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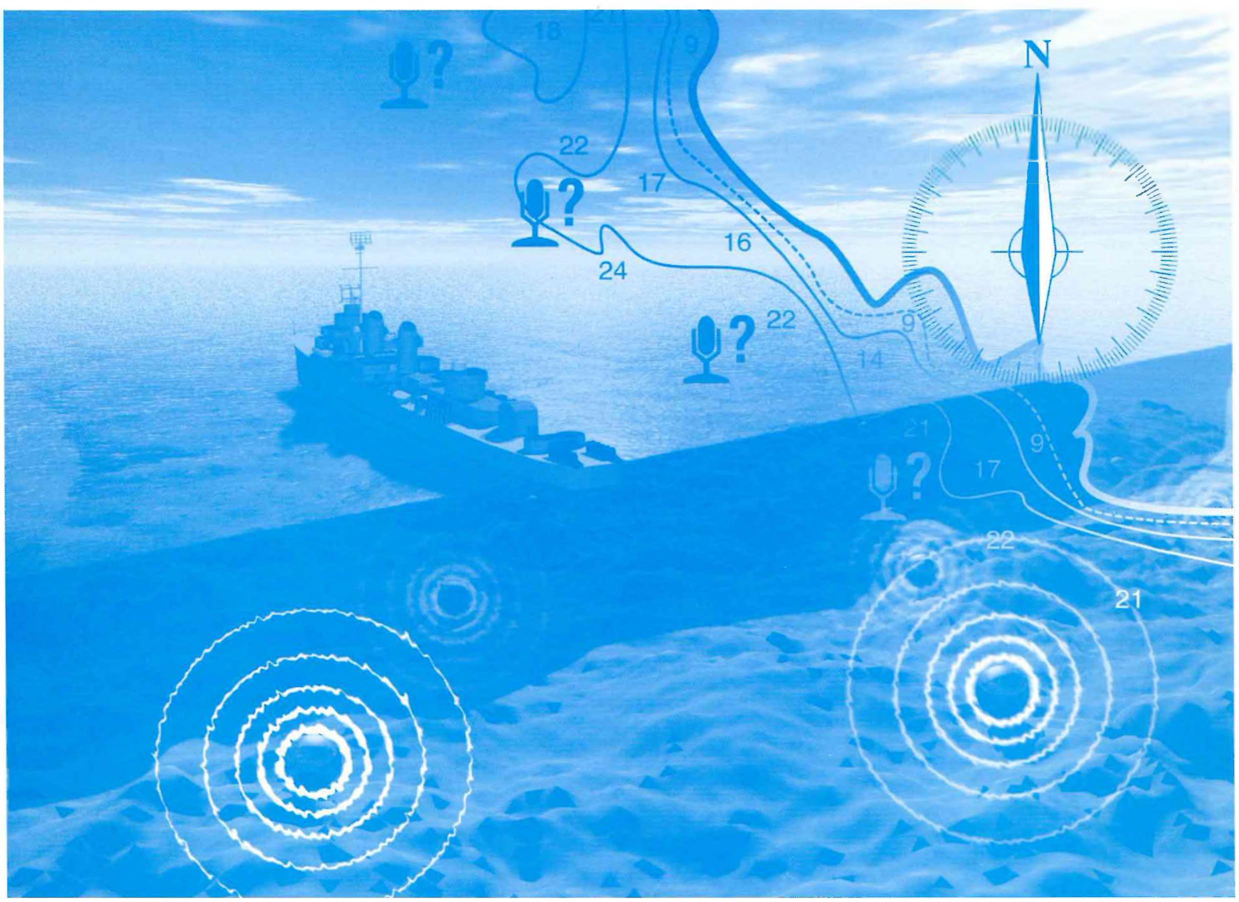

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EDITORIAL / ÉDITORIAL

Ah - The beauty of QuarkExpress or its conflict with Microsoft! Yours truly was very proud in learning the industry standard publishing software and producing an archival record of the journals by laying out each issue in QuarkExpress. I was even able to convince the CAA Executive to buy in to my line of thinking. Alas, all is not well in the land of e-mails, file links, font lists and what have you. The number of screw-ups that graced the lead paper in the March 2000 issue of Canadian Acoustics was too numerous that we decided to hide behind Monk's robes and reprint the article, "Binaural technology for application to active noise reduction communication headsets: Design considerations," by C. Giguere et al. We have conveyed our sincere regrets to the authors for creating this mess! Hope we have learnt from this experience. We have also started to revamp and upgrade the "Instructions for Authors" and we hope to publish the revised guidelines soon.

We had solicited responses to the article on the use of decibels, published in Volume 106, pp. 3048, 1999 of JASA. We include the responses below. Hope this creates further discussion on this elusive quantity.

Ah - La beauté de QuarkExpress ou de son conflit avec Microsoft! J'étais très fier d'apprendre à utiliser le logiciel d'édition QuarkExpress qui est le standard dans l'industrie, et de pouvoir reproduire des archives des journaux en incorporant chaque issue dans le logiciel. Je pouvais même convaincre le directeur de CAA d'être en accord avec moi. Hélas, tout n'est pas aussi bien dans cette terre du courrier électronique, liens de fichier, liste de fontes et bien d'autres. Les erreurs qui ont entaché l'article principal de l'édition de mars 2000 du journal d'acoustique canadienne étaient tellement importantes que nous avons décidé de nous cacher derrière de longues robes de moines et ré-édité l'article, "Binaural technology for application to active noise reduction communication headsets: Design considerations," par C. Giguere et al. Nous avons envoyé nos excuses aux auteurs pour ce désordre! Espérant que cette expérience nous a appris quelque chose. Nous avons également commencé à améliorer la rubrique "instructions pour les auteurs" et nous espérons éditer les nouvelles directives révisées bientôt.

Nous avions sollicité des réponses à l'article sur l'utilisation des décibels, éditée dans le volume 106, pp 3048, 1999 de JASA. Nous icluons les réponses ci-dessous. Nous espérons que ceci crée davantage de discussion sur cette quantité insaisissable

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MAIL

Decibels, SI Units, and Standards

by David M.F. Chapman
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Recently, the *Journal of the Acoustical Society of America* published two opinion articles in its Forum section advocating the abolition of the decibel measure in acoustics[1]. The following article was submitted to accompany the first two articles, but has not yet appeared in JASA (although this is promised!). In response to the invitation of the Editor of *Canadian Acoustics*, the *verbatim* article is offered as a contribution to a discussion on the topic of decibels in these pages.

* * * * * * * * * * * * *

Provided they are used correctly, decibels provide a convenient centigrade scale of sound levels in air that nicely matches human auditory experience: 0 dB re 20 μPa is about the smallest sound a human can detect and 100 dB re 20 μPa is about as loud a sound that we will tolerate for a short duration. The rationale for using decibels in other fields is not so well established, but decibel use has spread to applications in underwater acoustics, radio wave communications, and electronics in general. For acoustical measurements in liquids, the standard reference pressure has been chosen to be twenty times lower, at 1 μPa. Using different reference pressures in gases and in liquids is confusing, especially when considering cross-disciplinary acoustic issues that involve dissimilar acoustic media (mammal hearing in air and in water, for example). However, this dichotomy is entrenched in an ANSI standard and it is unlikely to change soon [2]. Many guidelines, regulations, and laws are written in the language of decibels, and it would take time to expunge the decibel from these, even if all the acoustical experts in the world woke up tomorrow sharing the conviction that the decibel had outlived its purpose.

The general public, most journalists, and even some scientists have difficulty with decibels. Non-experts use the decibel as if it were a physical unit itself, rather than a logarithmic measure of the ratio of like physical quantities. Perhaps they imagine that sound comes in decibels in the same way that cheese is sold by the pound. "Unadorned" decibels make sense only when reporting truly dimensionless ratios such as the gain of an amplifier or the attenuation of a filter. When decibels are used to represent absolute quantities having physical dimensions, such as sound pressure or sound intensity, it is imperative that the reference quantities be clearly stated, otherwise decibels are meaningless. This dictum is not always observed, despite the fact that the ANSI standard demands it. No wonder we see inappropriate comparisons between transmitted source levels, received sound pressure levels, and noise spectral density levels: often the reference pressures, distances, and bandwidths are missing! Even respected scientific journals and newsmagazines are guilty of mixing "apple" decibels with "orange" decibels [3].

In recent years, the argument over safe levels of underwater sound for marine animals has been a fertile breeding ground for decibel confusion. Correcting misconceptions due to ignorance of decibels distracts us and drains our resources. Abandoning the decibel might allow us to focus on the real issues; however, this alone will not terminate the debate. Underwater acousticians may report radiated sound powers in watts and received sound pressures in pascals if they like, but we should take care not to extrapolate the auditory experience of humans in air to animals that have adapted to living in entirely different habitats. Claiming that an underwater sound projector is "quieter" for a whale than a jet aircraft is for a human on the basis of lower radiated power is a specious argument; so is the claim that the projector is "louder" for the whale than the jet is for a human on the basis of higher received sound pressure. Loudness is subjective, particularly so when your sample space includes different species inhabiting different environments.

The attractiveness of a logarithmic system of units for reporting auditory perception cannot be overlooked. Anyone who listens to the ASA's *Auditory Demonstrations on Compact Disk* will soon be convinced of this. Also, it is evident that fluctuations of underwater noise, sonar pings, and reverberation are more sensibly displayed on a logarithmic scale. In cases where logarithmic units seem "natural", reporting quantities in linear units might over-emphasize differences that are actually of modest proportion in perceptual terms. The generation, transmission, and transduction of sound can be presented handily in SI units without recourse to decibels. The audition and perception of sound by humans—and possibly other mammals—could be presented in SI units, but our experience strongly suggests a logarithmic system of units that matches our senses. Between these poles lies a grey area in which either or both systems could be used if good judgement is applied.

For scientific purposes, the arithmetic convenience of decibels has largely been superseded—but not entirely. Computer-assisted calculations are now the norm, and the choice between linear and logarithmic plotting of data is only a few mouse clicks away. There is nothing to stop the scientific community from voluntarily abandoning decibels in favour of SI units; would any AIP editor or reviewer turn down a paper on that score? Changing the standards, guidelines, regulations, etc. will take more time. In making any changes, we might introduce additional confusion while everyone adjusts to the "new" modes of expression; there is no doubt that non-experts will find new ways to confuse themselves! Whether or not the decibel falls from favour, we—as scientists, authors, reviewers, and editors—are responsible for ensuring that decibel quantities are reported with appropriate reference values, according to the defined and accepted standards.

To end on a lighter note, I would like to pass on the following working definition of the decibel, collected at a recent acoustics meeting: "When you complain that the television is too loud, your children reduce the volume by precisely one decibel."

Many thanks to DREA colleagues Dale Ellis, Harold Merklinger, and Paul Hines for their contributions to this article.

[David M.F. Chapman is a Past President of the Canadian Acoustical Association, a Fellow of the Acoustical Society of America, and the Head of the Naval Sonar Section at DREA.]

References

1. Clarence S. Clay, "Underwater sound transmission and SI Units", J. Acoust. Soc. Am. **106**, 3047 (1999) and Robert Hickling, "Noise Control and SI units", J. Acoust. Soc. Am. **106**, 3048 (1999).

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

3. David M.F. Chapman and Dale D. Ellis, "The Elusive Decibel: Thoughts on Sonars and Marine Mammals", *Canadian Acoustics* **26**(2), 29–31 (1996). Also available on the Internet at www.drea.dnd.ca, in PDF format. (Follow the "documentation" link.)

After reviewing the letter about SI units and decibels, I was a little upset to think that Engineers do not have the capacity to understand decibels and logarithms.

I do not agree that we will gain a better appreciation of acoustics by the use of SI units. There is simply too much information in the literature that would have to be translated. The benefits are too limited. Any serious practitioner of acoustics has no trouble handling decibels. Changing the units will not make it any easier for the non-practitioner to understand how acoustical concepts are to be handled.

Cameron Sherry
Montreal, Quebec

With regard to Hickling, I find none of his arguments compelling. In our practise I believe that the use of linear" engineering units will be perceived as even more of an affectation than the venerable decibel. Will the layman be more at home with sound pressure expressed in exponential form or with the requisite number of zeros? In engineering acoustics the reference quantity is understood, and only causes ambiguity when working with poorly defined underwater acoustics work. The latter carries the historical baggage of several different systems of units — a condition which will certainly not be changed by adoption of another scheme, no matter how well meaning.

The question of standards alluded to by Hickling is a thorny one. Can we expect all the standards bodies to suddenly change over? Until they do, the situation would be worse than ever, with competing systems both in use and the real physics submerged in a sea of conversion problems. As we have discussed often enough, the "vector intensity" measurement is a straw man when it comes to field work. Even the practitioners who champion in seem to admit, whether tacitly or overtly, that it is not a tool for everyday noise problems, due to the extreme difficulty in overcoming the practical limitations of real world conditions. I cannot forsee the day when we will forsake sound level meters for intensity meters in our day to day work. I have yet to see any papers on auditorium acoustics which have attempted to measure the vector intensity of reflection patterns.

As for the gratuitous crack about decibel notation diminishing or subverting the "real engineering discipline" of acoustics, I cannot see how the units used, or the representation of the physical quantities can materially affect the quality of the engineering. There are examples of notation being developed at least partially to obscure the truth or elevate the status of the originator — for example, Newton's invention of differential calculus made his derivation of gravity virtually incomprehensibe to his contemporaries. The truth is out there — it only takes a perceptive eye to discern it in whatever form it appears. I cannot believe that the change being championed by Hickling can do anything to simplify the work we do.

Mike Noble
BKL Consultants
North Vancouver, BC

I would like to preface my comments by stating that my initial training in mechanical engineering, and in acoustics and noise control, was undertaken in the "era of slide rules and log tables" therefore my opinion may be somewhat biased. I will start by considering statements made by Robert Hickling regarding "Decibels have the following problems." He states "They create confusion, because the same word, decibel, is used for different acoustical quantities, such as sound pressure, intensity, and power". The decibel is a ratio of quantities and is used, not only in the fields of acoustics and noise control, but also in the field of electrical engineering. In any engineering measurement, to avoid confusion, one must specify the type of quantity measured and the units used in the measurement. This is true for a linear, a logarithmic or a decibel representation of the measurement. He also states that "The logarithmic form of decibels obscures reasoning and is a cover for inaccuracy." One of the main reasons for using a decibel scale is to consider and compare quantities which extend over a number of orders of magnitude, where small differences are usually not that important. This is very often the case in acoustics. He proposes to use plots in logarithmic form, which I agree will work just as well for visual graphing of this type of information. Logarithmic plots may also give the layman a better indication of the large dynamic range in a given acoustical measurement. However, a logarithmic plot will provide the same "cover for inaccuracy" that he suggests is a problem with the decibel. When comparing numerical values, one can calculate as many decimals as one wishes and make decibel quantities just as accurate as linear quantities.

He states that "Decibels are more related to the era of slide rules and log tables than to modern digital processing". I don't think this is true and I don't see how this can be a "problem" since, with modern digital programming, calculation of decibel quantities is now much easier than it was using log tables or slide rules. His statement "Mathematical expressions in decibel form are cumbersome and difficult to interpret" may be true in some cases but I have found more often the opposite is true. By using decibel form, one is replacing powers by multiplication, and replacing multiplication and division by addition and subtraction, leading to simpler expressions and functional forms. He goes on to suggest that "A major problem with decibels is that they contain the implicit assumption that sound intensity is represented by the farfield approximation." I don't see how this relates to the decibel. One can consider both farfield approximations and near field sound intensity using either decibel or linear units.

I suggest that there may be times when it is better to use linear units or logarithmic units but there are probably more instances where I would prefer to use decibel quantities in acoustics and noise control. When in the field measuring sound levels, I would rather record "sound pressure levels of 35, 57, 67, 65, and 90 dBA re 20μPa" and would probably make fewer mistakes than when recording respectively "sound intensities measured with an A-weighting filter of 3.1E-09, 4.9E-07, 4.9E-06, 3.10E-06, and 0.00098 W/m^2." A client may find the explanation that, "a factor of ten change in sound intensity is equivalent to roughly a doubling in loudness of the sound he hears", even more confusing than the statement, "an increase in sound pressure level of 10 dB is roughly equivalent to a doubling of loudness". If he were told that the noise, which is bothering him, could be reduced from 67 dBA to 65 dBA, he would likely conclude, intuitively and correctly, that this change in sound level is small and would not significantly change how well he sleeps at night. On the other hand, if he were told that the equivalent sound intensity would be reduced from 4.9 μW to 3.10 μW, a reduction of 37%, he would probably conclude, incorrectly, that this was a significant change in the sound level. Hickling states that "Decibels are a useless affectation" but I believe there is still a definite need for the decibel in acoustics. On the other hand, in specific situations where it may be more appropriate to use linear quantities or logarithmic scales, I would welcome their inclusion in acoustic and noise control standards.

David C. Stredulinsky, P.Eng, Ph.D
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When I read Mr. Hickling's article on Noise Control and SI Units, I immediately wondered if this person gets out into the real world very much. Far from confusing the public, referring to a level of 70 dBA versus 0.063 pascals is an inherently easier concept. As far as being a cover for inaccuracy, the decibel scale, perhaps taken to a limit of 0.1 dB is more than adequate when dealing with a subject affecting human hearing that can seldom differentiate sound with less than a 2 to 3 dB difference in level. In fact, when we are presented with data expressed to 0.01 dB precision, it is our practice to discount the data as being put forward by someone unclear of the concept.

Regarding Mr.Hickling's claims for "practical and accurate methods of measuring sound intensity in its vector form", I would like to know what planet he comes from. Clearly sound intensity measurement is demonstrably not practical when it comes to measuring field sound transmission loss. It is also next to impossible, and the results are clearly not accurate, when intensity measurements are attempted in a busy foundry or pulp mill.

For two centuries, sound was measured in linear units. However enlightened engineers at AT&T recognized the impractical nature of expressing ranges on numbers from 0.00002 Pa up to 100,000 Pa, and taking a great leap towards simplicity, introduced the BEL, and shortly thereafter the decibel. Hopefully acousticians will not abandon the decibel. Decibels are useful measures which assist in simplifying a practical understanding of sound levels. Surely simplification is a strong asset of any engineering discipline.

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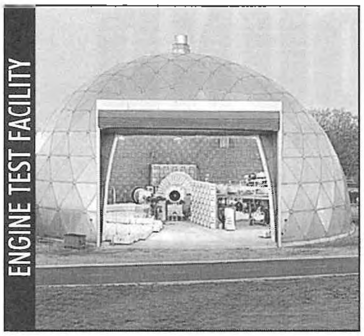
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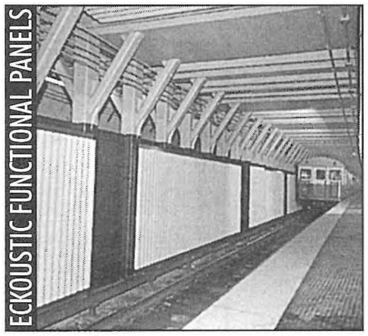
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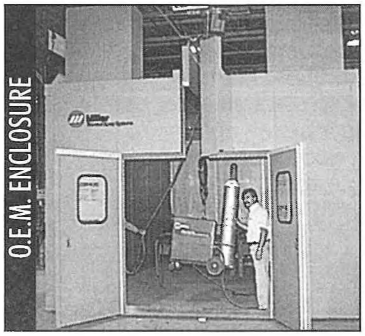
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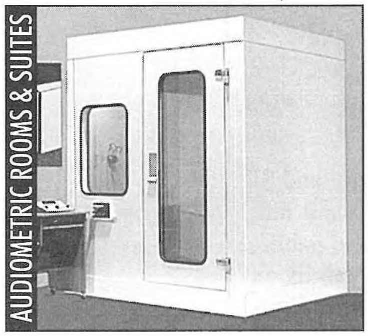
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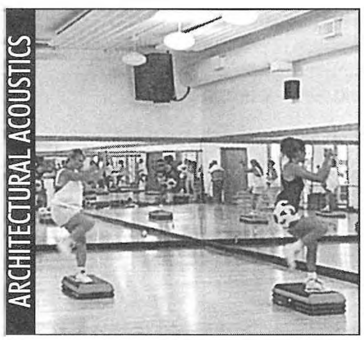
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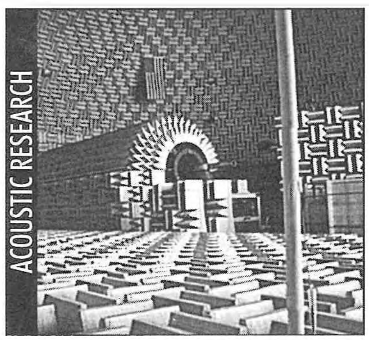
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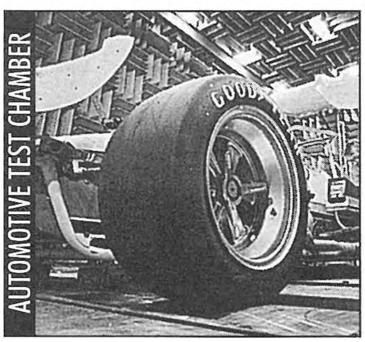
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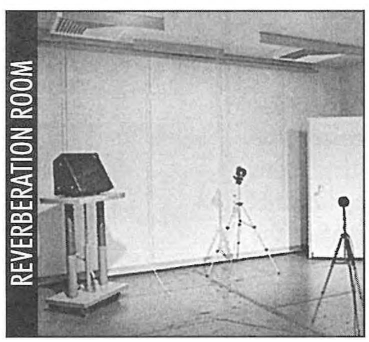
REVERBERATION ROOM

ARRAY ELEMENT LOCALISATION OF RAPIDLY DEPLOYED SYSTEMS

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ABSTRACT

Array processing techniques such as beamforming or matched field processing require accurate knowledge of the location of individual elements in the array. For horizontal arrays laid on the ocean floor, relative arrival times measured across the array from nearby implosive sources are often used to aid in estimating the sensor positions. However, the inverse problem of determining the sensor positions from the relative arrival times is both nonunique and ill-conditioned. Standard grid search techniques rely on very accurate measurements of the source locations and some knowledge of the array. This paper shows how simulated annealing can be used to solve the inverse problem with limited knowledge of the array or source locations. Synthetic studies show that relative sensor locations can be exactly found while tests with real data show an improvement in array gain comparable to the theoretical limit obtained from a perfectly known array.

