette échéance doit être scrupuleusement respectée. Les sommaires seront publiés dans les actes du Symposium.

Veuillez faire parvenir les résumés et les sommaires à:

Dr. Margaret F. Cheesman
Hearing Health Care Research Unit, School of Communication Sciences and Disorders
The University of Western Ontario, Elborn College, London, ON N6G 1H1
tél: (519) 661-3901; fax: (519) 661-3805 courrier électronique: cheesman@audio.hhcru.uwo.ca

Chambres d'hôtel et frais d'inscription: Toutes les activités au programme auront lieu au hôtel Westin, London. L'hôtel Westin offre un tarif special aux inscrits. Tous sont encouragés à soummettre le formulaire d'inscription en avance du symposium. Les frais d'inscription et le formulaire d'inscription doivent être remplis et envoyés avec le sommaire.

Résumé des dates importantes:

| | |
|---|---|
| 5 juin 1998 | Date limite de réception des résumés |
| 15 juin 1998 | Notification d'acceptation |
| 15 août 1998 | Date limite de réception du sommaire, du formulaire d'inscription et des frais d'inscription |
| 28-30 octobre 1998 | Symposium |

Concours étudiants: la participation des étudiants au symposium est fortement encouragée. Des prix seront décernés pour les meilleures communications. Les étudiants doivent inscrire en remplissant le formulaire ***"Prix annuels relatifs aux communications étudiantes"*** et en le remettant avec le résumé.

## ANNUAL STUDENT PRESENTATION AWARDS

The Canadian Acoustical Association makes awards to students whose papers are presented at the CAA Annual Symposium. Students contemplating presenting papers at the Symposium should apply for these awards with the submission of their abstract.

### RULES

1. These awards are presented annually to authors of outstanding student papers that are presented during the technical sessions at Acoustics Week in Canada.
2. In total, three awards of $500.00 are presented.
3. Presentations are judged on the following merits:
   - i) The way the subject is presented;
   - ii) The explanation of the relevance of the subject;
   - iii) The explanation of the methodology/theory;
   - iv) The presentation and analysis of results;
   - v) The consistency of the conclusions with theory and results.
4. Each presentation is judged independently by at least three judges.
5. The applicant must be:
   - i) a full-time graduate student at the time of application;
   - ii) the first author of the paper;
   - iii) a member of the CAA;
   - iv) registered at the meeting.
6. To apply for the award, the student must send this application simultaneously with the abstract. Multiple authors are permitted, but only the first author may receive an award.

## PRIX ANNUELS RELATIFS AUX COMMUNICATIONS ETUDIANTES

L'Association Canadienne d'Acoustique déceme des prix aux étudiant(e)s qui présenteront une communication au congrès annuel de l'ACA. Les étudiant(e)s qui considèrent présenter un papier doivent s'inscrire à ce concours au moment où ils (elles) soummettent leur résumé.

### REGLEMENTS

1. Ces prix sont décernés annuellement aux auteurs de communications exceptionelles presentées par des étudiants lors des sessions techniques de la Semaine Canadienne d'Acoustique.
2. Au total, trois prix de 500$ sont remis.
3. Les présentations sont jugées selon les critères suivants:
   - i) La façon dont le sujet est présenté;
   - ii) Les explications relatives à l'importance du sujet;
   - iii) L'explication de la méthodologie;
   - iv) La présentation et l'analyse des résultats;
   - v) La consistence des conclusions avec la théorie et les résultats.
4. Chaque présentation est evaluée séparément par au moins trois juges.
5. Le candidat doit être:
   - i) un étudiant à temps plein de niveau gradué au moment de l'inscription;
   - ii) le premier auteur du papier;
   - iii) un membre de l'ACA;
   - iv) un participant au congrès.
6. Afin de s'inscrire au concours, l'étudiant doit envoyer ce formulaire d'inscription en même temps que son résumé. Plusieurs auteurs sont permis, mais seul le premier auteur peut recevoir le prix.

---

## APPLICATION FOR STUDENT PRESENTATION AWARD AT ACOUSTICS WEEK IN CANADA

## FORMULAIRE D'INSCRIPTION POUR LES PRIX DECERNES AUX ETUDIANTS LORS DE LA SEMAINE CANADIENNE D'ACOUSTIQUE

NAME OF THE STUDENT:____________________NOM DE L'ETUDIANT

SOCIAL INSURANCE NUMBER:____________________NUMERO D'ASSURANCE SOCIALE

TITLE OF PAPER:____________________TITRE DU PAPIER

UNIVERSITY/COLLEGE:____________________UNIVERSITE/COLLEGE

NAME, TITLE OF SUPERVISOR:____________________NOM ET TITRE DU SUPERVISEUR

STATEMENT BY THE SUPERVISOR: The undersigned affirms that the above-named student is a full-time student and the paper to be presented is the student's original work.

Signature:____________________

DECLARATION DU SUPERVISEUR: Le sous-signé affirme que l'étudiant(e) mentionné(e) ci-haut est inscrit(e) à temps plein et que la communication qu'il (elle) présentera est le fruit de son propre travail.

Date:____________________

## APPLICATION FOR STUDENT TRAVEL SUBSIDY TO ACOUSTICS WEEK IN CANADA

Travel subsidies are available to students presenting papers at Acoustics Week in Canada if they live at least 150 km from the conference venue, if the subsidy is needed, if supporting receipts are submitted, and if they publish a summary of their paper in the proceedings issue of *Canadian Acoustics*.

I wish to apply for a CAA Travel Subsidy: _____yes _____no.

STATEMENT BY THE SUPERVISOR: The undersigned affirms that the CAA Travel Subsidy, combined with other travel funds that the above-named student may receive to attend the meeting will not exceed his/her travel costs.