RÉSUMÉ

Les techniques de traitement de signal de réseau, tel la conformation du faisceau et le traitement de champs appariés nécessitent une connaissance précise de la location des éléments individuels du réseau. Pour des réseaux horizontaux déployés sur le fond marin, les temps d'arrivée relatifs des signaux provenant de sources implosives proches, mesurés le long du réseau, sont souvent utilisés pour aider à l'estimation de la position des capteurs. Par contre, le problème inverse de la détermination de la position des capteurs à partir des temps d'arrivée relatifs est non-unique et mal défini. Les techniques de recherche sur une grille standard dépendent de la mesure très précise des positions de la source, et d'une première approximation de la position du réseau. Cet article démontre comment le traitement thermique simulé peut être utilisé pour résoudre le problème inverse avec une connaissance limitée de la position du réseau et de la source. Des études avec des données synthétiques démontrent que la position relative des capteurs peut être établie avec précision, et des essais avec des données réelles produisent une amélioration du gain de réseau comparable à la limite théorique pour un emplacement de réseau parfaitement connu.

1. INTRODUCTION

Remotely deployed systems often contain horizontal or vertical arrays mounted on the ocean floor and are used to acoustically monitor areas of the ocean. One problem with remotely deployed systems is accurately determining the sensor positions in the array. Conventional beamforming is often considered to require sensor position estimates accurate to within $\lambda/10$ where λ is the wavelength of the signal measured.[1] More advanced array processing techniques such as adaptive beamforming or matched field processing require even more accurate estimates of the sensor positions.[2,3]

Sensor positions in remote systems are often estimated by measuring correlations from nearby continuous sources,[4,5] or by measuring arrival times from nearby transient sources.[6-9] For transient sources, if the location of the sources and the travel times to the sensors are known, then the location of all the sensors in the array can be unambiguously determined using triangulation from three sources. However, the source locations are often only known approximately and the travel times from source to sensor are often unknown with the arrival times at any sensor only known relative to the arrival times at other sensors. The inverse problem of estimating the sensor positions from relative arrival times with unknown source locations is nonlinear (the radial distance between a source and sensor depends linearly on the arrival time but the sensor position also depends on the bearing to the source), nonunique (unknown source positions can allow for a rotation or translation of the combined system of sources and sensors without a change in the arrival times), and may be ill-conditioned depending on the source and sensor geometry[10] (a small error in a source position can lead to a large error in a sensor position).

One method of solving the nonlinear inverse problem is to transform it into a linearised problem and iterate towards a solution based on an initial starting model.[6,8,9] However, this is often found to be highly dependent on the initial estimate of the source locations. Another way to find the sensor positions from unknown source positions is simply to search a multidimensional space of estimated sensor and source positions and minimize the error between the measured and predicted relative arrival times. One technique designed specifically to search ill-conditioned, multidimensional spaces is called simulated annealing.[11] Although simulated annealing is also an iterative technique, it is not highly dependent on the initial estimate of the unknown parameters. This is important if source positions cannot be measured accurately, such as for sources at depth or if GPS is not available. Other authors have already used simulated annealing to localize a small vertical array with few sensors.[6] This paper will show that simulated annealing can be applied to the problem of locating a large horizontal array with many sensors and only a limited knowledge of the source locations.

2. EXPERIMENT

The data analyzed in this paper were collected during the RDS-2 trial in November 1998 in 107 m water in the Timor Sea off the northern coast of Australia at 12° 20¢ S, 128° 20¢ E. Bathymetric surveys show the area to be very flat with changes of depth of only several meters over several kilometers. A sound speed profile obtained using an XBT indicated a piecewise linear profile with a sound speed at the surface of 1549 m/sec, decreasing to 1544.5 m/sec at 75 m depth, and then increasing to 1545.5 m/sec at 107 m depth. The measured sound speed profile was used in estimating sensor positions for the real data.

The data examined were collected on the ULITE array deployed by the Space and Naval Warfare Systems Center (SPAWAR), San Diego, CA., USA. The planned deployment of the ULITE array and light bulb sources[12] is shown in Fig. 1. The array is a horizontal array which lies on the ocean floor and consists of three arms of 32 sensors each, tied in the center, with each arm containing a slight curvature to break the left/right symmetry from the arm. The sensors in each arm of the array are asymmetrically nested for three different design frequencies of 24, 48 and 96 Hz., resulting in sensor separations of 8, 16 and 32 m for sensors 1-17, 17-25 and 25-32 respectively. All light bulbs were imploded at 55 m depth with the planned locations to include three light bulbs on each side of each arm, approximately 100 m distance from the arm.

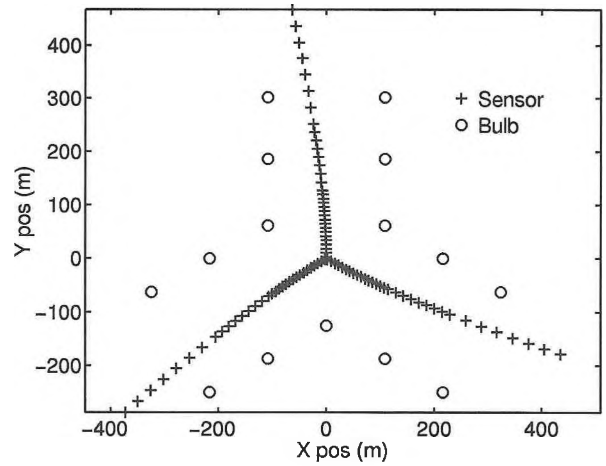

Figure 1: Trial layout plan.

The array was deployed using three boats which met at the center location for the array, then moved apart, each boat deploying one arm of the array. To keep the array from breaking, slow boat speeds were required to ensure low tension on the cables. The slow boat speeds, low cable tension, and high currents of two knots resulted in the deployment pattern of the array differing significantly from the planned deployment. The estimated array shape of the actual deployment is given in Sec. 4.

3. METHODOLOGY

This section shows how to apply simulated annealing to the problem of locating a horizontal array mounted on the ocean floor. A set of transient sources are used near mid-depth and the relative arrival times of the direct arrival and surface reflection are measured across the array. The problem then is to use simulated annealing to find a set of source and sensor locations which will reproduce the relative arrival times.

Simulated annealing involves a series of iterations in which the unknown parameters (ie. source and sensor locations) are perturbed. For each iteration, the relative arrival times of the direct arrival and surface reflection are calculated for the modelled parameters. The modelled arrival times are then compared with the measured arrival times and the total time error E is given as an estimate of the goodness of fit of the modelled source and sensor positions to their true values. For successive iterations, the change in error ΔE is calculated. If the error has decreased ($\Delta E < 0$), the new parameter configuration is accepted. If the error has increased ($\Delta E > 0$), the new configuration has a probability P of being accepted with the probability being drawn from the Boltzmann distribution:

$$P(\Delta E) = \exp(-\Delta E/T)\,, \tag{1}$$

where T is a controlling parameter analoguous to temperature in the physical process of annealing. Accepting some perturbations which increase E allows the algorithm to escape from local suboptimal minima in the search space. Decreasing T with successive iterations decreases the probability of accepting an increase in error, and the algorithm eventually converges to a solution which should approximate the global minimum.

Two factors involved in developing an efficient and effective simulated annealing algorithm are the method of decreasing the temperature T, and the method of perturbing the unknown parameters. A starting temperature T_0 was chosen which allows at least 90% of all perturbations to be accepted. This ensures that the final result does not depend on the starting estimate of the unknown parameters. A number of perturbations h are then performed before decreasing the temperature according to $T_{j+1} = \alpha\ T_j$, where $\alpha < 1$. The process is terminated when further temperature steps do not result in a lower error or when the error is within an acceptable margin. The values of η and α to use depend on the difficulty of the inversion. Increasing both η and α should decrease the final error but also increases the number of iterations and time required. An estimate of η and α can be obtained by using synthetic data and choosing η and α large enough that the final error is zero or the resulting sensor locations are accurate to within an acceptable tolerance. With real data, η and α can be initialized to the values obtained from the synthetic study and then allowed to increase until there is no further decrease in the final error.

The method of perturbing the parameters can have a major effect on the efficiency of simulated annealing. Changing only one parameter at a time allows the algorithm to converge for a sensitive parameter while continuing to search for less sensitive parameters. Changing multiple parameters in one perturbation allows for quicker convergence when coupled parameters are involved and also allows for easier jumping between local minima. Also, a parameter may be changed in different ways. In the algorithm used, when changing a parameter, either a new value is picked within a Gaussian distribution centered on the current value (allowing convergence towards a solution) or a new random value is chosen from the entire allowable range for that parameter (allowing escape from local minima).

After the simulated annealing algorithm stops, a gradient descent algorithm was applied using the result of the simulated annealing as the initial estimate. This is used to ensure that the absolute minimum of the current trough is found.

4. RESULTS

For our problem, the unknown parameters used are the source and sensor locations along with the bottom depth. For every perturbation a source, sensor or bottom depth is randomly chosen to be changed. If a source is picked, the position (in x,y and z) of only a single source is changed for a given perturbation. If a sensor is picked, either a single sensor, multiple sensors or the entire array can be changed in the following manner. The entire array can be changed by shifting it horizontally or by rotating it about some angle. A single sensor can be changed by changing its distance or bearing from the previous sensor without moving other sensors, providing that the separation between pairs of sensors does not exceed the length of cable joining them. Multiple sensors can also be changed by moving all sensors before or after a given sensor by the same change given to that sensor. For both the synthetic and real data studies, a flat bottom is assumed with all sensors considered to be at the bottom depth.

Using Fig. 1 to generate synthetic data, the simulated annealing algorithm was tested with the following uncertainties: the center of the array was assumed to be known to within 100 m; the bearing or range from one sensor to the next was unrestricted except that the range between a pair of sensors could not be greater than the length of cable separating them; the horizontal position of a source was assumed to be known within 100 m; the depth of a source was assumed to be known within 20 m; the water depth was assumed to be known within 20 m. Although the uncertainties are larger than the true uncertainties in the real data, using large uncertainties helps demonstrate the robustness of the technique. With the above uncertainties, if the relative arrival times were known exactly (ie. not digitised), then the relative array shape and light bulb positions were found within 10^{-2} m. If the relative times were only known within a digitisation sample, then the relative position of any sensor could shift from its true relative position by as much as the distance travelled by sound within the time of the digitisation sample. Increasing the number of light bulbs decreases the positional shift introduced by the digitisation.

The relative arrival times of the direct arrival and surface

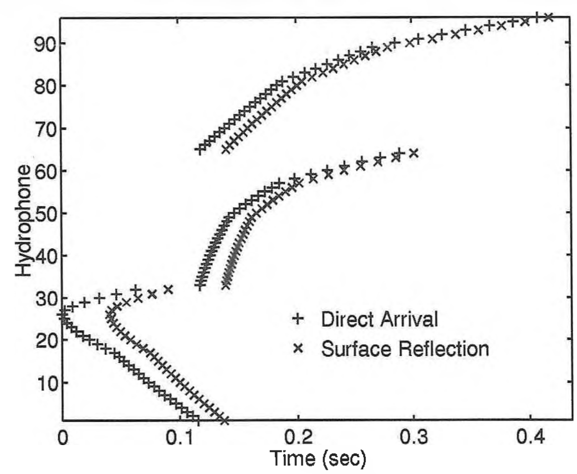

Figure 2: Relative arrival times.

reflection for synthetic data using the light bulb at (100 m, 300 m) are shown in Fig. 2. In this figure, hydrophone numbers 1-32 represent the north arm, numbers 33-64 represent the south-east arm, and 65-96 represent the south-west arm.

For the real data, The estimated source positions and array shape of the ULITE array as determined by both DSTO and SPAWAR are shown in Fig. 3. Although the shape differs significantly from the planned deployment, this is believed to be due to the high currents and the requirement to deploy the array at low speed without cable tension. The reason for the change in light bulb positions from the trial plan is simply because the array was known to have deviated from the trial plan but could not be accurately estimated on site. Thus, an estimate of where to deploy the light bulbs had to be made without accurate knowledge of the array position. The similarity of array shape estimated by DSTO and SPAWAR provides some confidence that the correct shape was obtained. The SPAWAR estimate used a grid search technique to find the hydrophone locations assuming a fixed bottom depth of 107 m and fixed light bulb locations at 55 m depth and at the recorded GPS positions. The simulated annealing technique used at DSTO allowed uncertainties of 20 m in the bottom depth, along with uncertainties of 100 m in the horizontal location and 20 m in the depth of a light bulb. The simulated annealing algorithm returned a bottom depth of 107.2 m and depths of 48-55 m for all light bulbs.

For individual arms of the array, the relative shapes estimated by DSTO and SPAWAR are very similar as shown in Fig. 4. The main difference in the estimates of the individual arms is that the SPAWAR estimates show smoother array arm shapes than the DSTO estimates. This is largely because the SPAWAR estimate only included the 16 sensors in each arm spaced by 32 m while the DSTO estimate included every sensor in all arms of the array. The roughness in array arm shapes shown by the DSTO estimate is believed to exist in the real array. If the array arms were deployed under tension, then this roughness would not be expected and an

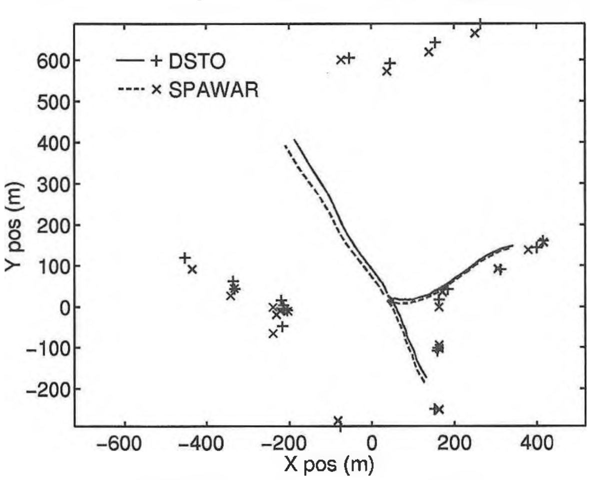

Figure 3: Estimated trial layout.

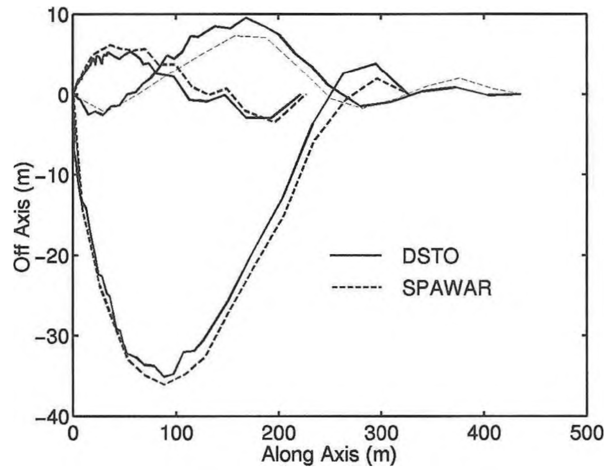

Figure 4: Estimated shapes of array arms.

inversion technique which would minimize the array curvature should be used.[8]

Although the relative shapes of individual arms are very similar, the relative shapes of the entire array show a difference of a 2° rotation in the direction of the northward pointing arm relative to the other two arms. The reason for the difference between the two estimated shapes is believed to be caused by the location of the light bulbs, which were not as tightly concentrated along the arms of the array as in the trial plan. Consider a light bulb which is near endfire to one arm and near broadside to another. A shift in the light bulb position can cause a large change in the relative arrival times across the broadside array but very little change across the endfire array. Thus, having many light bulbs near endfire of one arm can cause a relative shift in the heading between two arms of the array with little difference in the relative arrival times. This is the case for the ULITE deployment which has a large number of sources, the light bulbs west of the array, that are near endfire to the east arm of the array and near broadside to the north arm of the array.

When the uncertainty in the horizontal positions of the light bulb sources was reduced from 100 m to 10 m, the shape of the individual arms remained nearly identical and the rotation difference in the north arm of the array as estimated by DSTO and SPAWAR disappeared. This gives good confidence in the shape of individual arms of the array, but does not necessarily indicate which of the headings for the north arm of the array is correct. To determine this, the errors between the measured and modelled arrival times for the estimated source and sensor positions must be examined.

The errors between the measured and modelled arrival times are shown in Fig. 5 for the case when source positions are only known within 100 m. Each horizontal line shows the error in arrival times for both the direct arrival and surface reflection from all light bulbs. Nearly all errors fall within

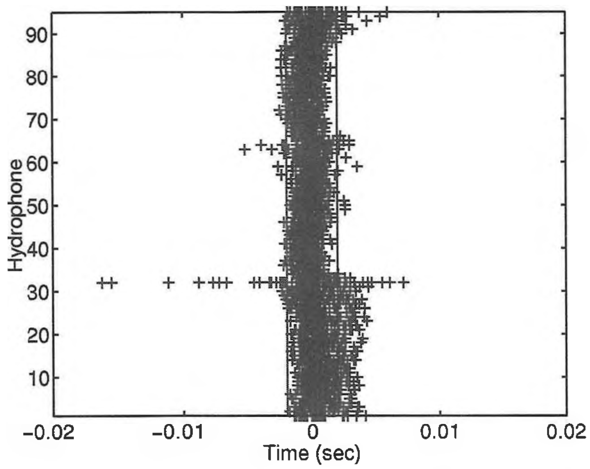

Figure 5: Arrival time errors.

the digitisation sample size of 0.002 sec, with the average error being 0.0009 sec per arrival (0.45 digitisation samples). This provides further confidence that the true array shape is well approximated. Sensor 32 is the outermost sensor on the northward pointing arm of the array and was connected to a surface buoy. It is believed that the buoy was causing this sensor to move and thus, an accurate estimate of its position could not be found, and it contains large errors in the arrival time estimates. This was also found by SPAWAR. This sensor is not plotted in Figs. 3 or 4.

When the uncertainty in the source locations was decreased to 10 m, the average error fell to 0.0007 sec per arrival (0.34 digitisation samples), and nearly all errors larger than the digitisation sample size occurred either for the first three light bulb sources measured on the north arm of the array, or for sensor number 32. The smaller error obtained using the tighter bounds on the light bulb positions indicates that this is likely a better estimate of the heading of the north arm of the array than that determined from the loose bounds on the light bulb positions. However, the larger errors obtained for the north arm of the array than for the east or south arms indicate that the north arm of the array may not be estimated as well as the other two arms. This was consistent using tight or loose bounds on the light bulb positions. The larger error in the shape of the north arm is believed to be caused both by the buoy which may be dragging that arm of the array in the high currents, and by the lack of nearby light bulb sources along the length of this arm of the array. Both the east and south arms have multiple light bulbs within 50 m of the hydrophones while the closest light bulb to the north arm is over 200 m away.