Signature:____________________

## FORMULAIRE DE DEMANDE DE REMBOURSEMENT POUR FRAIS DE DEPLACEMENT A LA SEMAINE CANADIENNE D'ACOUSTIQUE

Un remboursement de frais de déplacement est offert aux étudiants qui présentent une communication lors de la Semaine Canadienne d'Acoustique, s'ils demeurent à plus de 150 km du site du congrès, si le remboursement est nécessaire, si les reçus à l'appui sont soumis et s'ils publient un résumé dans les Actes du Congrès.

Je désire demander un remboursement: _______oui _______non.

DECLARATION DU SUPERVISEUR: Le sous-signé affirme que le remboursement, jumelé à d'autres fonds que l'étudiant(e) ci-haut mentionné(e) peut recevoir ne dépasseront pas ses coûts réels de voyage.

Date:____________________

## Canadian Acoustical Association
## l'Association Canadienne d'Acoustique

# Minutes of the CAA Board of Director's Meeting
**8 May 1998**
**Toronto, Ontario**

| | | | |
|---|---|---|---|
| *Present:* | John Bradley | John Hemingway | Trevor Nightingale |
| | Murray Hodgson | Don Jamieson | David Quirt |
| | Winston Sydenborgh | Annabel Cohen | Dalila Giusti |
| *Regrets:* | Jean Nicolas | Li Cheng | Stan Dosso |

*Meeting called to order at 11:02 a.m.*

Minutes of the October 97 Board of Director's Meeting as printed in Canadian Acoustics were reviewed. (Murray Hodgson moved that they be accepted as written, seconded by Winston Sydenborgh, and passed).

### President's Report (John Bradley)

The survey conducted by the President (published in the March '98 issue of Canadian Acoustics) indicated that by far the most important functions of the Association were publishing the journal Canadian Acoustics, and the annual conference. There was also considerable discussion regarding the relative importance of prizes and local chapter meetings. (David Quirt moved to accept the President's report, seconded by Murray Hodgson, passed).

### Secretary's Report (Trevor Nightingale)

The Secretary reported that membership levels appear to be stable with 349 paid memberships as of 01 May 1998 which is an increase of 2 over the same time last year. Correspondence with the Chair of the 1997 Windsor Conference indicated that a profit of $1724.71 was realized although this money and the $2000 seed money have not yet been returned to CAA. The Secretary has created a set of guidelines to make the task of organizing future CAA conferences easier. Secretarial operating costs during the first six months were $495.05, comparable to those incurred during the first six months of '97. (Don Jamieson moved to accept the Secretary's report, seconded by Murray Hodgson, and passed).

### Treasurer's Report (John Hemingway)

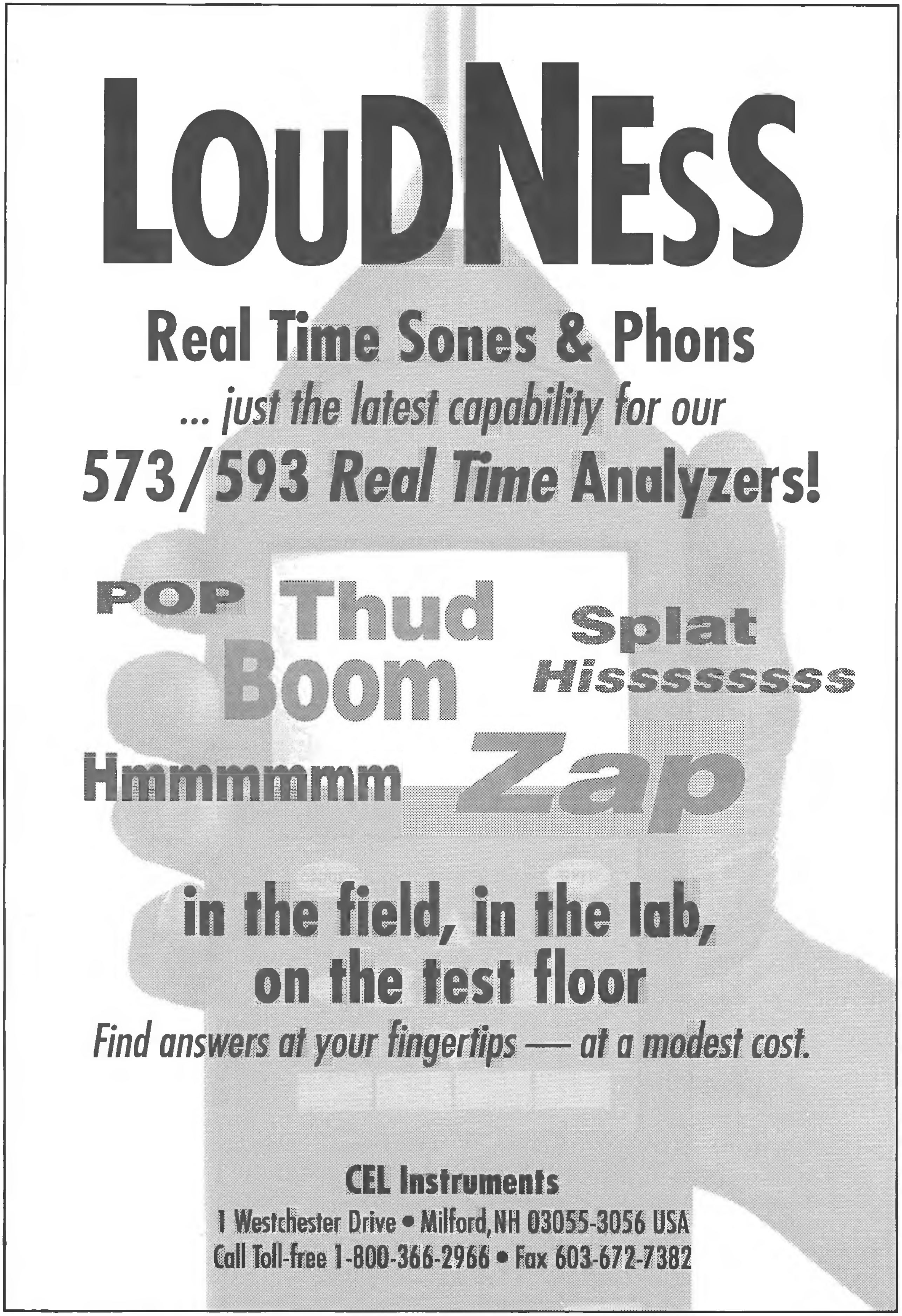
The Treasurer thanked the past Treasurer, Sharon Abel, for her hard work over the past two years. The Treasurer reported that the finances seem to be solid with revenues expected to exceed expenditures by about $12,000 for this fiscal year. It was also reported that the Operating Fund has significantly more money than is required for normal annual expenditures and that money over and above what is required for normal operating costs (such as printing of the Journal) should be transferred to the Capital Fund. The Treasurer recommended transferring $20,000 from the Operating Fund to the Capital Fund. This would be discussed at the October BoD, meantime the Treasurer would be investing $30,000 of the Operating Fund in short term GIC's that would mature in October '98 in time for a transfer, if need be. The Treasurer moved that, "Paul Busch, of 5780 Timberlea Blvd., Suite 207, Mississauga, Ontario, L4W 4W7, be appointed Auditor of the Canadian Acoustical Association, with an annual budget of $1,500 (plus GST) effective immediately." The motion was seconded by Murray Hodgson and passed. (Don Jamieson moved to accept the Treasurer's report, seconded by David Quirt, and passed).

### Membership Report (Don Jamieson)

It was reported that it is very difficult to gauge if efforts to increase the membership are successful. It was agreed that the CAA web page was an excellent way of increasing our exposure and should be continued and expanded, if possible. It was reported that the CAA brochure has been updated and that the awards brochure will also be revised. (Trevor Nightingale moved to accept the Membership Report, seconded by David Quirt, and passed).

### Awards Coordinator Report (Annabel Cohen)

The awards brochure needs updating as well as the distribution list for the brochure. The Secretary volunteered to integrate the distribution list into the master CAA database and update as required. The Secretary will forward the list to BoD members for additions/revisions. There has been about $2,500 given to support a prize in the memory of Raymond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Hetu and a preliminary report from the Committee has indicated that a $100 book prize awarded for the best undergraduate project in the area of acoustics might be suitable. The BoD approved this idea and asked Murray Hodgson, the Committee Chair, to draw-up a description of the prize for the October BoD. (David Quirt moved to accept the report, seconded by Murray Hodgson, and passed).

**Editor's Report (Murray Hodgson)**

Canadian Acoustics is now being printed at a new printer with no change in cost for FY98. It was reported that the number of papers received each year for publication is quite variable ('93: 5, '94: 6, '95: 8, '96: 14, '97: 6, and '98: 3). A lack of papers for the March issue delayed publication. There are currently 4 papers in various stages of review. Murray Hodgson will be stepping down as the Editor after the June issue. The President formally thanked Murray for his sterling service over the past eight years. Ramani Ramakrishnan has agreed to become the Editor beginning with the December issue. Meantime, the September issue will be handled by John Bradley and Trevor Nightingale. (Don Jamieson moved to accept Editor's Report, seconded by David Quirt, and passed).

**Past/Future Conferences:**

*Windsor 1997:* No report available. Dalila Giusti volunteered to meet with the Conference Chair to obtain a cheque for the proceeds and brief report.

*London 1998:* Conference Chair: Don Jamieson, Technical Chair: Meg Cheesman. The conference hotel has been chosen, a contract signed, and twelve special sessions are planned. Session organizers have been approached.

*Victoria 1999:* Stan Dosso agreed to organize the 1999 conference in Victoria.

**Duties of Directors**

The President tabled a discussion document in which it was suggested that the BoD members take on additional duties in support of the organization's key thrusts; the Journal and the annual conference. This would be accomplished by assigning specific tasks to each of the eight Directors and to rotating the responsibilities between the Conference and the Journal each year. In general the ideas were well received, however, it was felt that additional clarification for the duties of the Director's as well as the Executive was required. The President agreed to revise the duty descriptions and report to the Board.

**Other Business**

It was agreed that early May was a better time for the BoD meeting. Thanks were given to John Hemingway for providing a meeting room and arranging for lunch service.

*Motion to Adjourn:* Don Jamieson moved that the meeting be adjourned, seconded by Annabel Cohen, and passed. Meeting adjourned at 2:50 p.m.

## JOB SOUGHT

Acoustician from Russia, Ph. D., seeks a research opportunity abroad. 25 years research experience in musical acoustics and sound quality problems including about one year research in Sweden and two years of NATO Science Fellowship in Canada. Please contact Alexander Galembo: galembo@pavlov.psyc.queensu.ca, fax 1-613-5452499 or tel. 1-613-545600 ext. 5754.