Determining the number and location of light bulbs required to accurately estimate an array shape is array dependent. A study of a single, nearly linear array of 200 m length showed that four light bulbs with two along one side, one along the other side and one near endfire always provided solutions accurate to within the distance travelled by sound within the time of the digitisation sample, provided that the light bulbs were within 200 m of the array. A more complete analysis of optimal source locations is available in Dosso and Sotirin.[10]

A final indication of the accuracy of the estimated array shape can be obtained by beamforming on real data that contains only a single target with a large snr. Under this condition, the total energy measured across the array (or averaged into a covariance matrix) should be reproduced when beamforming at the target location. During the trial, an 80 Hz tonal target was deployed at 100° relative to North and 3200 m from the center of the array. The conventional beamformer reproduced 80%, 97%, 97% and 75% of the total power measured across the north arm, east arm, south arm and full array respectively when steered at this target. This represents a loss of only 1.0 dB, 0.1 dB, 0.1 dB and 1.2 dB respectively. Thus, the shapes of the east and south arms are assumed to be very accurately estimated while the north arm still contains some error. This was also indicated in Fig. 5 which showed that the largest arrival time errors were from the north arm.

The beamformed output of each individual arm of the array and of the full array is shown in Fig. 6. This figure shows the results of conventional focussed beamforming on the real data (solid line), conventional focussed beamforming on synthetic data for a source at the known target location and in infinitely deep water (dashed line), and adaptive focussed beamforming on the real data (dotted line). The minimum variance distortionless response (mvdr) beamformer was used as the adaptive beamformer. The close agreement between the real and synthetic data for the east and south arms again shows that the shapes of these arms are well estimated. The close agreement also indicates that very little signal loss occurs due to the multipath effects of the shallow water environment for this frequency and range. Finally, the shapes of the east and south arms are estimated accurately enough to give good performance for the adaptive beamformer, which is known to be highly sensitive to errors in the array shape.[2] Note that the adaptive beamformer nearly eliminates the large sidelobes of the conventional beamformer. Although the adaptive beamformer still returns the correct signal directions for the north arm and the full array, the array gain is poor.

5. Summary

This paper has shown how simulated annealing can be used to accurately perform array element localisation on remotely deployed systems. Synthetic studies have shown that the relative sensor positions can be determined within the distance travelled by sound during the time of the digitisation sample size if sufficient light bulbs are employed nearby along the

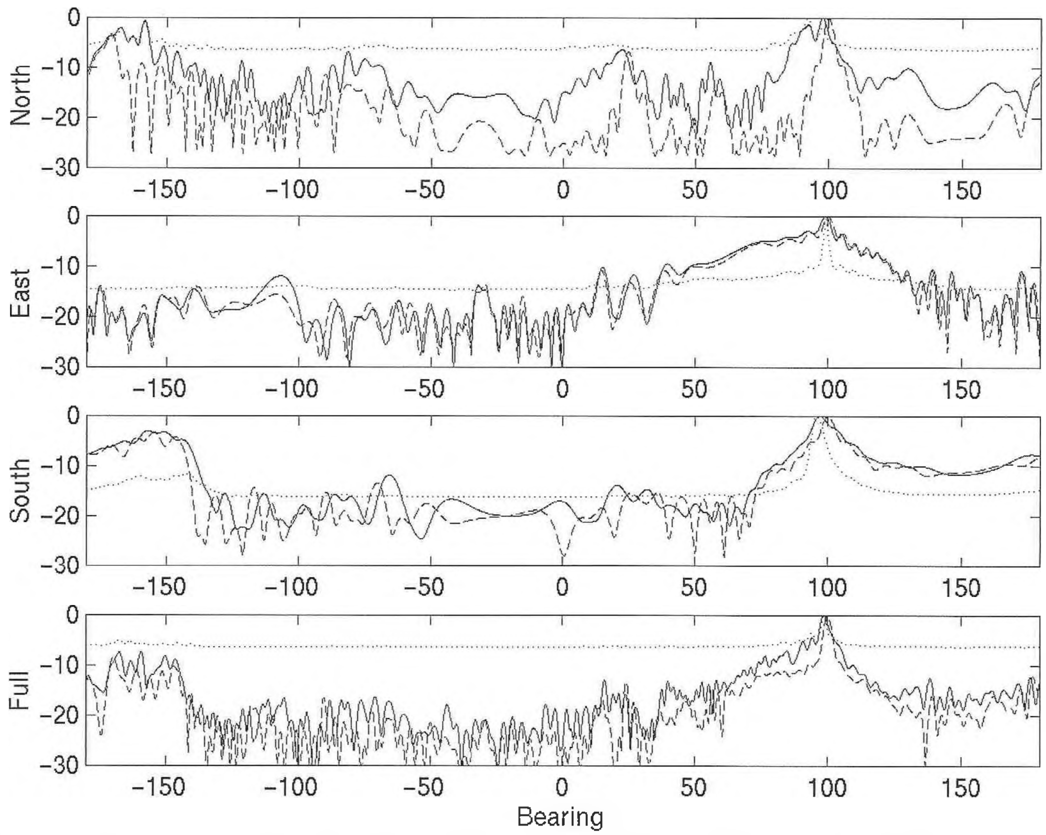

Figure 6: Beamformed output (in dB) of the north arm, east arm, south arm and full array. Results show the conventional beamformer on real data (solid line), the conventional beamformer on synthetic data with a target at the known true target location (dashed line), and the mvdr adaptive beamformer on real data (dotted line). The true target is at 100°, 3200 m.

array and at endfire to the array. Although ground truth was not available on the real data trial, there are four indications that the array shape is well estimated. These are: agreement with an independently performed localisation estimate; reproduction of up to 97% of the total energy measured across an individual arm of the array when beamformed at a dominant source; close agreement between the beamformed results for real data and for synthetic data using a single target at the known source location; and finally, good beamforming performance using the adaptive beamformer which is known to be sensitive to errors in array shape.

6. ACKNOWLEDGEMENTS

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ANNOUNCEMENT

To all my colleagues:

Many of us perform both acoustic and vibration work. Obviously, those of us in the acoustic side are members of the CAA. We interact with our peers by reading the journal and attending the annual meeting. For the vibration side, I would like to suggest participation in the Canadian Machinery Vibration Association, **CMVA**. There are regular Chapter meetings across Canada, with two chapters here in Ontario (Central and Eastern). For further information see the website **www.cmva.com**, or call me directly.

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COGÉNÉRATION BRUIT DU DÉLESTAGE DE VAPEUR

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SOMMAIRE

Une centrale électrique a été aménagée sur le site d'enfouissement de déchets domestiques de Montréal (ancienne carrière Miron) en 1996. Cette centrale s'alimente du biogaz se dégageant de la décomposition des déchets. Le biogaz se compose principalement de méthane. En raison de la localisation de la centrale, à proximité de zones résidentielles, une attention particulière sur le bruit avait été apportée au stade de l'ingénierie afin de s'assurer du respect des règlements applicables. Lors de la mise en service, il a été constaté que la centrale était conforme en mode d'opération continu. Toutefois, les opérations de délestage de vapeur, requises lors des cyles de démarrage et d'arrêt de la centrale, ainsi qu'en cas d'urgence, dépassaient largement les critères applicables entraînant des plaintes de la part des citoyens. Le bruit provenait de la détente rapide de la vapeur à l'atmosphère dans une valve à tournant (*globe valve*) dont la décharge donnait à l'extérieur. La mise en place d'un silencieux sur cette valve a permis d'éliminer la nuisance sonore et de respecter les normes applicables. Cependant, la performance fût moindre qu'anticipée selon les données du fabricant. Il est à conclure, par ailleurs, que les sources sonores de nature sporadique sont toutes aussi importantes que celles associées aux opérations continues. Toutes les conditions d'opération d'une centrale d'énergie doivent être considérées à l'ingénierie détaillée d'un projet afin d'assurer sa conformité aux critères acoustiques applicables.

ABSTRACT

A power plant was set up on the Montreal domestic waste landfill site (former Miron quarry) in 1996. This plant is powered by biogas given off during waste decomposition. The biogas is composed chiefly of methane. Due to the location of the plant near residential zones, noise was given special consideration during the engineering phase in order to ensure compliance with the applicable regulations. At commissioning, the plant was found to be compliant in continuous operation mode. However, the steam shedding operations, required during start-up and shutdown cycles as well as in emergency situations, greatly exceeded the applicable standards, drawing complaints from the citizens. The noise was caused by the rapid release of steam into the atmosphere through a globe valve that discharged to the outside. The installation of a muffler on this valve brought the noise emissions within the applicable standards. However, the performance was less than anticipated based on the manufacturer's data. It should also be concluded that sound sources of a sporadic nature are just as important as those associated with continuous operations. All the operating conditions of a power plant must be considered in the detail engineering of a project in order to ensure its compliance with the applicable acoustic standards.

1. INTRODUCTION

L'ancienne carrière Miron de Montréal, maintenant dénommée le complexe environnemental Saint-Michel, est utilisée comme site d'enfouissement des déchets domestiques. Les déchets sont recouverts d'une membrane qui permet de capter les biogaz provenant de leur décomposition. Ces gaz sont collectés grâce à un réseau de puits maintenus sous vide, canalisés vers un poste de surpression et ensuite acheminés vers une centrale d'énergie reliée au réseau d'Hydro-Québec. Les gaz sont brûlés dans une bouilloire à vapeur. La vapeur ainsi obtenue est détendue dans une turbine qui entraîne une génératrice de 25 MW. Le schéma de principe de cette unité de revalorisation des déchets domes-

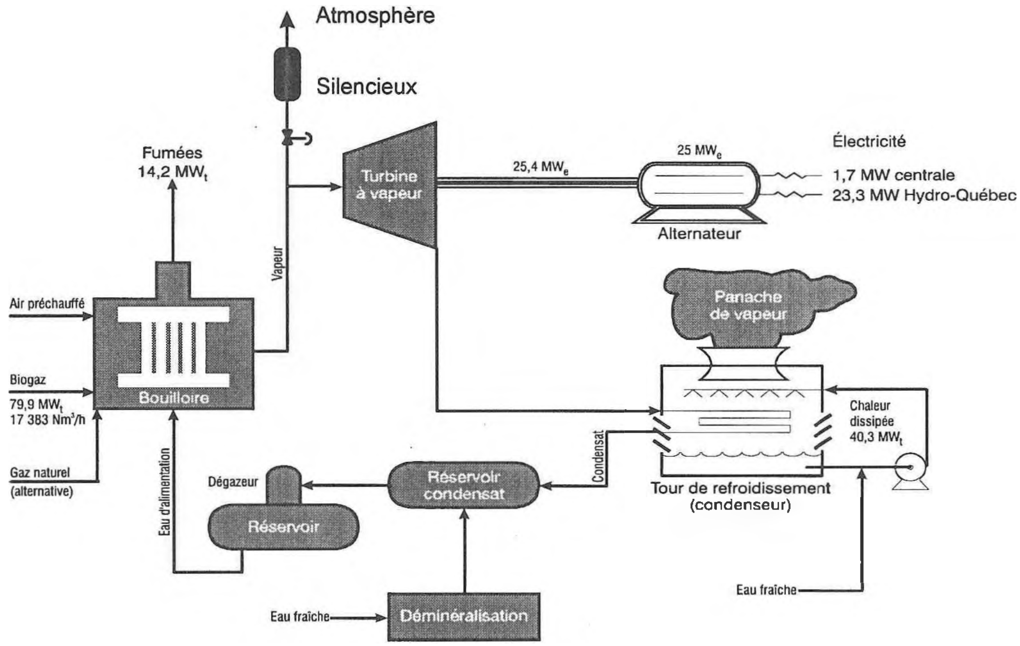

Figure 1 : Schéma de principe de la centrale de cogénération

tiques est montré à la Figure 1.

La centrale est située dans un milieu urbain, à proximité de secteurs résidentiels et commerciaux. L'étude d'impact environnemental du projet avait mis en évidence la proximité des résidences et évalué le niveau sonore maximum à respecter pour se conformer à la législation applicable et assurer l'acceptabilité du projet.

La législation applicable comportait deux volets, l'un attribuable au palier municipal et l'autre au palier provincial, ces deux volets devant être respectés concurremment. Cette législation nécessitait la détermination du climat sonore préexistant, ce qui avait fait l'objet de relevés sonores de longue durée sur le terrain dans le cadre de la réalisation de l'étude d'impact sur l'environnement. Par ailleurs, puisque la centrale électrique opère sur un mode continu, les limites les plus restrictives de la législation ont été utilisées, soit celles s'appliquant la nuit. Ajoutons que pour simplifier le présent texte, les limites de bruit sont présentées uniquement pour le récepteur le plus exposé parmi les onze considérés dans l'étude d'impact sur l'environnement.

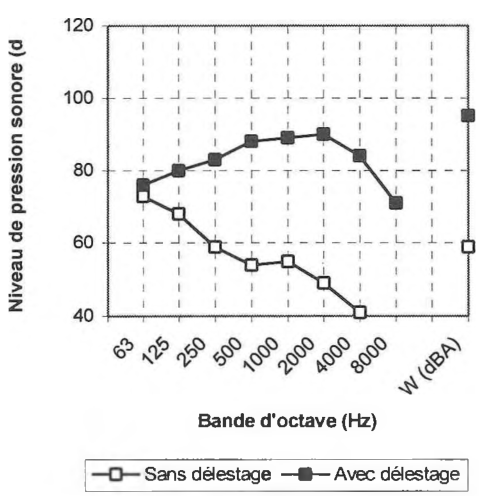

Figure 2 : Spectre sonore du délestage de vapeur, mesuré à 100 m de la centrale.

Le règlement municipal, Ville de Montréal, portant le numéro 4996, en vigueur en 1996, impose des limites de bruit à ne pas excéder selon différents paramètres touchant l'emplacement où le bruit est perçu (cour extérieure, chambre à coucher, ...), le type de bruit émis, le niveau de bruit de fond (L95) dans le quartier et finalement la période de la journée (jour, soirée, nuit). Cette limite pondérée, au récepteur considéré, est de 48 dBA.

Pour ce qui est du palier provincial, une directive est utilisée par le ministère de l'Environnement qui impose elle aussi des niveaux de bruit à ne pas excéder, uniquement à l'extérieur, selon le type de milieu où le bruit est perçu (type déterminé par le zonage municipal), le niveau de bruit ambiant (Leq) (sur une base horaire) dans ce milieu et finalement la période de la journée (jour, nuit). Cette limite, au récepteur considéré, est de 49 dBA, ce qui correspond au bruit ambiant minimal mesuré la nuit au récepteur le plus critique.

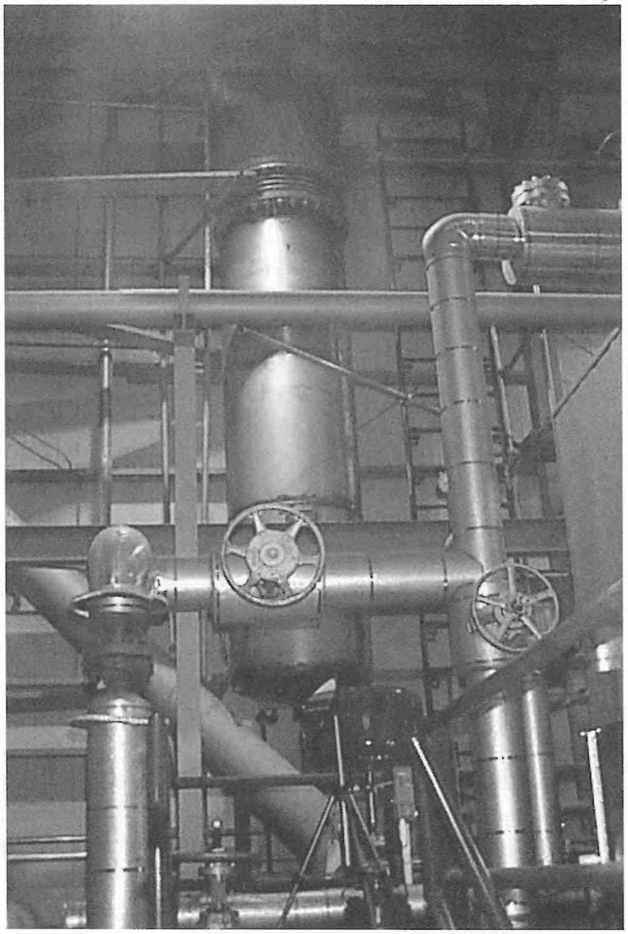

Figure 3 : Silencieux réactif/absorbtif, photo

La limite qui permet de respecter l'ensemble de la législation considérée, est donc 48 dBA pour le récepteur considéré.

Dès l'ingénierie préliminaire de la centrale, l'aspect acoustique a été pris en compte afin de réduire, à la source, le bruit des composantes majeures de la centrale telles les aérorefroidisseurs et les ventilateurs à tirage induit.

Lors de la mise en service de la centrale, l'ensemble des sources continues respectaient le niveau équivalent horaire de 49 dBA à la résidence la plus proche.

Toutefois, à la mise en service en 1996, la centrale a fait l'objet de nombreuses plaintes de la part des résidants avoisinants. Les plaintes étaient reliées au délestage de la vapeur lors des arrêts et démarrages de la centrale et non à l'exploitation de la centrale en service continu.

2. PROBLÉMATIQUE

Pour donner suite aux plaintes des résidants, le service de contrôle du bruit de la Ville de Montréal a effectué des relevés de bruit afin de déterminer si le bruit émis lors des délestages de vapeur est une nuisance au sens de la réglementation municipale. Le bruit du délestage est de type fluctuant. Il augmente et diminue en fonction de la quantité de vapeur délestée avec des durées variant de quelques minutes à plus d'une heure. Les dépassements constatés variaient de 8 à 12 dBA aux secteurs résidentiels et de 19 à 33 dBA aux bureaux de la Ville de Montréal situés sur le complexe environnemental Saint-Michel. Comme le montre le spectre de la figure 2, mesuré à 100 m de la centrale d'énergie, le bruit du délestage de la vapeur est dominé par les bandes d'octaves de 500, 1000, 2000 et 4 000 Hz. L'impression subjective est celle d'un sifflement ou d'un moteur d'avion.

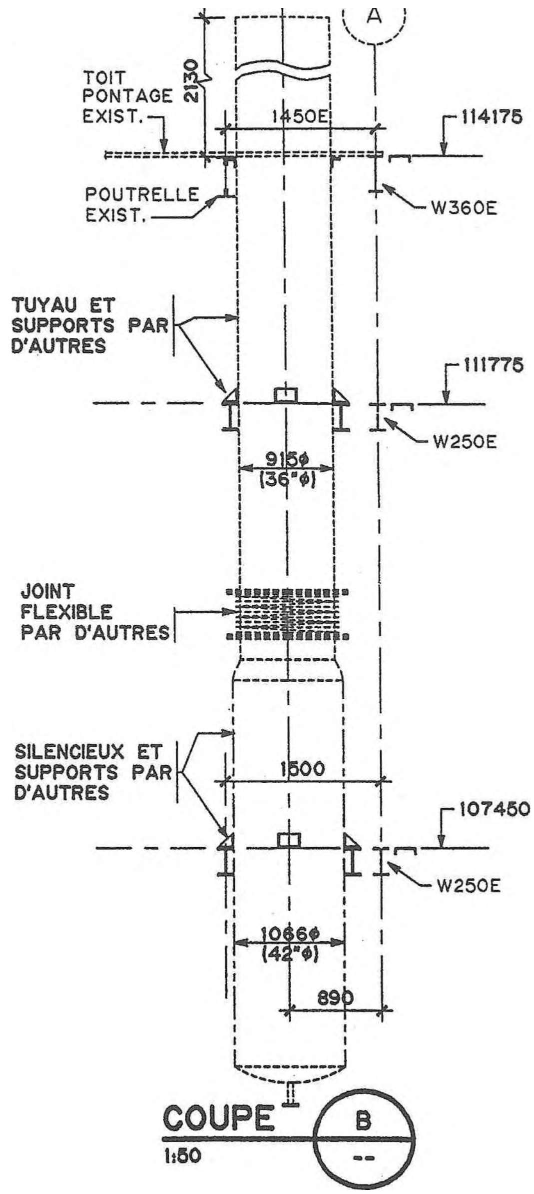

Figure 4 : Installation du silencieux

Le délestage de la vapeur est nécessaire lors des arrêts et

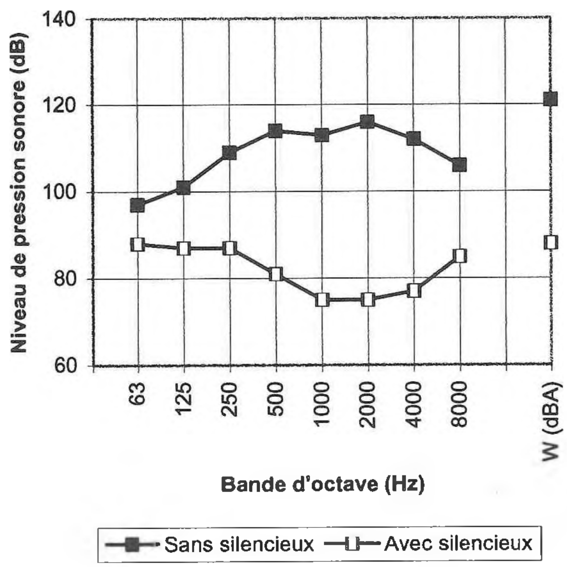

Figure 5: Spectre sonore avant et après l'installation du silencieux, mesuré à 10 pieds et 90 degrés de la décharge.

démarrages de la centrale. Lors des démarrages, la vapeur est délestée jusqu'à ce que les conditions requises pour la mise en ligne du turboalternateur soient atteintes. Lors des arrêts, la vapeur est délestée pour dépressuriser la bouilloire. Pendant la période de rodage et de mise en service de la centrale, de nombreux départs et arrêts ont été effectués, ce qui a aggravé la nuisance pour les résidants avoisinants.