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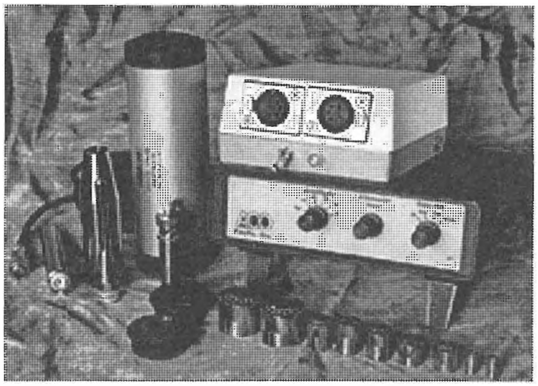

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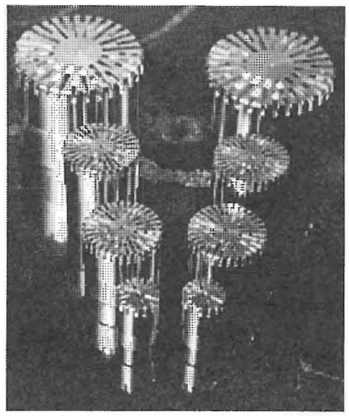

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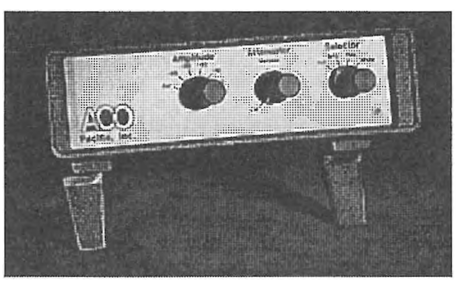

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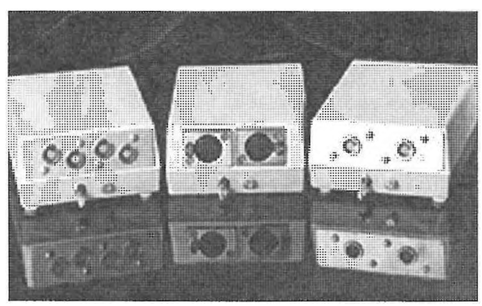

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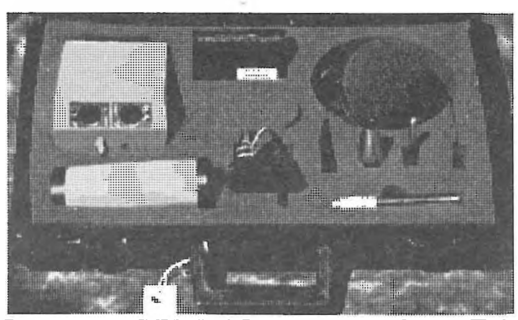

# ACOustics Begins With ACO™

# NEWS / INFORMATIONS

## CONFERENCES

*The following list of conferences was mainly provided by the Acoustical Society of America. If you have any news to share with us, send them by mail or fax to the News Editor (see address on the inside cover), or via electronic mail to desharnais@drea.dnd.ca*

*1998*

4-7 June: 7th Symposium on Cochlear Implants in Children, Iowa City, IA. Contact: Center for Conferences and Institutes, The University of Iowa, 249 Iowa Memorial Union, Iowa City, IA 52242-1317; Tel: 800-551-9029; Fax: 319-335-3533.

8-10 June: EAA/EEAA Symposium "Transport Noise and Vibrations", Tallinn, Estonia. Contact: East-European Acoustical Association, Moskovskoe Shosse 44, 196158 St.-Petersburg, Russia; FAX: +7 812 127 9323; Email: krylspb@sovam.com

9-12 June: 8th International Conference on Hand-Arm Vibration, Umea, Sweden. Contact: National Inst. for Working Life, Physiology and Technology Dept., P.O. Box 7654, 90713 Umea, Sweden; Fax: +46 90 165027; Email: hav98@niwl.se

22-26 June: 135th meeting of the Acoustical Society of America/16th International Congress on Acoustics, Seattle, WA. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; Email: asa@aip.org, WWW: http://asa.aip.org

26 June - 1 July: International Symposium on Musical Acoustics, ISMA 98, Leavenworth, WA. Contact: Maurits Hudig, Catgut Acoustical Society, 112 Essex Ave., Montclair, NJ 07042, Fax: 201-744-9197; Email: catgutas@msn.com, WWW: www.boystown.org/isma98

1-12 July: NATO ASI "Computational Hearing," Il Ciocco Tuscany, Italy. Contact: S. Greenberg, International Computer Science Institute, 1947 Center St., Berkeley, CA 94704, USA; Fax: 510-643-7684; Email: comhear@icsi.berkeley.edu; Web: www.icsi.berkeley.edu/real/comhear98/

7-12 July: Vienna and the Clarinet, Ohio State Univ., Columbus, OH. Contact: Keith Koons, Music Dept., Univ. of Central Florida, P.O. Box 161354, Orlando, FL 32816-1354; Tel.: 407-823-5116; Email: kkons@pegasus.cc.ucf.edu

9-14 August: International Acoustic Emission Conference, Hawaii, HI. Contact: Karyn S. Downs, Lockheed Martin Astronautics, PO Box 179, M.S. DC3005, Denver, CO 80201; Tel: 303-977-1769; Fax: 303-971-7698; Email: karyn.s.downs@lmco.com

7-9 September: Nordic Acoustical Society Meeting 98, Stockholm, Sweden. Contact: Swedish Acoustical Society, c/o Ingemansson AB, Box 47321, 10074 Stockholm, Sweden; Fax:+46 818 2678: Email: nam98@ingemansson.se

13-17 September: American Academy of Otolaryngology--Head and Neck Surgery, San Francisco, CA. Contact: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314. Tel.: 703-836-4444; FAX: 703-683-5100.

14-16 September: Biot Conference on Poromechanics, Louvain-la-Neuve, Belgium. Contact: J.F. Thimus, Unité de Génie civil, Université catholique de Louvain, Place du Levant 1, 1348 Louvain-la-Neuve, Belgium; Fax: +32 10 472179; Email: biotconf@gc.ucl.ac.be; Web: www.gc.ucl.ac.be/gc/geotech/geoma.html

## CONFÉRENCES

*La liste de conférences ci-jointe a été offerte en majeure partie par l'Acoustical Society of America. Si vous avez des nouvelles à nous communiquer, envoyez-les par courrier ou fax (coordonnées incluses à l'envers de la page couverture), ou par courrier électronique à desharnais @drea.dnd.ca*

*1998*

4-7 juin: 7e Symposium sur les implants de cochlée sur les enfants, Iowa City, IA. Info: Center for Conferences and Institutes, University of Iowa, 249 Iowa Memorial Union, Iowa City, IA 52242-1317; Tél: 800-551-9029; Fax: 319-335-3533.

8-10 juin: Symposium EAA/EEAA "Bruit et vibrations des transports", Tallinn, Estonia. Info: East-European Acoustical Association, Moskovskoe Shosse 44, 196158 St. Petersburg, Russia; FAX: +7 812 127 9323; Email: krylspb @sovam.com

9-12 juin: 8e conférence internationale sur les vibrations main-bras, Umea, Suède. Info: National Inst. for Working Life, Physiology and Technology Dept., P.O. Box 7654, 90713 Umea, Sweden; Fax: +46 90 165027; Email: hav98@niwl.se

22-26 juin: 135e rencontre de l'Acoustical Society of America/16e congrès international d'acoustique, Seattle, WA. Info: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tél.: 516-576-2360; FAX: 516-576-2377; Email: asa @aip.org; WWW: http://asa.aip.org

26 juin - 1 juillet: Symposium international sur l'acoustique de la musique, ISMA 98, Leavenworth, WA. Info: Maurits Hudig, Catgut Acoustical Society, 112 Essex Ave., Montclair, NJ 07042, Fax: 201-744-9197; Email: catgutas@msn.com, WWW: www.boystown.org/isma98

1-12 juillet: "Audition informatisée" de l'institut ASI de l'OTAN, Il Ciocco Tuscane, Italie. Info: S. Greenberg, International Computer Science Institute, 1947 Center St., Berkeley, CA 94704, USA; Fax: 510-643-7684; Email: comhear@icsi.berkeley.edu; Web: www.icsi.berkeley.edu/real/comhear98/

7-12 juillet: Vienne et la clarinette, Ohio State Univ., Columbus, OH. Info: Keith Koons, Music Dept., Univ. of Central Florida, P.O. Box 161354, Orlando, FL 32816-1354; Tél.: 407-823-5116; Email: kkons@pegasus.cc.ucf.edu

9-14 août: Conférence internationale sur les émissions acoustiques, Hawaii, HI. Info: Karyn S. Downs, Lockheed Martin Astronautics, PO Box 179, M.S. DC3005, Denver, CO 80201; Tél: 303-977-1769; Fax: 303-971-7698; Email: karyn.s.downs@lmco.com

7-9 septembre: Rencontre 98 de la Société nordique d'acoustique, Stockholm, Suède. Info: Swedish Acoustical Society, c/o Ingemansson AB, Box 47321, 10074 Stockholm, Sweden; Fax: +46 818 2678; Email: nam98@ingemansson.se

13-17 septembre: Académie américaine d'otolaryngologie - Chirurgie de la tête et du cou, San Francisco, CA. Info: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314. Tél.: 703-836-4444; FAX: 703-683-5100.

14-16 septembre: Conférence Biot sur la poro-mécanique, Louvain-la-Neuve, Belgique. Info: J.F. Thimus, Unité de Génie civil, Université catholique de Louvain, Place du Levant 1, 1348 Louvain-la-Neuve, Belgique; Fax: +32 10 472179; Email: biotconf@gc.ucl.ac.be; Web: www.gc.ucl.ac.be/gc/geotech/geoma.html

14-18 September: 35th International Conference on Ultrasonics and Acoustic Emission, Chateau of Treste, Czech Republic. Contact: H. Kotschova, Geophysical Institute, AS Bocni II/401, 14131 Prague 4, Czech Republic; Fax: +42 2 761 549; Email: hko@ig.cas.cs; Web: www.ig.cas.cz

21-25 September: 3rd International CARIS conference - CARIS and the year of the Ocean, De Ruwenberg, The Netherlands. Contact: Universal Systems Ltd., c/o CARIS and the Year of the Ocean, 264 Rookwood Avenue, Fredericton, New Brunswick, Canada, E3B 2M2; Tel.: 506-458-8533; FAX: 506-459-3849.

12-16 October: 136th meeting of the Acoustical Society of America, Norfolk, VA. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; Email: asa@aip.org; WWW: http://asa.aip.org

16-18 November: Inter-Noise 98, Christchurch, New Zealand. Contact: New Zealand Acoustical Society, P.O. Box 1181, Auckland, New Zealand.

20 November: Recreational Noise, Queenstown, New Zealand. Contact: P. Dickenson, NZ Ministry Health, PO Box 5013, Wellington, New Zealand; Fax: +64 4 496 2340; Email: philip.dickenson@mohwn.synet.net.nz

22-26 November: Noise Effects '98 - 7th International Congress on Noise as a Public Health Problem, Sydney, Australia. Contact: GPO Box 128, Sydney NSW 2001, Australia, Tel: +61 2 9262 2277; Fax: +61 2 9262 3135; Email: noise98@tourhosts.com.au; WWW: www.acay.com .au/~dstuckey/noise-effects98

30 November - 4 December: 5th International Conference on Spoken Language Processing, Sydney, Australia. Contact: ICSLP Secretariat, Tour Hosts, GPO Box 128, Sydney, NSW 2001, Australia; Fax: +61 2 9262 3135; Email: tourhosts @tourhosts.com.au; WWW: http://cslab.anu.edu.au/icslp98

*1999*

15-19 March: 137th Meeting of Acoustical Society of America/European Acoustics Association Forum Acusticum, Berlin, Germany. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797; Tel: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; WWW: asa.aip.org

17-20 May: Society of Automotive Engineers (SAE) and Noise and Vibration Conference & Exposition meeting, Traverse City, MI. Contact: M.J. Asensio, SAE/Troy, 3001 W Big Beaver Rd, Troy, MI, USA. Tel: 248-649-4920, ext. 3106.