3. SOURCE DE BRUIT

La vapeur de la bouilloire est délestée via un robinet à soupape relié à une conduite de 2 po de diamètre. Cette conduite est raccordée à un évent de 10 po de diamètre qui se termine sur le toit de la centrale. La valve abaisse la pression de 675 psi à partir d'une pression de 1 250 psig en amont. Le bruit est généré lors de la détente de la vapeur, en un seul stade, dans le robinet à soupape. Il se propage ensuite avec très peu d'atténuation dans les conduites jusqu'à l'extérieur.

4. MOYENS D'ATTÉNUATION

L'objectif de réduction fut établi à plus de 50 dBA afin de pouvoir atténuer le bruit aux conditions maximales de délestage en cas d'urgence (35 000 lb/h à 950°F).

Pour réduire le bruit du délestage de vapeur, deux options ont été évaluées, soit : le remplacement de la valve à globe, à un stade, par une valve moins bruyante, à plusieurs stades, assurant une détente plus progressive de la vapeur ou l'addition d'un silencieux en aval de la valve. Le remplacement de la valve n'a pas été retenu compte tenu que l'objectif de réduction ne pouvait être atteint et qu'un silencieux aurait été requis en plus.

Un silencieux de type diffusif/absorbtif fut spécifié pour contrôler le bruit du délestage de la vapeur. Le silencieux est raccordé directement à la conduite de 2 po de diamètre en aval de la valve à globe. Il est composé d'une première chambre de diffusion où le jet est séparé en plusieurs petits jet, suivi d'une deuxième chambre avec absorbant acoustique. Le silencieux est en acier inoxydable afin d'assurer une longévité comparable à celle de la centrale et mesure 42 po de diamètre, 14 pi de long, tel que montré à la Figure 3. Il est installé à l'intérieur de la centrale et relié à une cheminée de 36 po de diamètre par l'intermédiaire d'un joint d'expansion. La cheminée passe à travers le toit de la centrale pour se terminer à 7 pi au-dessus du toit tel que montré à la Figure 4. Le tout est supporté par une structure en acier.

5. SUIVI ACOUSTIQUE

Pour des fins de vérification, le niveau sonore fut mesuré avant et après l'installation du silencieux. Les niveaux mesurés à une distance de 10 pieds et 90 degrés de la décharge sont présentés à la Figure 5. La perte par insertion obtenue est de 33 dBA à 22 807 lb/h. La performance mesurée est suffisante pour se conformer à la réglementation municipale aux résidences avoisinantes et aux bureaux de la Ville de Montréal situés sur le complexe.

Aux résidences, le bruit du délestage, suite à l'implantation du silencieux, est à peine audible. Toutefois, la performance mesurée est inférieure à celle prévue. Les causes probables de la réduction de la performance anticipée sont :

- Selon la fiche technique du fournisseur, le silencieux est normalement raccordé à une ligne de 12 po de diamètre au lieu d'une ligne de 2 po de diamètre, tel qu'installé à la centrale. La vitesse et la longueur du jet à l'intérieur du silencieux sont plus élevées que pour la conception standard. Le jet à l'intérieur du silencieux peut se prolonger au-delà de la chambre de diffusion et en réduire la performance. De plus, le jet peut frapper une paroi interne du silencieux et générer du bruit. Une augmentation du niveau sonore à l'intérieur de la centrale a été constatée, principalement à proximité de la chambre de diffusion du silencieux.

- Avant l'installation du silencieux, la vapeur était délestée à l'extérieur par une conduite de 10 po de diamètre alors que la cheminée, à la décharge du silencieux, est de 36 po de diamètre. L'augmentation du diamètre de la décharge a pour effet de réduire le coefficient de réflexion à la décharge, ce qui facilite le passage du bruit de l'intérieur de la conduite vers l'extérieur.

6. CONCLUSION

Le délestage de vapeur lors des arrêts et démarrages de la centrale est une source importante de bruit qui a entraîné des plaintes de la part des résidants avoisinants. Les arrêts et démarrages fréquents lors du rodage et de la mise en service de la centrale ont contribué à aggraver la nuisance sonore. Le bruit du délestage de la vapeur est généré par la détente abrupte dans une valve à globe dont la décharge donne à l'extérieur. L'installation d'un silencieux à la décharge de la valve a permis de réduire le bruit du délestage de 33 dBA et de se conformer à la réglementation municipale. Lors de la conception d'une nouvelle centrale, le choix d'une valve à détente progressive, à plusieurs stades, et l'installation d'un silencieux devraient être considérés afin d'éviter les nuisances sonores lors du rodage et de la mise en service.

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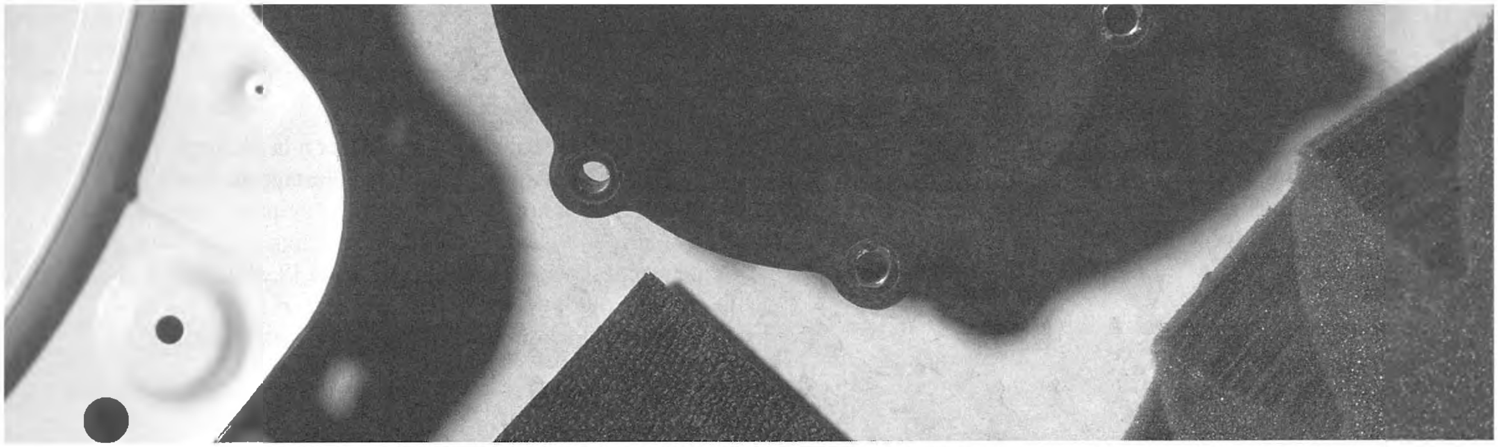

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BINAURAL TECHNOLOGY FOR APPLICATION TO ACTIVE NOISE REDUCTION COMMUNICATION HEADSETS: DESIGN CONSIDERATIONS

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SUMMARY

This article examines the fundamental basis and the technical aspects involved in integrating two emerging technologies in the design of communication headsets for use in noisy environments. The first technology, active noise reduction (ANR), can improve signal detection and speech intelligibility by reducing the amount of interfering noise from the environment. The second technology, known as binaural technology, allows the creation of 3D auditory displays, which can improve signal detection and speech intelligibility in noise, and situational awareness, over monaural listening. For an optimal integration of binaural technology into ANR headsets, digital devices are preferred over analog devices. The complexity of the integrated system, particularly the features of the binaural simulation, is found to be largely dependent on the specific demands of the application targeted. Two extreme cases relevant to an aircraft cockpit environment are analyzed. The greatest benefit is likely to be found in situations of divided attention listening in relatively low signal-to-noise environments.

SOMMAIRE

Cet article examine les principes de base et les aspects techniques nécessaires à l'intégration de deux technologies émergentes dans la conception de casques d'écoute pour les milieux bruyants. La première technologie, le contrôle actif du bruit, permet d'améliorer la détection de signaux et l'intelligibilité de la parole en réduisant l'interférence causée par le bruit environnant. La deuxième technologie, la technologie binaurale, permet de créer un environnement d'écoute 3D, ce qui en retour permet d'améliorer la détection de signaux et l'intelligibilité de la parole dans le bruit, ainsi que la vigilance en situation d'écoute, par rapport à l'écoute monaurale. L'utilisation de casques actifs numériques est préférable aux casques actifs analogiques pour assurer une intégration optimale avec la technologie binaurale. La complexité du système total, tout particulièrement les caractéristiques de la simulation d'écoute binaurale, dépend en grande partie des exigences de l'application ciblée. Deux situations extrêmes appliquées à un environnement de cockpit d'avion sont analysées. L'avantage d'appliquer la technologie binaurale aux casques actifs sera le plus important en situation d'écoute où l'attention doit être partagée entre plusieurs signaux dans des milieux dont le rapport signal au bruit est faible.

1.0 Introduction

This study was undertaken to evaluate the feasibility of applying binaural technology to the design of active noise reduction (ANR) communication headsets. The long-term objectives of combining both technologies are the improvement of the intelligibility of competing spoken messages presented simultaneously in the presence of noise, and the enhancement of situational awareness in complex auditory listening environments. ANR technology (Steeneken and Verhave, 1996) can improve speech intelligibility by reducing the amount of interfering noise from the environment. This is accomplished by electronic sound wave cancellation of the environmental noise inside the earcups of the device. Binaural technology (Moller, 1992), on the other hand, allows the transfer of coincident messages to different virtual spatial positions by filtering the incoming communication signals with the head-related transfer functions of the user. This processing generates interaural time difference (ITD) and interaural level difference (ILD) cues for each message. Variation in these cues normally signifies real-world differences in spatial location (Blauert, 1997), and impacts on the intelligibility of speech in noise (Bronkhorst and Plomp, 1988).

In this article, we begin by reviewing the fundamental research on binaural speech intelligibility in noise and binaural technology. The practical aspects involved in the creation of directional audio signals through ANR headsets are then discussed in terms of the required technical characteristics of the devices, and the application requirements. The process of integrating binaural technology into ANR headsets is illustrated through two different listening scenarios relevant to an aircraft cockpit environment. The potential benefits are assessed.

2.0 Binaural Signal Detection And Speech Intelligibility in Noise

Incident acoustic signals are transformed by the complex geometry of the human torso, head and external ear (Shaw, 1974). This filtering produces direction-dependent sound spectra at the ears, and encodes time and level differences in the sound across the left and right ears. Binaural analysis of these cues provides the basis for localizing sound sources in space (Blauert, 1997).

In addition, the detection, discrimination and recognition of a sound signal in the presence of other signals or noises can sometimes be markedly improved when listening binaurally rather than monaurally (Yost, 1997). It has long been suggested that the binaural hearing cues could play a major role in separating sound sources perceptually (Cherry, 1953). Data on the benefits of binaural over monaural listening have been collected over the past decades.

2.1 Headphone studies

A representative early study was conducted by Levitt and Rabiner (1967a). They presented speech signals interaurally out-of-phase over headphones in the presence of a broadband white noise masker interaurally in-phase ($S_\pi N_0$). They found a masked threshold for speech detection about 13 dB lower than if both signal and noise were presented interaurally in-phase (S_0N_0). Further experiments showed that this release from masking for detection was determined primarily by interaural phase opposition in the low-frequency region of the speech signal, typically below 0.5 kHz. In contrast, the maximum binaural gain at the 50% intelligibility level, i.e., the maximal decrease in speech reception threshold (SRT) with respect to the S_0N_0 condition, was about 6 dB and required interaural signal phase opposition over a wide frequency region. Presenting the speech signal in-phase with an interaurally uncorrelated noise masker (S_0N_u) led to a small decrease of about 3 dB in the masked detection threshold, but no advantage for speech intelligibility. In all conditions investigated, the binaural gain was substantially lower than the corresponding decrease in masked threshold. Levitt and Rabiner (1967b) predicted that even lower binaural gains could be expected for reference intelligibility levels greater than 50%.

2.2 Sound-field studies

Binaural speech intelligibility has also been investigated in rooms using spatially-separated loudspeakers for signal and noise sources (Figure 1). For example, Plomp and Mimpen (1981) measured the SRT for normal listeners in an anechoic room for a frontal speech signal, as a function of the azimuthal position θ of a speech noise masker. They found a general decrease in the SRT when the noise source was displaced from frontal to lateral positions. A maximal decrease in SRT, or binaural gain, of about 9-11 dB was found for a noise azimuth close to θ =90°.

Santon (1986) performed a similar experiment and found a maximal decrease in SRT of about 8 dB when a broadband white noise masker of moderate level was displaced from a frontal to a lateral position. If the broadband masker was divided into two noise bands, above and below 1.4 kHz, then the maximal decrease in SRT for each band was limited to 3 dB. Further experiments (Santon, 1987) showed that the variations in SRT for low (0.125-0.8 kHz) or mid (1-2 kHz) frequency noise bands were smaller than the corresponding variations in detection threshold for pure tones near the centre frequency of the bands, but followed the same trends as a function of noise azimuth. In the case of a high-frequency (2.5-6.3 kHz) noise band masker, the variations in SRT were also smaller than the corresponding variations in detection threshold, but the trends as a function of noise azimuth differed.

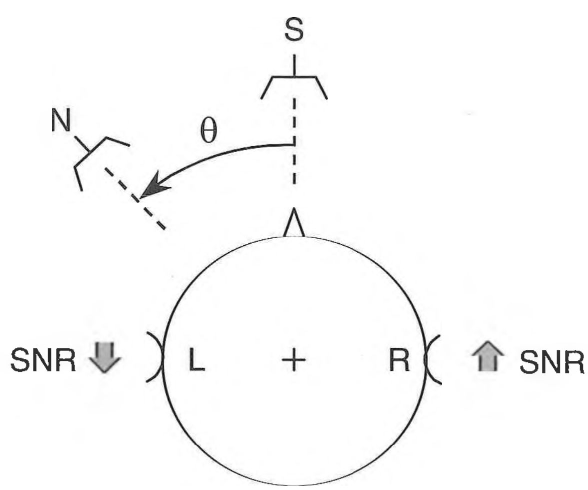

Figure 1: Schematic representation of binaural listening in sound field. L: left ear, R: right ear, S: signal source, N: noise source, θ: azimuth angle, SNR: signal-to-noise ratio.

Januška (1983) measured speech intelligibility for a frontal speech signal as a function of the level and spatial location (frontal, lateral, behind, above) of different noise maskers. Data were obtained for broadband white noise and octave bands of noise centred on 0.125 kHz to 8 kHz. The effect of varying the reverberation time of the listening space (anechoic, 0.4 s and 2.0 s) was also investigated. A coincident position of speech and noise sources was always the most unfavourable condition. Both the masking effect and the benefit of displacing the masker away from the frontal position were always greater for the broadband than the octave-band noises. Speech intelligibility gains were found for all conditions of the listening space, but were most evident in the anechoic environment. Typically, for broadband white noise, the maximum binaural gain at the 50% speech intelligibility level was about 14 dB in the anechoic environment, and 8 dB and 6 dB in the rooms with reverberation times of 0.4 s and 2.0 s respectively.

2.3 Simulated sound-field studies

In the sound-field studies discussed above, the speech and noise levels were defined with respect to the free field, typically at the head position in absence of the listener. Due to the direction-dependent transfer function of the external ear (Shaw, 1974), the actual signal-to-noise ratio (SNR) at the listener's ears will vary when the speech signal and/or noise sources are spatially displaced, and will be different across the two ears. For example, in Figure 1, when the noise source is displaced laterally, the noise level increases in the ipsilateral ear (SNR decreases) and decreases in the contralateral ear (SNR increases). Thus, the speech intelligibility benefit of spatially separating signal and noise sources from a common position may include a monaural contribution from the ear with the best SNR, as well as the contribution from binaural processing per se. Also, interaural time and level differences cannot be independently controlled in sound-field experiments, and thus their respective roles cannot be separated in the interpretation of speech intelligibility results.

To address these questions, Bronkhorst and Plomp (1988) simulated free-field conditions over headphones by presenting speech and noise signals recorded a-priori on a KEMAR manikin in an anechoic room. The speech signal recordings corresponded in all conditions to a frontal sound incidence. The noise recordings were made at several azimuth angles θ in the horizontal plane, and from each recording two additional noise signals were derived by computer processing, one containing only ITDs and one containing only ILDs. The results for normal-hearing listeners showed, as in Plomp and Mimpen (1981), a gain of about 10 dB in SRT, when the noise containing both ITDs and ILDs was presented laterally relative to the frontal position. In the same conditions, the noise containing ITDs alone provided a gain of about 5 dB, and the noise containing ILDs alone provided a gain of about 7 dB. Thus, the effects of ITDs and ILDs were not additive. The benefit of the ITD cues was essentially unaffected by simulating a one-sided attenuation of 20 dB at either ear. Also, the effect of the ILD cues was entirely dependent on monaural processing and not on binaural processing per se, since the same gain in intelligibility could be obtained by listening only through the ear with the best SNR. Overall, for a frontal speech signal and a lateral noise masker, the minimum and maximum gains observed for binaural listening compared to monaural listening were 2.5 dB and 13.2 dB respectively. The higher value is the binaural gain compared to monaural listening through the ear with the worst SNR, and the lower value is the binaural gain against the ear with the best SNR.

Bronkhorst and Plomp (1989) extended their experiments to hearing-impaired listeners. These listeners had a 2.5 dB higher SRT than normal-hearing listeners when the speech signal and noise masker were presented from the front, and a 2.6-5.1 dB smaller binaural intelligibility gain when the noise masker was displaced laterally depending on the configuration of the hearing loss. The shortfall in binaural gain for hearing-impaired listeners was mainly due to an inability to take full advantage of ILD cues. This was especially pronounced for listeners with asymmetrical high-frequency hearing losses when the noise source was displaced contralaterally to their best ear. In contrast, the gain in speech intelligibility due to ITD cues was less affected by hearing impairment. It was about 4-5 dB for normal-hearing listeners and listeners with symmetrical losses, but 2.5 dB for listeners with asymmetrical losses. When ITD cues were introduced in a noise already containing ILD cues, the resulting gain was 2-2.5 dB for both groups of hearing-impaired listeners.

Bronkhorst and Plomp (1992) further investigated binaural speech intelligibility in simulated free-field conditions with a frontal speech signal source in the presence of one to six mutually-uncorrelated noise sources located in the horizontal plane in various configurations. Over all conditions, the hearing-impaired listeners needed a 4.2-10 dB better SNR than normal listeners for equal intelligibility. The binaural advantage arising when the noise maskers were displaced from the frontal position to symmetrical or asymmetrical spatial configurations around the listeners varied from 1.5 to 8 dB for normal listeners, and from 1 to 6.5 dB for hearing-impaired listeners. The higher value corresponds to a single masker moved laterally to the side of the listeners, and the lower value corresponds to a configuration of six maskers located symmetrically around the listeners at 60° intervals. Comparison of binaural listening with monaural listening results through the best ear showed a fairly constant binaural advantage of about 3 dB across noise masker configurations and listener groups.

In summary:

- the advantage of binaural over monaural listening in noise is greater for detection than intelligibility tasks;
- only the ITD cues provide a true benefit for speech intelligibility in noise;
- the effects of ILD cues can be fully accounted by monaural SNR considerations alone;
- the maximum speech intelligibility benefit derived from binaural listening over monaural listening through the ear with the best SNR is limited to 4-6 dB under the most favourable conditions (anechoic environment, normal hearing or symmetrical hearing loss, single noise source spatially separated from the speech signal, broadband noise, low overall SNR); and
- the benefit of binaural hearing decreases with increasing signal level above masked threshold, and is very small when the signal is relatively easy to detect (Yost, 1997).

Most experiments described to date have been devoted to measures of selective attention, where the listener is asked to focus on a particular signal source and ignore all others (Yost, 1997). There is very little information on situations of divided attention, where the listener must attend to several or all the sound sources in the environment.