27-30 June: ASME Mechanics and Materials Conference, Blacksburg, VA. Contact: Mrs. Norma Guynn, Dept. of Engineering Science and Mechanics, Virginia Tech, Blacksburg, VA 24061-0219; Fax: 540-231-4574; Email: nguynn@vt.edu; WWW: http://www.esm.vt.edu/mmconf/

28-30 June: 1st International Congress of the East European Acoustical Association, St. Petersburg, Russia. Contact: EEAA, Moskovskoe Shosse 44, St. Petersburg 196158, Russia; Fax: +7 812 127 9323; Email: krylspb @sovam.com

4-9 July: 10th British Academic Conference in Otolaryngology, London, UK. Contact: BOA-HNS, The Royal College of Surgeons, 35-43 Lincoln's Inn Field, London WC2A 3PN, UK; Fax: +44 171 404 4200.

1-4 September: 15th International Symposium on Nonlinear Acoustics (ISNA-15), Gottingen, Germany. Contact: W. Lauterborn, Drittes Physikalisches Institut, Universitat Gottingen, Burgerstr. 42-44, 37073 Gottingen, Germany; Fax: +49 551 39 7720; Email: lb@physik3.gwdg.de

14-18 septembre: 35e Conférence internationale sur les ultrasons et les émissions acoustiques, Château de Treste, République Tchèque. Info: H. Kotschova, Geophysical Institute, AS Bocni II/401, 14131 Prague 4, Czech Republic; Fax: +42 2 761 549; Email: hko@ig.cas.cs; Web: www.ig.cas.cz

21-25 septembre: 3e conférence internationale CARIS - CARIS et l'Année de l'océan, De Ruwenberg, Pays-Bas. Info: Universal Systems Ltd., a/s CARIS et l'Année de l'océan, 264 Rookwood Avenue, Frédericton, Nouveau Brunswick, Canada, E3B 2M2; Tél.: 506-458-8533; Fax: 506-459-3849.

12-16 octobre: 136e rencontre de l'Acoustical Society of America, Norfolk, VA. Info: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tél.: 516-576-2360; FAX: 516-576-2377; Email: asa@aip.org; WWW: http:// asa.aip.org

16-18 novembre: Inter-Noise 98, Christchurch, Nouvelle-Zélande. Info: New Zealand Acoustical Society, P.O. Box 1181, Auckland, New Zealand.

20 novembre: Bruit récréatif, Queenstown, Nouvelle-Zélande. Info: P. Dickenson, NZ Ministry Health, PO Box 5013, Wellington, New Zealand; Fax: +64 4 496 2340; Email: philip.dickenson@mohwn.synet.net.nz

22-26 novembre: Noise Effects '98 - 7e congrès international sur le bruit comme problème pour la santé publique, Sydney, Australie. Info: GPO Box 128, Sydney NSW 2001, Australia, Tél: +61 2 9262 2277; Fax: +61 2 9262 3135; Email: noise98@tourhosts.com.au; WWW: www.acay .com.au/~dstuckey/noise-effects98

30 novembre- 4 décembre: 5e conférence internationale sur le traitement de la langue parlée, Sydney, Australie. Info: ICSLP Secretariat, Tour Hosts, GPO Box 128, Sydney, NSW 2001, Australia; Fax: +61 2 9262 3135; Email: tourhosts @tourhosts.com.au; WWW: http://cslab.anu.edu.au/icslp98

*1999*

15-19 mars: 137e Rencontre de l'Acoustical Society of America et de l'Association d'acoustique européenne Forum Acusticum, Berlin, Allemagne. Info: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797; Tél: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; WWW: asa.aip.org

17-20 mai: Conférence et exposition de la Société des Ingénieurs d'autos (SAE) et conférence Bruit et Vibrations, Traverse City, MI. Info: M.J. Asensio, SAE/Troy, 3001 W Big Beaver Rd, Troy, MI, USA. Tél: 248-649-4920, poste 3106.

27-30 juin: Conférence ASME sur la mécanique et les matériaux, Blacksburg, VA. Info: Mrs. Norma Guynn, Dept. of Engineering Science and Mechanics, Virginia Tech, Blacksburg, VA 24061-0219; Fax: 540-231-4574; Email: nguynn@vt.edu; WWW: http://www.esm.vt.edu/mmconf/

28-30 juin: 1er Congrès international de l'Association d'acoustique de l'Europe de l'Est, St. Petersburg, Russie. Info: EEAA, Moskovskoe Shosse 44, St. Petersburg 196158, Russia; Fax: +7 812 127 9323; Email: krylspb @sovam.com

4-9 juillet: 10e Conférence académique britannique sur l'otolaryngologie, Londres, Royaume-Uni. Info: BOA-HNS, The Royal College of Surgeons, 35-43 Lincoln's Inn Field, London WC2A 3PN, UK; Fax: +44 171 404 4200.

1-4 septembre: 15e Symposium international sur l'acoustique non-linéaire (ISNA-15), Gottingen, Allemagne. Info: W. Lauterborn, Drittes Physikalisches Institut, Universitat Gottingen, Burgerstr. 42-44, 37073 Gottingen, Germany; Fax: +49 551 39 7720; Email: lb@physik3 .gwdg.de

# The Canadian Acoustical Association
# l'Association Canadienne d'Acoustique

## PRIZE ANNOUNCEMENT

A number of prizes, whose general objectives are described below, are offered by the Canadian Acoustical Association. As to the first four prizes, applicants must submit an application form and supporting documentation to the prize coordinator before the end of February of the year the award is to be made. Applications are reviewed by subcommittees named by the President and Board of Directors of the Association. Decisions are final and cannot be appealed. The Association reserves the right not to make the awards in any given year. Applicants must be members of the Canadian Acoustical Association. Preference will be given to citizens and permanent residents of Canada. Potential applicants can obtain full details, eligibility conditions and application forms from the appropriate prize coordinator.