3.0 BINAURAL TECHNOLOGY

The input signals to the hearing system are the sound pressure waves inside the left and right ear canals. Three-dimensional auditory environments or displays could thus be simulated through headphones if the directional transfer functions of the human head and external ears were known. The methods and techniques necessary to create virtual auditory environments are together referred to as binaural technology (e.g., Moller, 1992; Blauert, 1997).

3.1 General methodology

There are two main steps involved in a typical binaural technology application (Moller, 1992). The first is the derivation of the head-related transfer functions (HRTFs) of the listener from binaural measurements. The second is the creation of binaural signals by filtering the desired acoustic source input with the HRTFs, and the playback of these signals to the listener using headphones.

HRTFs measurements:

An important aspect to consider in the derivation of the HRTFs is the selection of a reference position for the binaural measurements. The reference position should allow the recording of all the spatial information available to the ear. Moller (1992) investigated three possible positions: at the eardrum, at the entrance to the open ear canal, and at the entrance to the blocked ear canal. The entrance to the blocked ear canal offered several advantages (Moller, 1992; Moller et al., 1995b). Firstly, the blocked ear canal method is easier to implement because it is less prone to microphone fitting and stability problems, measurement noise, sound field interference and other artifacts. Secondly, recordings at the blocked ear canal entrance are free from individual subject differences in ear canal transmission that are not related to the spatial characteristics of the ear as such. HRTFs derived from the blocked ear canal method possess less interindividual variation than other reference positions. This was demonstrated theoretically (Moller et al., 1996) and verified experimentally (Moller et al., 1995b). In the latter study, blocked ear canal HRTFs measured in 40 human subjects showed a clear common structure with small interindividual variations up to about 8 kHz.

Headphone equalization:

In a practical application, the derivation of HRTFs is not an end in itself. Binaural signals must be created by convolution or filtering with the HRTFs and they must be played back to the listener. However, the electroacoustic transfer function of the headphones contributes to the total sound transmission to the ear, and thus require equalization for the correct playback of binaural signals.

Moller (1992) examined the correction functions required in the headphone equalization step. The first correction compensates for the electroacoustical pressure transfer function of the headphone (or PTF in the terminology of Moller et al., 1995a) from the electrical input terminals of the heaphone to the sound pressure at the reference position. There is no other correction needed when a location in the open ear canal is chosen for the reference position. When the blocked ear canal is used as the reference position, the equalization step also requires an extra correction to account for the different acoustical source impedance loading of the ear when listening through headphones instead in the free-field. This correction term is referred to as the pressure division ratio (PDR) (Moller et al., 1995a). It reduces to unity if the radiation impedance looking outwards from the ear canal entrance is unchanged by fitting of the headphone, or if this impedance is much smaller than the input impedance looking inwards from the ear canal entrance (Moller, 1992). In this case, the headphone is said to provide a free-air equivalent coupling (FEC) to the ear.

Moller et al. (1995a) measured the PTF and PDR functions of 14 commercial headphones at the blocked ear entrance of 40 human subjects. The PTF functions were found to be far from flat for all headphones and, in general, none of the headphones tested was deemed suitable for the playback of binaural signals without proper equalization. All PTF functions also showed considerable interindividual variations, especially above 8 kHz. A blocked ear canal reference position led to smaller interindividual variations than a position in the open ear canal, and was easier to implement from a methodological standpoint. The PDR functions were found to be much smaller than the PTFs for all headphones. In general, headphone constructions closely mounted to the ear canal had larger PDF functions than those mounted further away. Up to 2 kHz, all headphones gave flat PDR functions close to 0 dB. Above 2 kHz, the PDRs showed fluctuations with some degree of interindividual variations but with a common structure for each headphone.

Individualized versus generic binaural signals:

Perfect reproduction of the binaural signals can only be guaranteed only if binaural measurements and headphone equalization steps are realized with the target listener's own ears. Thus, the question of the possible errors introduced by using another subject, or artificial head, in either or both steps of the binaural technique must be considered. Moller et al. (1996) conducted an error analysis of the reproduced binaural signals, and compared the sensitivity of different reference positions to the use of non-individualized binaural measurements and/or headphone equalization. For the smallest possible error, they proposed: (1) a blocked ear canal reference position for the binaural measurements, (2) the use of an FEC headphone, and (3) the use of individualized PTF headphone equalization whether the binaural measurements were individualized or not.

3.2 Psychoacoustical evaluation

Wightman and Kistler (1989) studied the localization performance of 8 subjects over a set of 72 source directions for broadband white noise bursts presented either by loudspeakers in the free field or by headphones. The headphone stimuli were derived from individualized HRTFs and were individually equalized using a reference position at the eardrum of the open ear canal. Overall, the localization performance with headphone stimuli was nearly identical to that of the reference condition in the free field for each subject. The only noticeable differences that emerged were a greater percentage of front/back confusions (almost double) and a slightly poorer perception of elevation in the headphone condition than in the free field.

Moller et al. (1996) studied the localization performance of 8 subjects over 19 source directions and distances for speech stimuli presented either by loudspeakers in a listening room with reverberation time of 0.4 s or by headphones. The headphone conditions were meant to reproduce binaural recordings made in the same room with the same loudspeaker arrangement. These recordings were made at the blocked ear canal of several subjects. Subjects listened to their own recordings (i.e., individualized), or to those of another subject or a mixture of subjects (i.e., generic). The headphones were always individually equalized to the target listener for the localization experiment. The results for the loudspeaker condition and the headphone condition with individualized recordings were not significantly different. However, the headphone condition with generic recordings led to a significantly greater percentage of errors for sources in the median plane (approximately double), including front/back confusions, and a slight increase in the number of distance errors. Out-of-cone errors were very rare in all conditions tested. None of the subjects reported in-the-head perception of localization in any of the headphone conditions, regardless of whether or not the recordings had been individualized.

In the headphone experiments above, the binaural signals were not synchronized with the head movements of the subjects. Subjects were instructed to keep their heads fixed. Head movements may reduce localization ambiguities, especially front/back confusions and within cone-of-confusion errors (Wallach, 1940; Wightman and Kistler, 1999), and may facilitate the perception of externalization (Durlach et al., 1992). Head movements can be taken into account in a binaural technology application by using a head-tracking device to update the location of the source(s) with respect to the listener's head coordinate system (Blauert, 1997). The externalization of signals, and distance perception, may also be greatly facilitated by reflections and reverberant energy in the listening room (Durlach et al., 1992). The simulation of room acoustics for binaural technology applications requires sound-field modelling (Blauert, 1997).

4.0 Integrating Active Noise Reduction and Binaural Technologies

4.1 General concept

Communication headsets with sound attenuation capabilities are often used in situations where an individual must be in contact with others at a remote location while operating in a noisy environment. The most common design is based on a passive circumaural hearing protection device fitted with earphones inside the earcups and a boom microphone in front of the mouth. ANR communication headsets can provide a significantly higher amount of low-frequency attenuation compared to passive headsets. The potential benefits of this additional attenuation are a reduced noise exposure for the user, and improvements in speech intelligibility and signal detection through the communication channels. If the speech signals are spatialized and separated from the environmental noise using binaural technology (Section 3), further improvements in intelligibility and signal detection can be expected (Section 2). In addition, virtual auditory displays and 3D models of the listening space can be created through binaural technology, which can greatly facilitate the

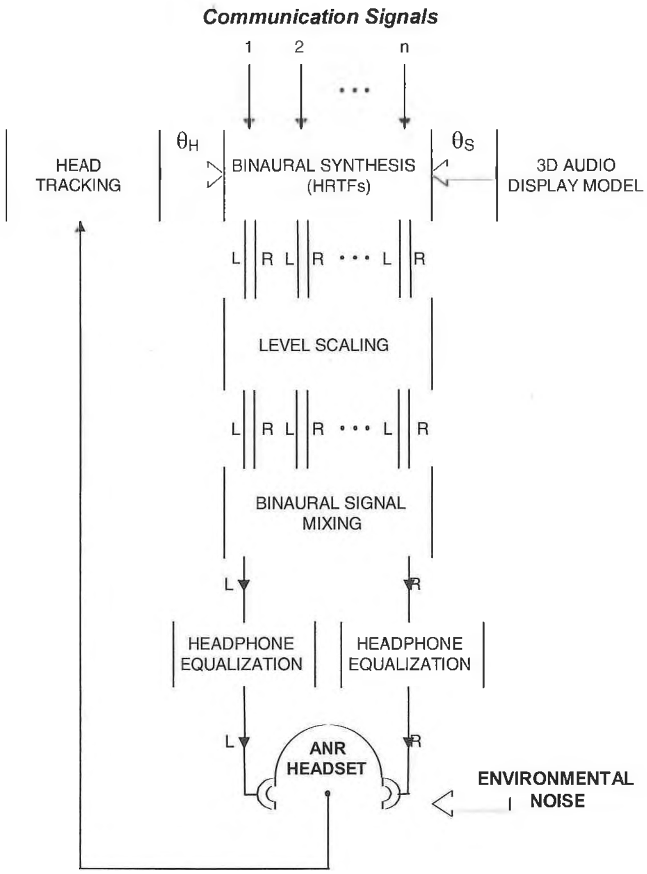

Figure 2: Sketch of an ANR-binaural communication headset

monitoring and interpretation of the various sources of information presented to the listener (Begault, 1993; McKinley et al., 1994; Bronkhorst et al., 1996).

Figure 2 illustrates the main components in a complete system integrating binaural technology to ANR communication headsets. A typical use of such a system would be inside a noisy aircraft cockpit. The different communication signals are individually spatialized on the basis of the HRTFs of the target listener, the desired spatial position θ_S of each signal source, and the current head position θ_H. The coordinates θ_S express the virtual display model to be created for the particular listening task. The right and left ear signals from each spatialized source are then scaled in level to compensate for head movement effects, as appropriate. The signals are then mixed to sum all left and right components. The resulting two signals are equalized for headphone sound transmission, and fed into the left and right communication channels of the ANR headset. The ANR headset, itself, reduces the external noise from the environment with little or no effect on the transmission of the communication signals.

There are several aspects to consider when combining ANR and binaural technologies as in Figure 2. Firstly, there is the hearing protection performance of the ANR device for the given environmental noise conditions. Secondly, there are the electroacoustical characteristics of the communication channels of the device that would be necessary for an adequate reproduction of binaural signals. Thirdly, there are the requirements of the application itself, which determine the complexity of the binaural simulation, and the design of a suitable virtual auditory display.

4.2 ANR devices for hearing protection and speech transmission

ANR technology provides a means of increasing the low-frequency attenuation in communication headsets or hearing protectors for use in high-level environmental noise (Casali and Berger, 1996). A miniature microphone housed within the earcup samples the incoming waveform. An inverted copy is created and added to the original for the purpose of cancellation. Components of the two waveforms that are out-of-phase will cancel, thereby reducing the overall sound pressure inside the earcup. ANR systems mounted on earmuffs are currently limited to frequencies below 0.5-1 kHz, where they add to the passive attenuation provided by the earcup (McKinley et al., 1996). Attenuation at higher frequencies is achieved by passive means only. Maximum active low-frequency attenuation in the order of 10-20 dB has been measured around 0.125-0.25 kHz over the passive mode (McKinley et al., 1996; Abel and Spencer, 1997; Abel and Giguère, 1997).

The additional low-frequency noise reduction achieved by ANR headsets over passive devices points to improvements in auditory perception for signals transmitted through the communication channels. Objective predictions based on the Articulation Index (Nixon et al., 1992) and the Speech Transmission Index (Steeneken and Verhave, 1996) procedures have demonstrated the speech intelligibility gains that can be realized. However, this has not always been achieved in practice (Gower and Casali, 1994). The frequency response of the communication channels and the effects of the ANR circuitry on the speech transmission quality are important determinants of intelligibility (Steeneken and Verhave, 1996). Several studies have also shown that ANR devices fail to operate when noise levels saturate the ANR circuitry, typically in the range of 120-135 dBA (Brammer et al., 1994). Other characteristics of the device affecting performance are the presence of transients or shut down periods after overload, and the comfort during use (Crabtree, 1996; Steeneken and Verhave, 1996).

4.3 ANR devices for binaural technology

Previous work:

ANR headsets have not been used extensively in binaural technology applications. Ericson and McKinley (1997) from the Armstrong Laboratory at the Wright-Patterson AFB (Ohio) reported a virtual audio presentation of speech communication signals over a Bose AH-1A headset, an ANR headset, configured for binaural operation. The HRTFs from the KEMAR manikin were measured at a 1° spacing in azimuth angle and used to simulate the virtual audio sources. The elevation angle of the sources was maintained fixed in the horizontal plane and distance cues were essentially absent. A head-tracking system measured the orientation of the listener's head and was used to maintain the virtual sources fixed in space.

Currently, the research group at the Armstrong Laboratory is utilizing the blocked ear canal method to derive the HRTFs and headphone equalization functions with the microphone inserted about 2-3 mm inside the canal entrance (McKinley, 1997). The critical factor is the repeatability of the microphone/plug location during the measurements. However, once a consistent fitting of the headset and microphone/plug assembly can be ascertained, they find no particular problems in equalizing their ANR headsets. To facilitate the equalization process, they choose headsets with matched left/right earphone drivers, typically within 2 dB in sensitivity. Tracking and integrating the head movements of the listener into the binaural simulation is found important for the perception of externalization of signals, and to maintain the highest possible speech intelligibility (McKinley, 1997).

Design criteria:

Commercial ANR headsets have not been specifically designed for binaural technology applications. The following criteria are proposed for the selection or design of a suitable ANR headset for experimentation with virtual audio signals. A minimal set of recommended electroacoustic specifications is given.

Listening mode — The ANR headset must support stereo communication signals for dichotic listening as a pre-condition to a binaural technology application.

Volume control — Headsets equipped with a single knob to control the signal volume under both earcups are preferred over headsets equipped with dual controls to independently adjust the volume in the left and right ears. Independent volume control interferes with the correct reproduction of ILDs from the virtual audio sources.

Cross-talk attenuation — The amount of cross-talk attenuation between the left and right communication channels should exceed the maximum possible ILD in the free field plus a safety margin of about 10 dB. The maximum ILD arises for lateral incidence, and is typically 20-25 dB around 5 kHz and in excess of 30-35 dB above 8 kHz (Moller et al., 1995a; Wightman and Kistler, 1997).

Earphone linearity — The earphone frequency response from the electrical input of the communication channel to the sound pressure output in the earcup must be linear over the full dynamic range of signal levels, and must be insensitive to environmental noise conditions. Two conditions are to be met: (1) the sound pressure output of the signal must grow linearly with the electrical input signal under constant environmental noise presented at different levels, and (2) the sound pressure output of the signal must remain constant for a constant electrical input signal under variable environmental noise levels.

Earphone frequency response — The earphone frequency response must be as uniform and smooth as possible to simplify the headphone equalization procedure. The upper frequency limit depends on the location of the virtual audio sources in the intended application. In the case of virtual sources positioned in the horizontal plane and frontally, an upper frequency limit of 4-5 kHz may be adequate. In the case of sources distributed both in front and at the back, in the median plane, or elevated from the horizontal plane, an upper frequency limit of 8 kHz or more may be required.

Coupling to the ear — The headset selected should provide a free-air equivalent coupling (FEC) to the ear, as defined in Moller et al. (1995a), to simplify the headphone equalization process when the blocked ear canal method is used. This could be verified by measuring the pressure difference ratio (PDR) of the headsets (Moller et al., 1995a), or by measuring the radiation impedance looking outwards from the ear canal using an artificial head with and without headset fitted (Schroeter and Poesselt, 1986).

Interaural earphone matching — The interaural amplitude and phase matching in the earphone frequency response is obtained by equalizing independently the left and right sides at the headset. In cases where a single generalized equalization function is required, the left and right earphone transmission should be closely matched in frequency response and sensitivity.

Device selection:

The characteristics of nine commercial ANR headsets were assessed against the proposed design criteria (Abel and Giguère, 1997). The devices surveyed are listed in Table I. The manufacturers differ greatly in their methods of presenting the communication signals through the devices. For example, the Bose Aviation approach (Gauger, 1995) is based on the conventional feedback servosystem where the output, the sound pressure wave inside the earcup, is tracking a desired input, the electrical communication signal, while minimizing interfering noise. Thus, the effect of the ANR feedback loop on the communication signal must be compensated for by an equalization filter to flatten the transmission response. The communication signal of the Telex ANR Headset System is injected electronically just before the earphone transducer, but is subtracted from the sensing microphone output. The communication signal is in effect removed from the ANR feedback loop and its transmission becomes essentially insensitive to the operation of the ANR circuitry. A similar approach is used in the Sennheiser NoiseGard (Crabtree, 1997). The David Clark H1013X uses two earphone transducers, one for the communication signal and one for the ANR cancellation procedure (Crabtree, 1997).

Table I: Analog ANR communication headsets surveyed

Peltor ANR Aviation Headset
Sennheiser NoiseGard
Bose Aviation Headset
Bose Aviation Series II
David Clark DCNC Headset
David Clark H1013X
Telex ANR Headset System
Telex ANR 4000
TechnoFirst NoiseMaster

The survey showed that the most likely candidates for binaural technology applications are the Peltor, Sennheiser and TechnoFirst devices (see Abel and Giguère (1997) for additional technical details). They all support stereophonic listening, have a single control knob for volume control in both earcups, and provide good sound attenuation properties. Unfortunately, the manufacturers' specifications do not provide sufficient information to assess adequately all the technical characteristics necessary for binaural technology on any device. In particular, the amount of cross-talk attenuation, interaural earphone matching and type of coupling to the ear are essentially unspecified. In practice, earmuff-type ANR devices are likely to behave as FEC or near-FEC headphones because of their relatively large earcup volumes that are necessary for good low-frequency passive attenuation. Indeed, Schroeter and Poesselt (1986) found that the radiation impedance looking outwards from the ear canal is essentially unchanged above 0.4 kHz by fitting an earmuff-type hearing protector. Below 0.4 kHz, these hearing protectors do affect the radiation impedance of the ear, but then, this impedance is much smaller than the impedance looking into the ear canal. Under these conditions (Moller, 1992), earmuff-type ANR headsets could be considered FEC.

The commercial devices surveyed in Table I were all based on analog ANR technology. Prototype ANR devices based on digital technology have been tested in research laboratories in the past few years (Pan et al., 1995), and the first commercial digital ANR headsets have been recently introduced (e.g. Telex ANR-1D). Since binaural technology applications are also based on digital signal processing, digital ANR headsets could lead to more completely integrated and compact ANR-binaural systems than analog ANR headsets would allow. A particularly attractive digital ANR design for use with binaural technology is based on adaptive feedforward noise control. The feedforward control structure does not perturb the communication signals, and so offers the potential for higher fidelity reproduction than the commonly used feedback control structure (Brammer and Pan, 1998)

4.4 Aircraft cockpit application

The complexity of the binaural simulation depends on the requirements of the application at hand. In an aircraft cockpit application, very different listening situations could arise. Two extreme scenarios are detailed below.

<u>**Simple selective attention task:**</u>

In this task, the pilot must focus on the speech of one and only one speaker through the communication channel of the headset, in the presence of the environmental noise in the cockpit. Using binaural technology, the speech communication signal could be externalized and positioned in space to provide an angular separation with the environmental noise. The goal would be either to (1) maximize the speech intelligibility score for a given signal level, or (2) minimize the signal level for a given speech intelligibility score. For listeners with normal hearing or symmetrical hearing losses, a gain at the 50% speech intelligibility level up to about 4-6 dB with respect to diotic listening can be expected under the most favourable noise conditions (Section 2). Under conditions of reverberation, band-limited noise, or multiple noise sources, the speech intelligibility gain will be smaller. For listeners with asymmetrical hearing losses, the speech intelligibility gain due to ITDs is typically half that of normal-hearing listeners. Nonetheless, given the very steep slope of the intelligibility function near the 50% level, typically 15% per dB for sentence material, a gain of only a few decibels could give rise to substantial intelligibility improvements for all classes of listeners, but only in communication systems with low signal-to-noise ratios. It is also under conditions of low SNRs that the greatest benefits of ANR over passive communication headsets are anticipated for speech intelligibility.

In the simple listening application above, there is no localization task involved per se. Thus, the design of the binaural system could be simplified by the use of generic HRTFs (Section 3), particularly if a direction of incidence in the horizontal plane is selected for the virtual speech signals. Likewise, the benefits of synchronizing the binaural signal simulation with the head movements of the user may be minimal in this application, so the binaural signal procesing could be further simplified.

The selection of an optimal direction of incidence for the spatialized communication signal will depend on the characteristics of the environmental noise sound field at the location of the pilot's head. Cockpit noise spectra and levels are dependent on the type of aircraft, and the speed and altitude of the aircraft, among other factors (Rood, 1988). To achieve the maximal binaural speech intelligibility gain, the speech signal should be spatially separated from the noise by about 45° or more. However, several difficulties can arise because there are in general more than one source of noise in an aircraft, and the noise field in a typical cockpit is not free field. Another problem is that earmuff-type devices can severely disrupt the localization cues from external sounds (Abel and Hay, 1996). In practice, the sources of noise are large and distributed in typical aircrafts, and there is minimal or no acoustical treatment in the cockpit. Under these conditions, the environmental noise at the pilot's head could be classified as diffuse or quasi-diffuse, and thus the selection of a speech signal incidence would not be too critical.

An important related aspect to consider is the scaling of the binaural signal level at the ears. In a system where the HRTFs are synchronized with the head movements of the user, the sound exposure arising from the communication signal will vary with the selected direction of incidence. Moreover, if at any time the speech incidence lies outside the median plane, exposure from the speech signal will be asymmetric across the two ears, typically larger on the ipsilateral ear than the contralateral ear. A possible solution to maintain a constant and symmetric exposure is to scale the HRTFs with a direction-dependent gain so that the total speech energy becomes independent of sound incidence and equal in each ear. Another possibility, suggested by Bronkhorst and Plomp's (1988) experiments, is to scale the amplitude spectrum of each HRTF to a common reference amplitude spectrum, such as that corresponding to a frontal incidence, while keeping the phase spectrum intact. These solutions are based on the observation that it is the ITDs alone that provide a true binaural benefit for speech intelligibility (Section 2), and that accurate localization of the speech signal is secondary in this task.

Complex divided attention task:

At the other extreme, in a complex divided attention task, the pilot must attend to several speakers (e.g., co-pilot, pilots in other aircrafts, ground crew, etc.) through the communication channel of the headset, in the presence of cockpit noise. The pilot may also need to be alert to various visual targets in his/her environment that are cued to characteristic warning sounds. In this case, the speech signals from the different speakers and the other sounds to attend to would be externalized using binaural technology and positioned in space on the basis of ergonomic considerations. The goal would be to provide the user with a model of his/her complex acoustic environment in order to facilitate the interpretation of the various sources of information (Figure 2). The actual display design would depend on the specific demands placed on the user (Mack et al., 1998).

Localization errors, particularly elevation errors and front/back confusions, would be very detrimental in this application, because of the need to maintain a consistent spatial model of the environment. To maximize localization performance, this application would likely require individualized HRTFs and headphone equalization. It would also be highly desirable for ergonomic considerations and for optimizing accuracy in sound localization to track the head movements of the user and update the binaural simulation synchronously, so that the acoustic sources of information remained fixed in space. Because of the localization needs, both the ITD and ILD binaural cues are important in this task. This prevents manipulations of the amplitude spectrum of the HRTFs, other than applying a direction-dependent gain to each pair of left/right HRTFs.

5.0 Conclusions

This article reviewed the fundamental research and several practical aspects relevant to the integration of ANR and binaural technologies in the design of improved communication headsets, with particular attention to an aircraft cockpit application. ANR technology can reduces the interfering noise from the environment. Binaural technology allows the creation of 3D auditory displays to transfer coincident messages to different spatial positions.

In a simple selective attention task, the requirements of the binaural simulation are not very stringent as far as localization performance and the tracking of head movements are concerned, but careful consideration must be given to the direction of incidence of the environmental noise and to the scaling of the binaural signal levels. Under the most favorable conditions, a speech intelligibility improvement equivalent to a gain of 4-6 dB in SNR can be expected for this task with a ANR-binaural headset system over a system with ANR capabilities alone.

In a complex divided attention task, the binaural simulation system must provide for accurate sound localization performance, but the scaling of the binaural signals is less critical. The greatest benefits of 3D virtual auditory displays may well be found for this type of task, when there are more than two speakers or signals to attend to simultaneously (Ericson and McKinley, 1997). However, more fundamental research is needed to quantify the real advantage gained in terms of improved speech intelligibility, total information transfer, increased situational awareness or reduced workload fatigue (Begault, 1993; McKinley et al., 1994; Bronkhorst et al., 1996)

Commercial analog ANR communication headsets are not designed for binaural applications and would require extensive testing before making firm recommendations on specific devices. A list of features relevant to binaural technology includes the listening mode and volume control options, the cross-talk attenuation across the two channels, the earphone linearity, the earphone frequency response and degree of interaural matching, and the type of coupling to the ear. Newly developped digital ANR headsets may faciltate the integration with binaural technology.

6.0 Acknowledgements

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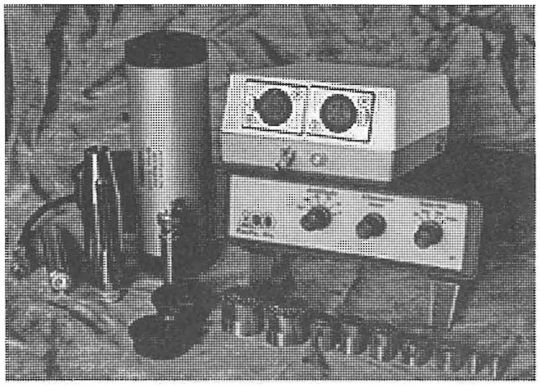

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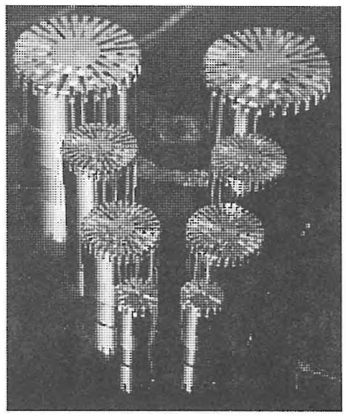

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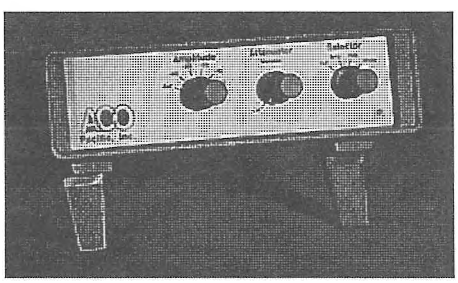

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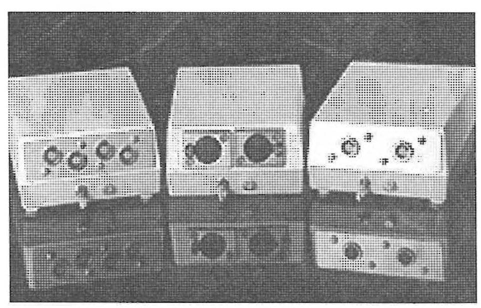

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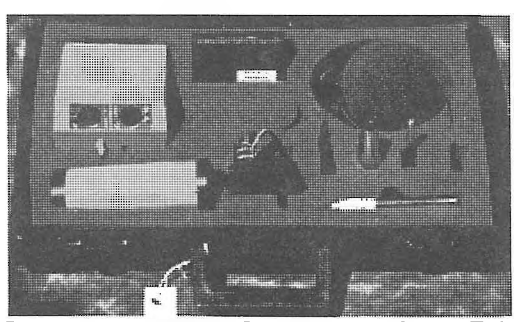

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Canadian Acoustical Association
Minutes of the Board of Directors Meeting
13 May 2000
Toronto, Ontario

Present: J. Bradley, T. Nightingale, D. Streduiinsky, D. Giusti, R. Ramakrishnan, D. Jamieson, K. Pichora-Fuller, D. Whicker, T. Kelsall, K. Fraser, M. Cheeseman, A. Berry.

Regrets: D. DeGagne, J. Hemingway, N. Atalla.

Guests: A. Berry

Meeting called to order at 10:10 p.m.

Minutes of the 18 October Board of Director's meeting were approved as written in the December 1999 issue of Canadian Acoustics. (Moved by R. Ramakrishnan, seconded by D. Whicker, carried).

President's Report

J. Bradley reported that various members of the Board and Executive were working on initiatives to promote the society (e.g., formulation of a new website, new accounting procedures, by law revisions, etc.) and that these would be discussed as separate agenda issues. (Acceptance of the President's report was moved by R. Ramakrishnan, seconded by K. Pichora-Fuller, carried).

Secretary's Report

T. Nightingale was very pleased to report that the membership is reasonably constant when compared to the last three years. The total paid membership (including non-voting journal subscriptions) is 346 as of 08 May 2000. The same time one year earlier the number was 341. Two years earlier it was 346. The Secretary provided a summary of the key activities which included formulation of an integrated database containing both Association members and contacts at Canadian universities to whom prize announcements would be sent, a database of past exhibitors at conferences, and a special solicitation to ASA members who live in Canada but who are not members of CAA.

The Secretary tabled a letter from Hugh Jones, (a past president of the Association) proposing the creation of a new membership category that would recognize persons who are retired but have been long time members. The Board was in favour of the idea but asked the Secretary to refine the conditions and circulate a new definition for their consideration before the next meeting in September. If ratified, it would be brought before the membership at the AGM in Sherbrooke.

In late April, the CAA received a letter announcing that the organization will be gifted with $10,000. The money comes from the Signal Processing Institute Fund that was created to help off-set the cost of students travelling to attend NATO Advanced Study Institutes on Underwater Acoustics and Signal Processing. The only restriction is that the gift be used to assist students in attending technical conferences. The Secretary has responded to Dr. Chan and has requested that the cheque be sent directly to the Treasurer in care of Jade Acoustics. A letter of thanks will be sent to Dr. Chan on behalf of the CAA.

Secretarial operating costs for first ten months of this fiscal year was $1067 and are very comparable to those of last year. There were some notable costs that traditionally have not been incurred in past. They were the integration of the prize mailing list into the membership database, creation of an exhibitors database, and the special mailing to the ASA members. (Acceptance of the Secretary's report was moved by D. Jamieson, seconded by R. Ramakrishnan, carried).

Treasurer's Report

The Treasurer provided an itemized report of the Association's finances for the last two years which can be used as the basis for formulating a budget. In both fiscal years revenues exceeded operating costs. This was due partly to the significant surplus generated by the conferences and that several prizes were not awarded (i.e., no applicants). The Treasurer reported that the Operating Account had more money than was required to cover the normal expenditures of the Association. It was recommended that $15k from this account be invested in high-yield term deposits. The Board concurred.

The Treasurer reported that it would be quite easy for the Association to obtain a Visa or MasterCard merchant account. After considerable discussion of the benefits that would be afforded to our members (e.g., easier payment of dues, and conference registration, etc.), D. Whicker moved that, "The Treasurer obtain a Visa merchant account for the association provided that the annual fee is less than $150 and the monthly processing cost is less than $50." The motion was seconded by M. Cheesman and was carried.

The Treasurer reported that CAA had received an invoice for the Association's membership in International Institute of

Noise Control Engineers (I-INCE). D. Whicker moved that, "The Treasurer pay the 450 Euro invoice for year 2000 membership." The motion was seconded by K. Fraser and carried.

It was requested that the Treasurer prepare a detailed budget for FY00/01 (identifying all significant expenditures e.g., journal printing, prizes, I-INCE membership, website, Secretarial operating costs, etc.) and present this to the Board at the next meeting for their consideration and ultimate approval.

Membership Chair's Report

Improvement of the CAA Website (located on the server of the University of Western Ontario at WWW://uwo.ca/hhcru/caa/) continued throughout the year. There was much discussion of how the organization could best attract and retain new members. All persons present agreed that the most effective method is word-of-mouth to colleagues. In the discussion it was recognized and agreed that CAA must be promoted as being a complement to larger international organizations such as ASA, INCE, etc. Therefore, we should begin to identify other smaller Canadian associations that we can partner with. An example would be to schedule our conference immediately before or after theirs, or if they do not have a journal, consider publishing their papers, new letters, articles, etc. in Canadian Acoustics. (The former is being considered for the 2001 Toronto conference.) Thus, the Membership Chair would become the liaison between the CAA and other acoustical organizations.

Editor's Report

R. Ramakrishnan reported that he has been successful at attracting research papers and technical articles. Currently, there is sufficient material for the June and December issues of Canadian Acoustics (CA). Printing costs remain on budget. K. Fraser agreed to take over coordinating the judging of the student presentations from R. Ramakrishnan. D. Whicker reported that his company had a listing of the titles and authors of all the research articles published in CA since its inception, and that he would make this available for publishing in CA and posting at the website.

It was also suggested that one of the conditions of the Shaw Prize is that the recipient publishes their annual progress report in the journal. (Acceptance of the Editor's report was moved by T. Kelsall, seconded by K. Pichora-Fuller, carried).

Past and Future Conferences

1999 Victoria: S. Dosso, Conference Chair, in his very detailed final report indicated that the conference was a financial success. A surplus of approximately $5400 was received by the Treasurer. The Board expressed their thanks to Stan, the Victoria organizing committee, including Susan Dunlop of the University of Victoria. (Acceptance of the Convener's report was moved by M. Cheesman, seconded by T. Kelsall, carried).

2000 Sherbrooke: A. Berry, on behalf of N. Atalla, reported that the conference will be two days (Thursday and Friday, 28, 29 September). There will be several special sessions with the emphasis being on physical acoustics and structure borne sound. The conference website is http:www.caa2000.gme.usherb.ca/

2001 Toronto: D. Giusti, reported that the organizing committee had not chosen the dates or a hotel and that they would be meeting soon to draft a request for proposals that would be submitted to perspective venues.

2002 Charlottetown: A. Cohen has agreed to organize the conference this year.

Award Coordinator's Report

K. Pichora-Fuller provided a report summarizing the Awards activities this year. The following is a summary by prize:

Edgar and Millicent Shaw Postdoctoral Prize in Acoustics: No applications have been received. S. Abel is the prize coordinator.

Alexander Graham Bell Graduate Student Prize in Speech Communication and Behavioural Acoustics: One or more applications have been received. D. Jamieson is the prize coordinator.

Fessendon Student Prize in Underwater Acoustics: One or more applications have been received. D. Chapman is the prize coordinator.

Eckel Student Prize in Noise Control: One or more applications have been received. M. Hodgson is the prize coordinator.

CAA Canada-Wide Science Fair Award (Youth Science Foundation): Meg Cheesman will be a judge as the fair will be held in London this year. A. Cohen is the prize co-ordinator.

Raymond Hetu Memorial Undergraduate Award: Book prize: Approximately $2200 dollars has been donated by the membership to honour the memory of Dr. Hetu. A description of this prize has not been forwarded to the Board by the Committee. K. Pichora-Fuller will be joining the Committee which consists of M. Hodgson, C. Laroche, and B. Gosselin.

Directors' Awards: For best papers in Canadian Acoustics (December through September issues). K. Pichor-Fuller is the prize co-ordinator.

New Initiatives

a). Possible by-law changes
J. Bradley tabled revised by-laws that would make it easier for quorum to be obtained at Board meetings. In the revised by-laws three members of the Executive, the President, Treasurer, and Executive Secretary, will also be members of the Board and can therefore exercise a vote. In the revisions, a provision was made to allow for student memberships at a reduced rate. D. Whicker moved that, "The revised by-laws be brought before the membership at the next Annual General Meeting for their consideration and vote." The motion was seconded by M. Cheesman and carried.

b). CAA Website
D. Whicker reported on efforts to create a new website and register a dot-ca domain name. The board has chosen "CAA-ACA.CA" as the preferred domain name. T. Nightingale will help with filing the application. D. Whicker and B. MacKinnon have very kindly offered to design and build the website. The Board thanked Doug and Barry and agreed to recognize their efforts by placing a credit on the website. D. Whicker agreed to act as the first English webmaster but informed the Board that it is necessary to find a French webmaster. The webmasters would be responsible for soliciting and mounting the material at the site in their respective languages. Translation will be done by a series of volunteers. A. Berry agreed to ask his colleagues at the University of Sherbrooke if one of them might be interested in being the first French Webmaster. It was estimated that the site might have 25-45 pages in each language and the cost from the service provider would be approximately $55/month. D. Whicker reported that approximately $500 would be required to cover the cost of registering the domain name and paying the service provider for the first six months. M. Cheesman moved that, "Five hundred dollars be allocated by the Treasurer from the Operations Account to cover expenses associated with registering the domain name and operating the site until the next Board meeting." The motion was seconded by K. Fraser and was carried.

Other Business

CAA has been asked to forward the names of experts that would be interested to serve as a technical advisor on one or more of the five new I-INCE technical initiatives. They are:
· Noise of Recreational Activities in Outdoor Areas
· Noise Labels for Consumer and Industrial Products
· Assessing the Effectiveness of National and International Noise Policies and Regulations
· Noise and Reverberation Control for School Rooms
The Board requested that interested parties contact the Secretary who can provide more information regarding the scope of the initiatives and forward names to I-INCE when he attends the I-INCE General Assembly in August.

Adjournment

D. Giusti moved to adjourn the meeting, seconded by D. Jamieson, carried. Meeting adjourned at 4:10 p.m.

Special Action Items Arising from the Meeting

J. Bradley
· Finalise the revisions to the by-laws and present them to the Board in September.

T. Nightingale
· Draft letter of thanks to Dr. Chan of the Signal Processing Institute Fund.
· Revise the proposal of Dr. Jones to create a new membership category "member emeritus" and present to the Board at the next meeting.

D. Whicker
· Find a suitable service provider and, in conjunction with B. MacKinnon, design and build the English version of the CAA website.
· With T. Nightingale, complete and file application for "CAA-ACA.CA" domain name

D. Giusti
· Prepare a business plan for FY00/01 for the fall Board meeting.
· Open a merchant's account to allow the Association to collect VISA payments.

"The Loudspeaker Design Cookbook; 5th Edition", Vance Dickason. Published by Audio Amateur Press, 1997 Peterborough New Hampshire, 03458. ISBN 1-882580-10-9, Price $34.95, USD

The Loudspeaker Design Cookbook, in its various stages of release, has been in existence for over 20 years. I first came across it in the late 1980's when it was in its 3rd edition. The 4th edition, which was released in 1991, added significant work and found its way into the library of many loudspeaker hobbyists. The latest edition, released in 1997 does not disappoint. The book is divided into 12 chapters, which describe the diverse field of loudspeaker design. A complete book can easily be written on each of the chapters in the 'Cookbook'; but that is not the intent of this publication. It is meant to be a 'hands on' reference for the hobbyist or serious speaker designer.

The first chapter, Chapter 0, presents the fundamentals of how loudspeakers actually work. It remains virtually unchanged from the 4th Edition. Of particular note is the thorough description of the properties of electrodynamic speakers; especially the Xmax and Bl terms.

Chapter 1, "Closed Box Low Frequency Systems", is self-explanatory. There are sufficient graphs and tables and definitions of the various parameters of a closed box loudspeaker system to come up with a successful initial design. This chapter excels in describing the effects of stuffing materials on the low frequency performance of closed box or acoustic suspension loudspeaker systems. Chapter 1 also introduces bandpass loudspeakers and provides some discussion of potential pipe resonance and 'out of band' anomalies of a bandpass loudspeaker.

Chapter 2, entitled, "Vented Box Low Frequency Systems" describes 'bass reflex' type designs. This chapter goes into detail on the sensitivity of driver (woofer) selection to box design. The mechanical and electrical properties that are desirable in a woofer that is suitable for a vented box are identified. There are several detailed tables to select the desired vented box alignment and comprehensive design details are provided that will allow evaluation and design. Suggestions on how to use the tables are described. This chapter also provides details on vent dimension calculations (i.e., helmholtz resonator) in order to achieve optimal tuning. Guidelines are defined to avoid undesirable affects such as aerodynamic noise, pipe and vent resonances and other non-linearities. There is also a concise procedure for calculating box losses as well as a discussion on the optimal placement for damping materials. Active equalization of a vented box is also described as a means of increasing low frequency output. An example is provided in the 'Cookbook' where low frequency extension is realized but at the expense of a large box in comparison to a standard closed box or vented box alignment. One of the common themes in the book is that there is always a trade-off for box size, bandwidth efficiency and power handling.

Chapter 3, "Passive Radiator Low Frequency Systems," describes an enclosure concept where a diaphragm or passive radiator shares the load with the active driver. This system eliminates vent anomalies and provides a reasonable approach to low frequency augmentation without requiring a long vent. The passive radiator system essentially replaces the mass of air in the vent by the mass of the passive driver. A complete set of tables is provided. There is also an intriguing (for the subwoofer enthusiast) discussion on bandpass designs and augmented passive radiators.