### Edgar and Millicent Shaw Postdoctoral Prize in Acoustics

This prize is made to a highly qualified candidate holding a Ph.D. degree or the equivalent, who has completed all formal academic and research training and who wishes to acquire up to two years supervised research training in an established setting. The proposed research must be related to some area of acoustics, psychoacoustics, speech communication or noise. The research must be carried out in a setting other than the one in which the Ph.D. degree was earned. The prize is for $3000 for full-time research for twelve months, and may be renewed for a second year. Coordinator: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Past recipients are:

| | | | | | |
|---|---|---|---|---|---|
| *1990* | *Li Cheng* | *Université de Sherbrooke* | *1995* | *Jing-Fang Li* | *University of British Columbia* |
| *1993* | *Roland Woodcock* | *University of British Columbia* | *1996* | *Vijay Parsa* | *University of Western Ontario* |
| *1994* | *John Osler* | *Defense Research Estab. Atlantic* | | | |

### Alexander Graham Bell Graduate Student Prize In Speech Communication and Behavioural Acoustics

The prize is made to a graduate student enrolled at a Canadian academic institution and conducting research in the field of speech communication or behavioural acoustics. It consists of an $800 cash prize to be awarded annually. Coordinator: Don Jamieson, Department of Communicative Disorders, University of Western Ontario, London, ON N6G 1H1. Past recipients are:

| | | | | | |
|---|---|---|---|---|---|
| *1990* | *Bradley Frankland* | *Dalhousie University* | *1994* | *Michael Lantz* | *Queen's University* |
| *1991* | *Steven D. Turnbull* | *University of New Brunswick* | *1995* | *Kristina Greenwood* | *University of Western Ontario* |
| | *Fangxin Chen* | *University of Alberta* | *1996* | *Mark Pell* | *McGill University* |
| | *Leonard E. Cornelisse* | *University of Western Ontario* | *1997* | *Monica Rohlfs* | *University of Alberta* |
| *1993* | *Aloknath De* | *McGill University* | | | |

### Fessenden Student Prize in Underwater Acoustics

The prize is made to a graduate student enrolled at a Canadian university and conducting research in underwater acoustics or in a branch of science closely connected to underwater acoustics. It consists of $500 cash prize to be awarded annually. Coordinator: David Chapman, DREA, PO Box 1012, Dartmouth, NS B2Y 3Z7.

| | | | | | |
|---|---|---|---|---|---|
| *1992* | *Danlela Dilorio* | *University of Victoria* | *1994* | *Craig L. McNeil* | *University of Victoria* |
| *1993* | *Douglas J. Wilson* | *Memorial University* | *1996* | *Dean Addison* | *University of Victoria* |

### Eckel Student Prize in Noise Control

The prize is made to a graduate student enrolled at a Canadian academic institution pursuing studies in any discipline of acoustics and conducting research related to the advancement of the practice of noise control. It consists of a $500 cash prize to be awarded annually. The prize was inaugurated in 1991. Coordinator: Murray Hodgson, Occupational Hygiene Programme, University of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3.

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|---|---|---|---|---|---|
| *1994* | *Todd Busch* | *University of British Columbia* | *1996* | *Nelson Heerema* | *University of British Columbia* |
| *1995* | *Raymond Panneton* | *Université de Sherbrooke* | *1997* | *Andrew Wareing* | *University of British Columbia* |

### Directors' Awards

Three awards are made annually to the authors of the best papers published in *Canadian Acoustics*. All papers reporting new results as well as review and tutorial papers are eligible; technical notes are not. The first award, for $500, is made to a graduate student author. The second and third awards, each for $250, are made to professional authors under 30 years of age and 30 years of age or older, respectively. Coordinator: David Quirt, Acoustics Section, Institute for Research in Construction, NRCC, Ottawa, ON K1A 0R6.

### Student Presentation Awards

Three awards of $500 each are made annually to the undergraduate or graduate students making the best presentations during the technical sessions of Acoustics Week in Canada. Application must be made at the time of submission of the abstract. Coordinator: Alberto Behar, 45 Meadowcliffe Drive, Scarborough, ON M1M 2X8.

# The Canadian Acoustical Association
# l'Association Canadienne d'Acoustique

## ANNONCE DE PRIX

Plusieurs prix, dont les objectifs généraux sont décrits ci-dessous, sont décernés par l'Association Canadienne d'Acoustique. Pour les quatre premiers prix, les candidats doivent soumettre un formulaire de demande ainsi que la documentation associée au coordonnateur de prix avant le dernier jour de février de l'année durant laquelle le prix sera décerné. Toutes les demandes seront analysées par des sous-comités nommés par le président et la chambre des directeurs de l'Association. Les décisions seront finales et sans appel. L'Association se réserve le droit de ne pas décerner les prix une année donnée. Les candidats doivent être membres de l'Association. La préférence sera donnée aux citoyens et aux résidents permanents du Canada. Les candidats potentiels peuvent se procurer de plus amples détails sur les prix, leurs conditions d'éligibilité, ainsi que des formulaires de demande auprès du coordonnateur de prix.