Chapter 4 presents 'Transmission Line Low Frequency Systems'. Such systems have always been a source of debate in audiophile magazines. This type of system is basically an open-ended pipe with a quarter wavelength pressure maximum at the exit. In the case of the transmission line, this pipe is lined internally to achieve desirable bandwidth and minimize any resonant effects. A transmission line is very much a trial and error type of system; but it is this aspect of speaker building that is most fascinating to the hobbyist. The transmission line system just happens to be one of the trickier recipes in this cookbook that can yield wonderful results if prepared correctly.

Chapter 5 elaborates on cabinet construction as well as cabinet shape, and it references Harry Olsen's work from the 1950s. Driver mounting, edge defraction and discontinuity effects in terms of degrading performance are discussed. In general, it is accepted that these effects are measurable but questionably audible. There is an additional discussion on enclosure shape and the control of potential standing waves. Enclosure construction techniques are presented along with bracing and damping suggestions to minimize any coloration that the enclosure itself may impart.

Chapter 6 describes mid and high frequency drivers and enclosures. Mid frequency driver enclosures are common in three-way systems such that the midrange driver is physically separated from the low frequency driver. There is also a discussion on the enclosure versus no enclosure scenario. The latter has the benefit of being somewhat resonance free and providing a dipole radiation effect, which can manifest itself as being spacious. The chapter ends with a series of useful guidelines for realizing the maximum performance from an enclosure.

Chapter 7 presents 'Passive and Active Crossover Networks'. This section is crucial to loudspeaker performance and this particular art has advanced significantly in the past 10 years. The basics of crossover design are presented in this book. The difference between a passive and active crossover is that no power supply or amplification is needed for a passive design whereas it is for an active system. There are various orders of high pass and low pass configurations and this chapter explains each configuration clearly. There is also a lengthy discussion on the effects of driver separation and alignment of approximate acoustic centres. The book also outlines how current software for passive crossover design can take into account the natural rolloff of the various loudspeakers in the desired response of a given crossover. This book treats the passive crossover options thoroughly and there seems to be some intimation that the fourth order Linkwitz Riley is preferred amongst hobbyists. This chapter also evaluates the pros and cons of several two-way crossover design alignments and three-way designs. Tables are provided to help design these systems. The latter section of the chapter on active crossovers is brief but several useful references are listed. This book excels in providing good references for the reader interested in pursuing some of the various subjects in greater detail.

Chapter 8 is entitled, "Loudspeaker Testing" and this chapter provides the basics for obtaining the driver characteristics that are key in designing loudspeaker enclosures. These include impedance, resonant frequency, and T/S parameters (Thiele-Small). This chapter provides a discussion on instrumentation and procedures that are required to obtain these properties. Frequency response measurement techniques and equipment selection are suggested. These include ground plane, nearfield, half space and cabinet vibration measurements, as well as FFT and other time domain measurements such as waterfall or cumulative decay spectra, Wigner frequency distribution plots and energy time curves. The author clearly states that an explanation and interpretation of all these types of measurement systems are beyond the scope of his book but again, ample references are provided for the reader that is searching for additional insight. A significant addition to the 5^{th} edition is a section on voice coil temperature measurements. The effects of the temperature gradient in a voice coil on the dynamic performance of a loudspeaker system, cannot be ignored; hence, measurements are suggested to help account for this thermal behaviour and to compensate for it in the speaker recipe.

Chapter 9 discusses CAD software for loudspeaker design. The programs that are referenced are useful and some are exceptional; but it should be noted that most of these software packages have been available for more than 10 years. In this respect this chapter does not seem to be current. There may not have been much recent progression in speaker CAD software, but it may be worthwhile to consider software driven measurement systems in the next edition as there have been significant advances in this area over the past 5 years.

Chapter 10 is entitled, "Home Theatre Loudspeakers." This is a rapidly evolving area and the performance level of home theatre systems has increased considerably over the last 10 years especially with the prevalence of Dolby Digital. This chapter was prepared three years ago and it may already be out of date. The importance of maintaining speaker timbre in any surround system especially throughout the three front channels and in the surround channels are stressed. The chapter discusses an arrangement where specific height and aspect ratios of the loudspeaker position to the listener should be achieved. There are some helpful tips for everyone that wants to set up a home theatre system regardless of whether it is Dolby Pro Logic or Dolby Digital. There is also a very good discussion of the shielding requirements for a loudspeaker system. This chapter also defines the importance of the centre channel speaker and there is a compelling argument for preventing any timbre shift from the front to rear channels. There is also a fascinating discussion on the use of dipole speakers in home theatre applications and of course a very frank discussion on the 'ubiquitous' subwoofer.

Chapter 11 is entitled, "Car Audio Loudspeakers." This is a very unique chapter as it discusses at length the key design issues for realizing excellent performance in car audio for the design constraints are significant. The chapter also suggests noise control techniques that range from active noise cancellation to panel treatment. Other practical 'hands on' approaches are encouraged to reduce noise levels in an automobile.

The Cookbook ends off with an up to date advertiser listing for design software, speakers parts, test hardware equipment, access to other publications and audio electronics. It is a thorough list that will get the hobbyist excited about building his next loudspeaker systen. A conspicuous omission in the book is servo driven loudspeakers. Very briefly a servo speaker utilizes a feedback system to correct any non-linearities. What results is a well-behaved, efficient and powerful low frequency system. The 6^{th} Edition may want to consider this addition since such low frequency type systems are becoming readily available today on the hi-fi market.

The book succeeds wonderfully in creating enthusiasm about speaker building projects and hence becomes a true 'Cookbook.' The book encourages experimentation to achieve the desired personal tastes and if the loudspeaker does not meet expectations, then one simply goes back to the 'Cookbook' to start afresh.

Reviewed By:

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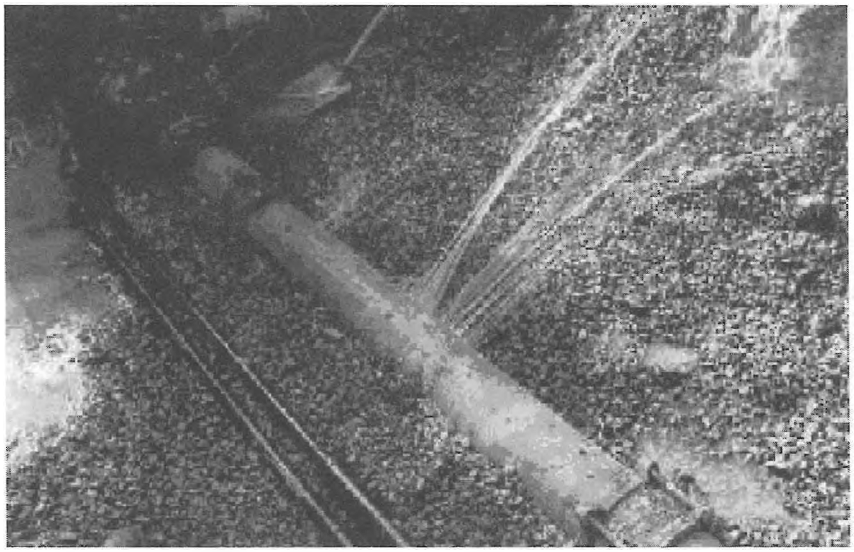
CANADIAN NEWS.... / NOUVELLES CANADIENNES....



Leak Detection Methods for Plastic Water Distribution Pipes
Méthodes de détection des fuites dans les conduites de distribution d'eau en plastique

Osama Hunaidi, Wing Chu, Alex Wang, and Wei Guan
Institute for Research in Construction/Institut de recherche en construction
National Research Council of Canada/Conseil national de recherches du Canada

The main objective of this research project was to investigate the effectiveness of commonly used acoustic leak detection equipment, in particular leak noise correlators, for locating leaks in PVC plastic pipes. Emphasis was placed on evaluating the methods on which the equipment is based, not on comparing different equipment makes.

The project was undertaken in response to the need for effective detection methods by water utilities in their efforts to control water leakage. Objectives of the research also included: Worldwide survey of leak-detection equipment; Characterization of leak sounds in plastic pipes; Identification of necessary improvements to existing equipment and procedures; Evaluation of the potential of alternative non-acoustic technologies from other industries

The research involved extensive field tests that were performed at a specially constructed leak-detection facility at the National Research Council (NRC) in Ottawa, Ontario.

More details of this research project can be found in the following web-site:

http://www.nrc.ca/irc/leak/leakdetect.html

Ce projet de recherche visait principalement à déterminer la capacité du matériel courant de détection acoustique des fuites, en particulier les corrélateurs de bruit, à repérer les fuites dans les tubes de PVC. Le but poursuivi était d'évaluer les principes de fonctionnement des appareils et non de comparer différentes marques.

C'est notamment pour venir en aide aux services publics d'eau, qui ont besoin de méthodes efficaces de détection des fuites pour réduire les pertes d'eau, que ce projet a été mis sur pied. Les responsables des travaux de recherche en question avaient aussi pour but: de mener une enquête mondiale concernant le matériel de détection des fuites; de caractériser les bruits de fuite dans les conduites en plastique; de déterminer les améliorations à apporter au matériel et aux techniques existants; d'évaluer les possibilités d'utilisation des autres techniques non acoustiques employées dans d'autres industries.

Dans le cadre de ces travaux de recherche, toute une série d'essais *in situ* a été réalisée sur le site de détection des fuites spécialement aménagé sur les terrains du Conseil national de recherches (CNRC), à Ottawa, en Ontario.

Pour obtenir plus de précisions concernant ces travaux de recherche, on peut se rendre sur ce site Web:

http://www.nrc.ca/irc/leak/leakdetect.html

Abstract of Master theses
Nicole E. Collison and Mark R. Fallat

Regularized Matched-mode Processing for Ocean Acoustic Source Localization

Nicole E. Collison
School of Earth and Ocean Sciences , University of Victoria
(Supervisor: Stan Dosso)

Localizing an acoustic source in the ocean is an important problem in underwater acoustics. Matched-field processing (MFP) can localize a source by matching acoustic pressure fields measured on an array of sensors with modelled replica fields computed for a grid of possible source locations. Matched-mode processing (MMP) consists of first decomposing the measured fields into the constituent modal components (a linear inverse problem), and then matching the resulting mode excitations with modelled replica excitations. However, standard modal decomposition methods used in MMP can have poor solutions when the inversion is ill-posed due to an inadequate sampling of the acoustic field. This thesis develops and tests a new approach to MMP based on a regularized modal decomposition which successively applies the replica excitations for each grid point as an a priori estimate. This method, referred to as regularized matched-mode processing (RMMP), is compared to standard MMP techniques in terms of resolution and solution variance. RMMP is further compared with both MMP and MFP in terms of the probability of correct localization for realistic synthetic test cases, including cases that involve environmental mismatch in the seabed parameters. All methods provide similar localization results for well-sampled cases. For all cases considered, RMMP provides a substantial improvement in localization compared to other MMP methods. In addition, RMMP localizes at an equivalent or better level than MFP for most cases, and provides significantly better results when environmental mismatch is included.

Simplex Simulated Annealing: A Hybrid Approach to Geoacoustic Inversion with Application to Mediterranean Sea Acoustic Data

Mark R. Fallat
School of Earth and Ocean Sciences, University of Victoria
(Supervisor: Stan Dosso)

Recently, in ocean acoustics, there has been a great deal of interest in the development of inversion algorithms. One problem these algorithms have been applied to is geoacoustic inversion. In geoacoustic inversion, properties of the seabed are sought from measurements of acoustic energy that has propagated through the ocean environment. In this thesis, a new hybrid algorithm, simplex simulated annealing, is developed for the geoacoustic inversion problem. Simplex simulated annealing is a combination of the local downhill simplex method and a fast simulated annealing global search algorithm. Hybrid algorithms are effective and efficient because they exploit the good features of the local and global algorithms while overcoming the respective weaknesses of the techniques. The simplex simulated annealing algorithm is validated using synthetic test-cases and is compared with both downhill simplex and simulated annealing. The new algorithm is also applied to experimental data recorded in the Mediterranean Sea. For the experimental inversion the results are in good agreement with the available ground truth data and results from previous experiments conducted in the same region.

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

2000

30 May-3 June: 139th Meeting of the Acoustical Society of America, Atlanta, GA. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; Web: asa.aip.org

5-9 June: International Conference on Acoustics, Speech and Signal Processing (ICASSP-2000), Istanbul, Turkey. Contact: T. Adali, EE and Computer Science Department, University of Maryland Baltimore County, 1000 Hilltop Circle, Baltimore, MD 21250; Fax: +1 410 455 3639; Web: icassp2000.sdsu.edu

6-9 June: 5th International Symposium on Transport Noise and Vibration, St. Petersburg, Russia. Contact: East-European Acoustical Association, Moskovskoe Shosse 44, 196158 St. Petersburg, Russia; Fax: +7 812 1279323; Email: noise@mail.rcom.ru

14-17 June: IUTAM Symposium on Mechanical Waves for Composite Structures Characterization, Chania, Crete, Greece. Contact: IUTAM 2000, Applied Mechanics Laboratory, Technical University of Crete, Chania 73100, Greece; Fax: +30 821 37438; Web: www.tuc.gr/iutam

4-7 July: 7th International Congress on Sound and Vibration, Garmisch-Partenkirchen, Germany. Contact: H. Heller, DLR, Postfach 3267, 38022 Braunschweig, Germany; Fax: +49 531 295 2320; email: hanno.heller@dlr.de; WWW: www.iiav.org/icsv7.html

9-13 July: 19th International Congress on Education of the Deaf & 7th Asia-Pacific Congress on Deafness, Sydney, Australia. Contact: ICED 2000 Secretariat, GPO Box 128, Sydney, NSW 2001 Australia; Fax: +61 2 9262 3135; Web: www.iced2000.com

10-13 July: 5th European Conference on Underwater Acoustics, Lyon, France. Contact: LASSSO, 43 Bd du 11 novembre 1918; Bat. 308; BP 2077, 69616 Villeurbanne cedex, France; Fax: +33 4 72 44 80 74; Web: www.ecua2000.cpe.fr

10-14 July: 5th International Conference on Mathematical and Numerical Aspects of Wave Propagation, Santiago de Compostela, Spain. Contact: Waves2000 Secretariat, Domaine de Voluceau, BP 105, 78153 Le Chesnay Cedex, France; Web: www.usc.es/waves2000

28-30 August: Inter-Noise 2000, Nice, France. Contact: SFA, 23 avenue Brunetière, 75017 Paris, France; Fax: +33 1 47 88 90 60; Web: www.inrets.fr/services/manif

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

2000

30 mai - 3 juin: 139e rencontre de l'Acoustical Society of America, Atlanta, GA. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél.: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; Web: asa.aip.org

5-9 juin: Conférence internationale sur l'acoustique, la parole et le traitement de signal (ICASSP-2000), Istanboul, Turquie. Info: T. Adali, EE and Computer Science Department, University of Maryland Baltimore County, 1000 Hilltop Circle, Baltimore, MD 21250; Fax: +1 410 455 3639; Web: icassp2000.sdsu.edu

6-9 juin: 5e symposium international sur le bruit et vibrations du transport, St Petersbourg, Russie. Info: East-European Acoustical Association, Moskovskoe Shosse 44, 196158 St. Petersburg, Russia; Fax: +7 812 1279323; Email: noise@mail.rcom.ru

14-17 juin: Symposium IUTAM sur les ondes mécaniques pour la caractérisation de structures composites, Chania, Crète, Grèce. Info: IUTAM 2000, Applied Mechanics Laboratory, Technical University of Crete, Chania 73100, Greece; Fax: +30 821 37438; Web: www.tuc.gr/iutam

4-7 juillet: 7e Congrès international sur le son et les vibrations, Garmisch-Partenkirchen, Allemagne. Info: H. Heller, DLR, Postfach 3267, 38022 Braunschweig, Germany; Fax: +49 531 295 2320; email: hanno.heller@dlr.de; WWW: www.iiav.org/icsv7.html

9-13 juillet: 19e Congrès international sur l'éducation des sourds et 7e Congrès Asie-Pacifique sur la surdité, Sydney, Australie. Info: ICED 2000 Secretariat, GPO Box 128, Sydney, NSW 2001 Australia; Fax: +61 2 9262 3135; Web: www.iced2000.com

10-13 juillet: 5e Conférence européenne sur l'acoustique sous-marine, Lyon, France. Info: LASSSO, 43 Bd du 11 novembre 1918; Bat. 308; BP 2077, 69616 Villeurbanne cedex, France; Fax: +33 4 72 44 80 74; Web: www.ecua2000.cpe.fr

10-14 juillet: 5e Conférence internationale sur les aspects mathématiques et numériques de la propagation d'onde, Santiago de Compostela, Espagne. Info: Waves2000 Secretariat, Domaine de Voluceau, BP 105, 78153 Le Chesnay Cedex, France; Web: www.usc.es/waves2000

28-30 août: Inter-Noise 2000, Nice, France. Info: SFA, 23 avenue Brunetière, 75017 Paris, France; Fax: +33 1 47 88 90 60; Web: www.inrets.fr/services/manif

31 August – 2 September: International Conference on Noise and Vibration Pre-Design and Characterization Using Energy Methods (NOVEM), Lyon, France. Contact: LVA, INSA de Lyon, Bldg. 303, 20 avenue Albert Einstein, 69621 Villeurbane, France; Fax: +33 4 7243 8712; Web: www.insa-lyon.fr/Laboratoires/lva.html

3-6 September: 5th French Congress on Acoustics — Joint meeting of the Swiss and French Acoustical Societies, Lausanne, Switzerland. Contact: M.-N. Rossi, Ecole Polytechnique Fédérale, 1015 Lausanne, Switzerland; Fax: +41 21693 26 73.

13-15 September:International Conference on Noise and Vibration Engineering (ISMA 25), Leuven. Contact: Mrs. L. Notré, K. U. Leuven, PMA Division, Celestijnenlaan 300B, 3001 Leuven, Belgium; Fax:+32 16 32 24 82; Email:lieve.notre@mech.kuleven.ac.be

17-21 September: Acoustical Society of Lithuania First International Conference, Vilnius, Lithuania. Contact: Acoustical Society of Lithuania, Kriviu 15-2, 2007 Vilnius, Lithuania; Fax: +370 2 223 451; Email: daumantas.ciblys@ff.vu.lt

18-22 September: 47th Seminar on Acoustics (OSA2000), Zalem Solinski, Poland. Email: osa@atena.univ.rzeszow.pl

3-5 October: WESPRAC VII, Kumamoto, Japan. Contact: Computer Science Dept., Kumamoto Univ., 2-39-1 Kurokami, Kumamoto, Japan 860-0862; Fax: +81 96 342 3630; Email: wesprac7@cogni.eecs.kumamoto-u.ac.jp

3-6 October: EUROMECH Colloquium on Elastic Waves in Nondestructive Testing, Prague, Czech Republic. Contact: Z. Prevorovsky, Institute of Thermomechanics, Dolejskova 4, 182 00 Prague 8, Czech Republic; Fax: +420 2 858 4695; Email: ok@bivoj.it.cas.cz

12-14 October: International Conference on Newborn Hearing Screening, Milan. Fax: +39 2 23993360; Web: www.biomed.polimi.it/diphtm/WorkShops.html

16-18 October: 2nd Iberoamerican Congress on Acoustics, 31st National Meeting of the Spanish Acoustical Society, and EAA Symposium, Madrid, Spain. Contact: Spanish Acoustical Society, c/Serrano 144, 28006 Madrid, Spain; Fax: +34 91 411 7651; email: ssantiago@fresno.csic.es

16-20 October: 6th International Conference on Spoken Language Processing, Beijing, China. Contact: ICSLP 2000 Secretariat, Institute of Acoustics, PO Box 2712, 17 Zhong Guan Cun Road, 100 080 Beijing, China; Fax: +86 10 6256 9079; Email: mchu@plum.ioa.ac.cn

28-29 October: Acoustics Week in Canada, Sherbrooke, QC. Contact: Alain Berry, Université de Sherbrooke, 2500, boul. Université, Sherbrooke, QC, J1K 2R1, Canada. Fax:819-821-7163; Email:alain.berry@gme.usherb.ca; Web:www.gaus.gme.usherb.ca

4-8 December: 140th Meeting of the Acoustical Society of America, Newport Beach, CA. Contact: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tel: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; Web: asa.aip.org

31 août – 2 septembre: Conférence internationale sur l'utilisation des méthodes d'énergie pour la prévision vibroacoustique (NOVEM), Lyon, France. Info: LVA, INSA de Lyon, Bldg. 303, 20 avenue Albert Einstein, 69621 Villeurbane, France; Fax: +33 4 7243 8712; Web: www.insa-lyon.fr/Laboratoires/lva.html

3-6 septembre: 5e Congrès français d'acoustique — Rencontre conjointe des Sociétés suisse et française d'acoustique, Lausanne, Suisse. Info: M.-N. Rossi, Ecole Polytechnique Fédérale, 1015 Lausanne, Suisse; Fax: +41 21693 26 73.

13-15 septembre: Conférence internationale d'ingénierie sur le bruit et les vibrations (ISMA 25), Leuven, Belgique. Info: Mrs. L. Notré, K. U. Leuven, PMA Division, Celestijnenlaan 300B, 3001 Leuven, Belgium; Fax: +32 16 32 24 82; Email: lieve.notre@mech.kuleuven.ac.be

17-21 septembre: 1e Conférence internationale de la Société d'acoustique de Lithuanie, Vilnius, Lithuanie. Info: Acoustical Society of Lithuania, Kriviu 15-2, 2007 Vilnius, Lithuania; Fax: +370 2 223 451; Email: daumantas.ciblys@ff.vu.lt

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3-6 octobre: Colloque EUROMECH sur les ondes élastiques pour les tests non-destructifs, Prague, République tchèque. Info: Z. Prevorovsky, Institute of Thermomechanics, Dolejskova 4, 182 00 Prague 8, Czech Republic; Fax: +420 2 858 4695; Email: ok@bivoj.it.cas.cz

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16-18 octobre: 2e congrès ibéro-américain sur l'acoustique, 31e Rencontre nationale de la Société d'acoustique espa-gnole, Symposium de l'EAA, Madrid, Espagne. Info: Spanish Acoustical Society, c/Serrano 144, 28006 Madrid, Spain; Fax: +34 91 411 7651; email: ssantiago@fresno .csic.es

16-20 octobre: 6e conférence internationale sur le traitement de la langue parlée, Beijing, Chine. Info: ICSLP 2000 Secretariat, Institute of Acoustics, PO Box 2712, 17 Zhong Guan Cun Road, 100 080 Beijing, China; Fax: +86 10 6256 9079; Email: mchu@plum.ioa.ac.cn

28-29 octobre: Semaine d'acoustique canadienne, Sherbrooke, QC. Info: Alain Berry, Université de Sherbrooke, 2500, boul. Université, Sherbrooke, QC, J1K 2R1, Canada. Fax:819-821-7163; Email: alain.berry@gme.usherb.ca; Web: www.gaus.gme.usherb.ca

4-8 décembre: 140e recontre de l'Acoustical Society of America, Newport Beach, CA. Info: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Tél.: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org; Web: asa.aip.org

2001

30 April-3 May: 2001 Society of Automotive Engineers (SAE) Noise & Vibration Conference and Exposition, Traverse City, MI. Contact: Patti Kreh, SAE Int'l., 755 W. Big Beaver Rd., Suite 1600, Troy, MI 48084; Tel.: 248-273-2474; Fax: 248-273-2494; Email: pkreh@sae.org

28-30 August: Inter-Noise 2001, The Hague, The Netherlands. Email: secretary@internoise2001.tudelft.nl; Web: internoise2001.tudelft.nl

2-7 September: 17th International Congress on Acoustics (ICA), Rome, Italy. Contact: A. Alippi, Dipartimento di Energetica, Universita di Roma "La Sapienza," Via A. Scarpa 14, 00161 Rome, Italy; Fax: +39 6 4424 0183; WWW: www.uniroma1.it/energ/ica.html

10-13 September: International Symposium on Musical Acoustics (ISMA 2001), Perugia, Italy. Contact: Perugia Classico, Comune di Perugia, Via Eburnea 9, 06100 Perugia, Italy; Fax: +39 75 577 2255; Email: perugia@classico.it

17-19 October: 32nd Meeting of the Spanish Acoustical Society, La Rioja, Spain. Contact: Serrano 144, Madrid 28006, Spain; Fax: +34 91 411 76 51; Web: www.ia.csic.es/sea/index.html

2001

30 avril-3 mai: Conférence et exposition 2001 de la Société des Ingénieurs d'autos (SAE) sur le bruit et les vibrations, Traverse City, MI. Info: Patti Kreh, SAE Int'l., 755 W. Big Beaver Rd., Suite 1600, Troy, MI 48084; Tél.: 248-273-2474; Fax: 248-273-2494; Email: pkreh@sae.org

28-30 août: Inter-Noise 2001, La Haye, Pays-Bas. Email: secretary@internoise2001.tudelft.nl; Web: internoise2001 .tudelft.nl

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10-13 septembre: Symposium international sur l'acoustique musicale (ISMA 2001), Perugia, Italie. Info: Perugia Classico, Comune di Perugia, Via Eburnea 9, 06100 Perugia, Italy; Fax: +39 75 577 2255; Email: perugia@classico.it

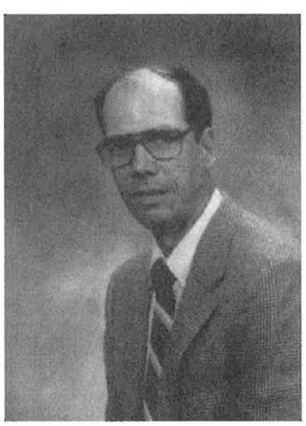
17-19 octobre: 32e rencontre de la Société espagnole d'acoustique, La Rioja, Espagne. Info: Serrano 144, Madrid 28006, Spain; Fax: +34 91 411 76 51; Web: www.ia.csic.es/sea/index.html

CANADIAN STANDARDS NEWS

Compiled by Tim Kelsall, Hatch Associates, Mississauga, Ontario

Cameron Sherry Wins CSA Award of Merit

Cameron Sherry, who is well known as a former President of CAA, will be honoured this June by CSA International, who are awarding him with their Award of Merit at their annual convention in Ottawa. Cameron has been active with CSA Standards since the 70's and has served as Chair of the Industrial Noise Subcommittee and more recently as chair of CSA's Acoustics and Noise Control Main Committee, which coordinates most Acoustics Standards writing and review in this country. The subcommittee also acts as an unofficial technical reviewer for many pieces of noise legislation across the country. Cameron has also been active with ASTM standards for many years.



New CSA Z107.9 Standard on Noise Barriers

Specifying a noise barrier just became a lot easier with the introduction of CSA Z107.9-00 Standard for Certification of Noise Barriers. Based on the widely used Ontario MTO noise barrier specifications, the standard sets out the requirements for manufacturers to certify their barrier systems as suitable for noise attenuation and capable of standing up for at least 20 years. In addition, municipalities, engineers, etc. can have particular barrier installations certified as meeting the requirements of the standard.

The standard fills a big hole across the continent. The provisions of the standard will also be found in the FHWA Highway Noise Barrier Design Handbook and ANSI is currently looking at adopting the standard. This would mean that one certification might one day be recognised throughout North America. At this point it is believed that the standard is the only barrier certification available. As such it is hoped that acoustical consultants, municipalities, provincial authorities etc. will start to require CSA certified barrier systems and installations to avoid sub-standard installations which have on occasion happened in the past. Copies of the standard can be obtained directly from CSA or ordered electronically. Questions about the standard can be directed to Soren Pedersen at 416-231-4514 or Tim Kelsall at 905-403-3932.

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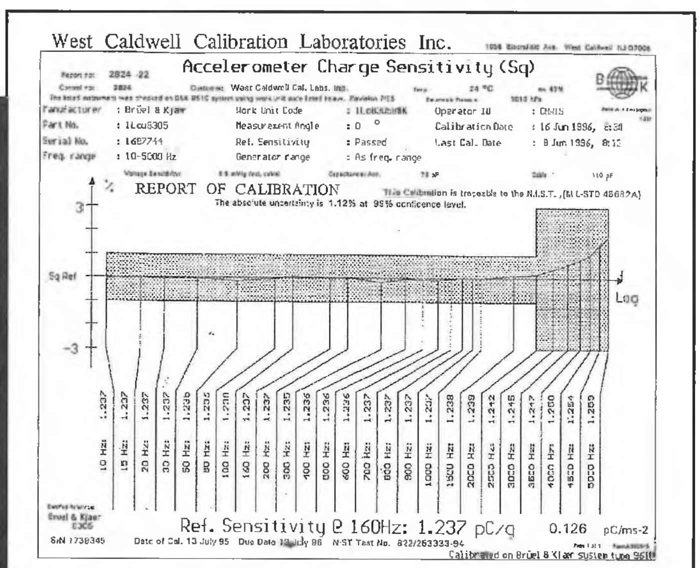
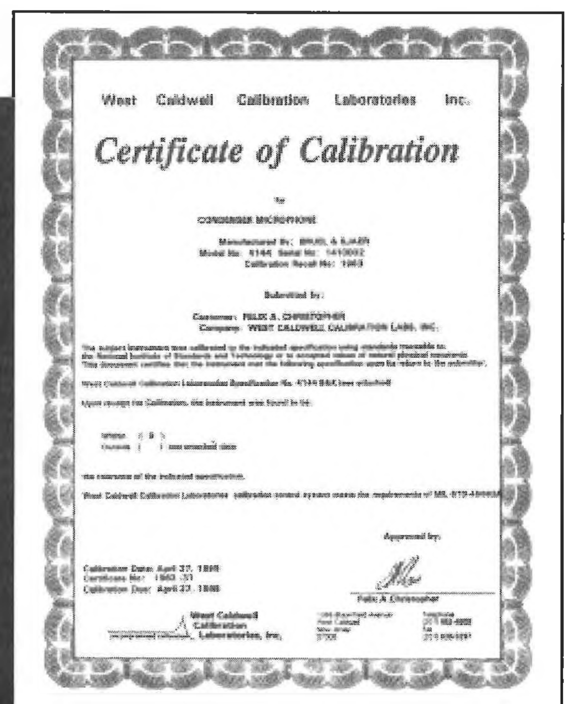


Accelerometer calibration performed on the system chosen by the U.S. Air Force [PMEL]



Appel de Communications
Semaine canadienne d'acoustique 2000
Hôtel Delta, Sherbrooke, Qc
28-29 septembre, 2000

COMITÉ D'ORGANISATION

GAUS
GROUPE D'ACOUSTIQUE DE L'UNIVERSITÉ DE SHERBROOKE

La semaine de l'acoustique canadienne de l'an 2000 se déroulera à l'hôtel Delta de Sherbrooke, Québec, du 28-29 septembre 2000. La semaine sera organisée, sous l'égide de la société canadienne d'acoustique, par le groupe d'acoustique de l'Universite de Sherbrooke. Le professeur Noureddine Atalla de l'université de Sherbrooke agira comme président et le professeur Alain Berry de l'université de Sherbrooke agira comme directeur du programme technique.

Courriel : caa2000@gaus.gme.usherb.ca
Site web : http://www-gaus.gme.usherb.ca

PROGRAMMES SCIENTIFIQUES ET TECHNIQUES

Le comité d'organisation planifie un programme technique de grand calibre avec une emphase particulière sur **le contrôle du bruit industriel**. Le congrès comprend des conférences générales, des communications sur invitation, des sessions et des expositions techniques. Les sujets traités recouvrent :

Le contrôle actif du bruit et des vibrations	Contrôle du bruit industriel	Matériaux acoustiques
Les méthodes analytiques et numériques	Vibro-acoustique	Acoustique architecturale
Aéroacoustique	Propagation du son	Acoustique sous-marine
Psycho-acoustique	Physio-acoustique	Perception et production du langage
Acoustique musicale	Qualité du son	Audiologie
Normalisation canadienne	Règlements et bruit environnemental	

Enseignement et démonstration en acoustique et Vibrations, **et tout autre sujet relevant de l'acoustique.**

RÉSUMÉS

Les résumés de 250 mots maximums doivent être soumis avant le 31 mai 2000. Les résumés devront être préparés suivant les instructions incluses dans ce numéro d'*Acoustique canadienne*. Les soumissions par courrier électronique sont fortement encouragées; les documents peuvent être édités avec n'importe quel traitement de texte. Pour ceux qui n'ont pas accès au courrier électronique, les documents digitaux sur disquette ou papier devront être envoyés à l'adresse indiquée ci-dessous. Une notification d'acceptation des résumés sera envoyée aux auteurs avant le 15 juin 2000 avec un formulaire d'inscription. Un sommaire de la présentation devra être envoyé avant le 31 juillet 2000. Cette échéance sera strictement respectée afin de pouvoir publier le programme dans les actes d'*Acoustique canadienne*.

Les propositions pour les sessions spéciales sur un sujet particulier en acoustique sont les bienvenues. Contactez Dr A. Berry (alain.berry@gme.usherb.ca) avant le 31 mai 2000 si vous désirez organiser une session spéciale durant la conférence de cette année.

Proposals for **Special Sessions** on a particular topic in acoustics are welcome. Contact Dr. Alain Berry (alain.berry@gme.usherb.ca) prior to Mai 31, 2000 if you are interested in having a special session at this year's meeting.

Student participation in Acoustics Week in Canada is strongly encouraged. Awards are available to students whose presentations at the conference are judged to be particularly noteworthy. To qualify, students must apply by enclosing an *Annual Student Presentation Award* form with their abstract. Students presenting papers may also apply for a travel subsidy to attend the meeting if they live at least 150 km from Sherbrooke, Qc. To apply for this subsidy, students must submit an *Application For Student Travel Subsidy*, included in this issue.

ACCOMMODATION

Accommodation and meeting space for delegates of Acoustics Week in Canada 2000 will be at the Delta Hotel (http://www.deltahotels.com/properties/sherbrk.html) located in downtown Sherborooke, Qc. The special room rate for delegates is $95.00 per night. To reserve your accommodation, please contact the hotel directly by telephone (1 800 268-1133), Fax (819-822-8990) and mention the identification code: GPACOU. The reservation cut-off date is August 27, 2000. After these dates, the special rates are subject to availability.

There are several other hotels for every budget, located within walking distance from the conference site.
For details, check the tourist web site of Sherbrooke : http://www.sders.com/tourisme.

TRANSPORTATION

Sherbrooke is 150-km from Dorval Airport in Montreal. A regular bus service links the airport to Downtown Sherbrooke. A special transport service can be organized, on request. If you are interested in such a service, please contact Rémy Oddo (819) 821-8000x1965, remy.oddo@gme.usherb.ca

EXHIBITS

A permanent exhibition showing the latest technologies in acoustics and vibration equipment, instrumentation, materials and software will be open continuously during the congress.

Space will be available for **exhibits** by companies and organizations in the field of acoustics. **Sponsorship** of nutrition breaks and/or lunches is also welcome. If you are interested in either of these opportunities, please contact Rémy Odd (819) 821-8000x1965, remy.odd@gme.usherb.ca

IMPORTANT DATES

May 31, 2000	Deadline for submission of abstracts
June 15, 2000	Notification of acceptance of abstracts
July 31, 2000	Deadline for receipt of summary paper and early registration
September 28-29, 2000	Acoustics week in Canada 2000

For more information contact:

Acoustics Week in Canada 2000
c/o Dr. Noureddine Atalla
Génie mécanique, Université de Sherbooke
Sherbooke (Qc), Canada J1K 2R1
Téléphone : (819) 821-8000 x1209 Fax : (819) 821-7163
Noureddine.atalla@gme.usherb.ca

Call for papers
Acoustics Week in Canada 2000
Delta Hotel, Sherbrooke, Qc
2000, September 28-29

ORGANISING COMMITTEE

GAUS
GROUPE D'ACOUSTIQUE DE L'UNIVERSITÉ DE SHERBROOKE

Acoustics Week in Canada 2000 will be held at the Delta hotel, in Sherbrooke, Quebec, in 2000 September. The week will open on Thursday, September 28 and will conclude in the afternoon of September 29. The conference, sponsored by the Canadian Acoustics Association, will be organized by the groupe d'acoustique de l'université de Sherbrooke. Professor Noureddine Atalla of l'université de Sherbrooke is the president of CAA2000. Professor Alain Berry of l'université de Sherbrooke is the Technical Program Chair.

Conference Email : caa2000@gaus.gme.usherb.ca
Web Site : http://www-.gaus.gme.usherb.ca

SCIENTIFIC AND TECHNICAL PROGRAMS

The organizing committee is planning a high caliber and motivating technical program. The program will deal with topics from throughout the field of acoustics and vibrations with a special emphasis **on industrial passive and active control**. The meeting will consist of an opening plenary lecture, invited and contributed papers and exhibits. Technical papers in all areas of noise control engineering will be considered for presentation at the conference. The following technical areas are of particular interest:

Active Noise Control for Industry	Industrial Noise Control - case studies	Acoustic materials
Analytical and Numerical prediction tools in Acoustics	Structural acoustics and vibrations	
Building acoustic	Aeroacoustics	Underwater acoustics
Outdoor sound propagation	Psycho-acoustics	Physiological acoustics
Speech perception and production	Musical acoustics	Sound quality
Occupational Hearing Loss and Hearing protection	Canadian Standards	Legislation /Environmental Noise

Education and Demonstration in Noise Control Engineering, **and other related topics**

ABSTRACTS

Abstracts of maximum 250 words must be submitted by May 31, 2000. The abstract should be prepared and sent in accordance with the instructions appearing in this issue of *Canadian Acoustics*. Submission by e-mail is strongly encouraged; files can be prepared in any word processing software. For those without access to e-mail, digital files on diskette or paper copy should be mailed to the address given in the instructions. Notification of acceptance of abstracts will be sent to authors by June 15, 2000 along with a registration form. Summary papers are due July 15, 2000. This deadline will be strictly enforced to meet the publication schedule of the proceeding issue of *Canadian Acoustics*.

Proposals for **Special Sessions** on a particular topic in acoustics are welcome. Contact Dr. Alain Berry (alain.berry@gme.usherb.ca) prior to Mai 31, 2000 if you are interested in having a special session at this year's meeting.

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Sherbooke (Qc), Canada J1K 2R1
Téléphone : (819) 821-8000 x1209 Fax : (819) 821-7163
Noureddine.atalla@gme.usherb.ca

INSTRUCTIONS TO AUTHORS FOR THE PREPARATION OF MANUSCRIPTS

Submissions: The original manuscript and two copies should be sent to the Editor-in-Chief.

General Presentation: Papers should be submitted in camera-ready format. Paper size 8.5" x 11". If you have access to a word processor, copy as closely as possible the format of the articles in Canadian Acoustics 18(4) 1990. All text in Times-Roman 10 pt font, with single (12 pt) spacing. Main body of text in two columns separated by 0.25". One line space between paragraphs.

Margins: Top - title page: 1.25"; other pages, 0.75"; bottom, 1" minimum; sides, 0.75".

Title: Bold, 14 pt with 14 pt spacing, upper case, centered.

Authors/addresses: Names and full mailing addresses, 10 pt with single (12 pt) spacing, upper and lower case, centered. Names in bold text.

Abstracts: English and French versions. Headings, 12 pt bold, upper case, centered. Indent text 0.5" on both sides.

Headings: Headings to be in 12 pt bold, Times-Roman font. Number at the left margin and indent text 0.5". Main headings, numbered as 1, 2, 3, ... to be in upper case. Sub-headings numbered as 1.1, 1.2, 1.3, ... in upper and lower case. Sub-sub-headings not numbered, in upper and lower case, underlined.

Equations: Minimize. Place in text if short. Numbered.

Figures/Tables: Keep small. Insert in text at top or bottom of page. Name as "Figure 1, 2, ..." Caption in 9 pt with single (12 pt) spacing. Leave 0.5" between text.

Line Widths: Line widths in techincal drawings, figures and tables should be a minimum of 0.5 pt.

Photographs: Submit original glossy, black and white photograph.

Scans: Should be between 225 dpi and 300 dpi. Scan: Line art as bitmap tiffs; Black and white as grayscale tiffs and colour as CMYK tiffs;

References: Cite in text and list at end in any consistent format, 9 pt with single (12 pt) spacing.

Page numbers: In light pencil at the bottom of each page.

Reprints: Can be ordered at time of acceptance of paper.

DIRECTIVES A L'INTENTION DES AUTEURS PREPARATION DES MANUSCRITS

Soumissions: Le manuscrit original ainsi que deux copies doivent être soumis au rédacteur-en-chef.

Présentation générale: Le manuscript doit comprendre le collage. Dimensions des pages, 8.5" x 11". Si vous avez accès à un système de traitement de texte, dans la mesure du possible, suivre le format des articles dans l'Acoustique Canadienne 18(4) 1990. Tout le texte doit être en caractères Times-Roman, 10 pt et à simple (12 pt) interligne. Le texte principal doit être en deux colonnes séparées d'un espace de 0.25". Les paragraphes sont séparés d'un espace d'une ligne.

Marges: Dans le haut - page titre, 1.25"; autres pages, 0.75"; dans le bas, 1" minimum; latérales, 0.75".

Titre du manuscrit: 14 pt à 14 pt interligne, lettres majuscules, caractères gras. Centré.

Auteurs/adresses: Noms et adresses postales. Lettres majuscules et minuscules, 10 pt à simple (12 pt) interligne. Centré. Les noms doivent être en caractères gras.

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