### Prix Post-Doctoral Edgar et Millicent Shaw en Acoustique

Ce prix est attribué à un(e) candidat(e) hautement qualifié(e) et détenteur(rice) d'un doctorat ou l'équivalent, qui a complèté(e) ses études et sa formation de chercheur, et qui désire acquérir jusqu'à deux années de formation supervisée de recherche dans un établissement reconnu. Le thème de recherche proposée doit être relié à un domaine de l'acoustique, de la psycho-acoustique, de la communication verbale ou du bruit. La recherche doit être menée dans un autre milieu que celui où le candidat a obtenu son doctorat. Le prix est de $3000 pour une recherche plein temps de 12 mois avec possibilité de renouvellement pour une deuxième année. Coordonnatrice: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Les récipiendaires antérieur(e)s sont:

| | | | | | |
|---|---|---|---|---|---|
| *1990* | *Li Cheng* | *Université de Sherbrooke* | *1995* | *Jing-Fang Li* | *University of British Columbia* |
| *1993* | *Roland Woodcock* | *University of British Columbia* | *1996* | *Vijay Parsa* | *University of Western Ontario* |
| *1994* | *John Osler* | *Defense Research Estab. Atlantic* | | | |

### Prix Étudiant Alexander Graham Bell en Communication Verbale et Acoustique Comportementale

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne et menant un projet de recherche en communication verbale ou acoustique comportementale. Il consiste en un montant en argent de $800 qui sera décerné annuellement. Coordonnateur: Don Jamieson, Department of Communicative Disorders, University of Western Ontario, London, ON N6G 1H1. Les récipiendaires antérieur(e)s sont:

| | | | | | |
|---|---|---|---|---|---|
| *1990* | *Bradley Frankland* | *Dalhousie University* | *1994* | *Michael Lantz* | *Queen's University* |
| *1991* | *Steven D. Turnbull* | *University of New Brunswick* | *1995* | *Kristina Greenwood* | *University of Western Ontario* |
| | *Fangxin Chen* | *University of Alberta* | *1996* | *Mark Pell* | *McGill University* |
| | *Leonard E. Cornelisse* | *University of Western Ontario* | *1997* | *Monica Rohlfs* | *University of Alberta* |
| *1993* | *Aloknath De* | *McGill University* | | | |

### Prix Étudiant Fessenden en Acoustique Sous-marine

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne et menant un projet de recherche en acoustique sous-marine ou dans une discipline scientifique reliée à l'acoustique sous-marine. Il consiste en un montant en argent de $500 qui sera décerné annuellement. Coordonnateur: David Chapman, DREA, PO Box 1012, Dartmouth, NS B2Y 3Z7.

| | | | | | |
|---|---|---|---|---|---|
| *1992* | *Daniela Dilorio* | *University of Victoria* | *1994* | *Craig L. McNeil* | *University of Victoria* |
| *1993* | *Douglas J. Wilson* | *Memorial University* | *1996* | *Dean Addison* | *University of Victoria* |

### Prix Étudiant Eckel en Controle du Bruit

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne dans n'importe quelle discipline de l'acoustique et menant un projet de recherche relié à l'avancement de la pratique en contrôle du bruit. Il consiste en un montant en argent de $500 qui sera décerné annuellement. Ce prix a été inauguré en 1991. Coordonnateur: Murray Hodgson, Occupational Hygiene Programme, University of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3.

| | | | | | |
|---|---|---|---|---|---|
| *1994* | *Todd Busch* | *University of British Columbia* | *1996* | *Nelson Heerema* | *University of British Columbia* |
| *1995* | *Raymond Panneton* | *Université de Sherbrooke* | *1997* | *Andrew Wareing* | *University of British Columbia* |

### Prix des Directeurs

Trois prix sont décernés, à tous les ans, aux auteurs des trois meilleurs articles publiés dans l'*Acoustique Canadienne*. Tout manuscrit rapportant des résultats originaux ou faisant le point sur l'état des connaissances dans un domaine particulier sont éligibles; les notes techniques ne le sont pas. Le premier prix, de $500, est décerné à un(e) étudiant(e) gradué(e). Le deuxième et le troisième prix, de $250 chacun, sont décernés à des auteurs professionnels âgés de moins de 30 ans et de 30 ans et plus, respectivement. Coordonnateur: David Quirt, Section d'acoustique, Institut de Recherche en Construction, NRCC, Ottawa, ON K1A 0R6.

### Prix de Presentation Étudiant

Trois prix, de $500 chacun, sont décernés annuellement aux étudiant(e)s sous-gradué(e)s ou gradué(e)s présentant les meilleures communications lors de la Semaine de l'Acoustique Canadienne. La demande doit se faire lors de la soumission du résumé. Coordonnateur: Alberto Behar, 45 Meadowcliffe Drive, Scarborough, ON M1M 2X8.

## 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.

**Photographs**: Submit original glossy, black and white photograph.

**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.

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

**Références**: Les citer dans le texte et en faire la liste à la fin du document, en format uniforme, 9 pt à simple (12 pt) interligne.

**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 | C A A | l'Association Canadienne d'Acoustique |
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## SUBSCRIPTION INVOICE

Subscription for the current calendar year is due January 31. New subscriptions received before July 1 will be applied to the current year and include that year's back issues of *Canadian Acoustics*, if available. Subscriptions received from July 1 will be applied to the next year.

## FACTURE D'ABONNEMENT

L'abonnement pour la présente année est dû le 31 janvier. Les nouveaux abonnements reçus avant le 1 juillet s'appliquent à l'année courante et incluent les anciens numéros (non-épuisés) de *l'Acoustique Canadienne* de cette année. Les abonnements reçus après le 1 juillet s'appliquent à l'année suivante.

Check ONE Item Only: / Cocher la case appropriée :

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| CAA Student membership | ___ | $ 10 | ___ | Membre étudiant(e) |
| Institutional Subscription | ___ | $ 50 | ___ | Membre institutionnel |
| Sustaining Subscription | ___ | $ 150 | ___ | Abonnement de soutien |
| Total Remitted | | $_______ | | Versement total |

INFORMATION FOR MEMBERSHIP DIRECTORY / RENSEIGNEMENT POUR L'ANNUAIRE DES MEMBRES

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| Hearing Sciences | 7. | _____ | 7. | Audition |
| Speech Sciences | 8. | _____ | 8. | Parole |
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| Signal Processing / Numerical Methods | 10. | _____ | 10. | Traitement des signaux / Méthodes numériques |
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