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PROCEEDINGS ISSUE

ACOUSTICS WEEK IN CANADA
SEMAINE CANADIENNE D'ACOUSTIQUE

1996

CAHIER DES ACTES

canadian acoustics

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THE CANADIAN ACOUSTICAL ASSOCIATION
P.O. BOX 1351, STATION "F"
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ÉDITORIAL / EDITORIAL

Bienvenu au Cahier des Actes 1996 de l'*Acoustique Canadienne* - le cahier des actes de la Semaine Canadienne d 'Acoustique 1996 qui aura lieu à Calgary en octobre. J'ai le plaisir de publier plus de 70 sommaires de la conférence sur des sujets variés d'acoustique et vibrations, ce qui confirme de nouveau le haut degré d'activité dans ces domaines au Canada. J'espère vous voir à Calgary.

Depuis mon dernier éditorial, l'*Acoustique Canadienne* a dû confronter deux problèmes importants. Premièrement, notre imprimeur a fait faillite! Heureusement, notre représentant avec cette compagnie a rejoint une autre imprimerie, il nous a aidé à sauver notre matériel du premier imprimeur, et a soumis une offre de prix raisonnable pour prendre à sa charge l'impression d'*Acoustique Canadienne*. J'ai bon espoir que le problème soit résolu.

Le deuxième problème, toujours à régler, concerne les implications à long terme des difficultés financières présentes d'*Acoustique Canadienne* (voir le compte rendu dans le dernier numéro). Ce problème sera discuté en détail à Calgary, mais nous devrons accroître nos revenus (publicité, prix) ou réduire nos dépenses, incluant une réduction du coût de production de l'*Acoustique Canadienne.*

Le coût de production du journal dépend surtout du nombre de copies imprimées (i.e. le nombre de membres de l'ACA) et le nombre de pages. Réduire le nombre de pages implique une réduction du nombre d'articles publiés - ce qui est contraire à nos objectifs - et/ou une réduction du nombre de varias, tel l'annuaire annuel des membres. On pourrait aussi supprimer les illustrations de couverture. Tout ceux qui ont des commentaires à ce sujet devraient assister à l'Assemblée générale annuelle à Calgary, ou me contacter avant la conférence.

Welcome to the 1996 Proceedings Issue of *Canadian Acoustics* - the proceedings of Acoustics Week in Canada 1996 to be held in Calgary in October. I have the pleasure of publishing over 70 conference summaries on a wide range of topics in acoustics and vibration - reaffirming the considerable activity taking place in Canada in these fields. I hope to see you in Calgary.

Since my last editorial, *Canadian Acoustics* has had to deal with two difficult problems. First, our printer went bankrupt! Fortunately, our 'rep' with that company joined another printing firm, helped me to 'rescue' our property from the old printer and has submitted a reasonable quote to take over printing *Canadian Acoustics.* Hopefully, problem resolved.

Yet to be resolved are the long-term implications of the Association's current financial difficulties for *Canadian Acoustics* (see the Minutes in the last issue). As will be discussed in detail in Calgary, either we must increase revenue (advertising, fees) or decrease expenditures - including reducing the cost of *Canadian Acoustics.*

The cost of producing the journal depends mainly on the number of copies printed (i.e. the number of CAA members) and the number of pages. Reducing the number of pages means reducing the number of papers published - contrary to current objectives - and/or cutting features, such as the annual membership directory. It could also mean dispensing with the cover illustrations. Anyone with comments on this situation should attend the Annual General Meeting in Calgary or contact me before the conference.

PROCEEDINGS OF ACOUSTICS WEEK IN CANADA 1996
ACTES DE LA SEMAINE CANADIENNE D'ACOUSTIQUE 1996

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RECENT TECHNICAL INNOVATIONS IN THE HARDWARE AND APPLICATION OF ACTIVE NOISE CONTROL FOR INDUSTRIAL EQUIPMENT.

Active Noise Control has been used since the late 1980's to solve low frequency duct-borne noise problems facing industry. Early efforts consisted of complex, narrowly applied systems that often required custom engineering to meet the requirements of a particular application.

These initial efforts were followed by strong engineering efforts concentrating on two fronts: development of robust, inexpensive hardware and development of accurate application guidelines. This paper will provide an overview of the results of these efforts.

The basic components of an industrial active noise control system (Figure #1) consist of a digital controller/amplifier, cancellation loudspeakers, and detection microphones.

The dsp based controller/amplifier uses adaptive filtering techniques to make an electronic representation of the duct acoustics. The controller computes the delay and amplitude changes from the input microphone to the loudspeaker and determines the appropriate cancellation signal. A downstream microphone provides the controller with feedback on the level of performance being achieved. On-line calibration is also used to maintain optimum cancellation under changing conditions. Integrated into the controller package are eight high powered amplifier modules. The controller has front panel failure lights to indicate any system problems to the customer.

The cancellation loudspeakers are capable of matching the sound power generated by the noise source (fan, pump, or engine). An acoustically transparent diaphragm is often used to isolate the driver from the exhaust environment.

The detection microphones are able to measure high sound pressure levels (150 dBspl) without significant distortion. They are designed to perform continuously in wet, corrosive exhaust streams.

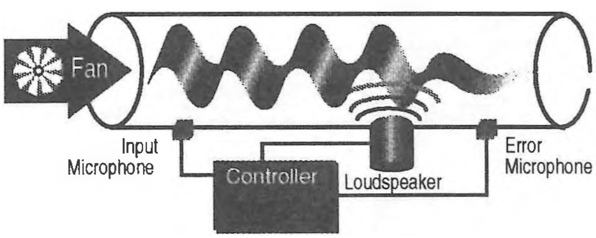

Figure #1: Component Layout of Active Silencer

Although hardware improvements have played a major role in making this technology more competitive with existing technology, the accumulation of substantial amounts of field knowledge has also played an important role. This knowledge has allowed for the development of a fairly broad application window.

A summary of the current application guidelines for active silencers:

*Tonal & Broadband reduction of 10 -30 dB.
* Performance from 31- 500 Hz.
*Upper temperature limit of 400 degrees F.
* Flow velocity of <8000 fpm.
* Maximum sound power of 135 dB.

The silencer can now be placed directly downstream of the noise source or packaged with an absorptive silencer for additional mid frequency noise control. All hardware is placed outside the flow stream and imparts no additional pressure drop on the system. Figure #2 depicts a typical industrial installation on a large centrifugal fan application. On fan applications it is often possible to place all active components in the first duct diameter from the fan.

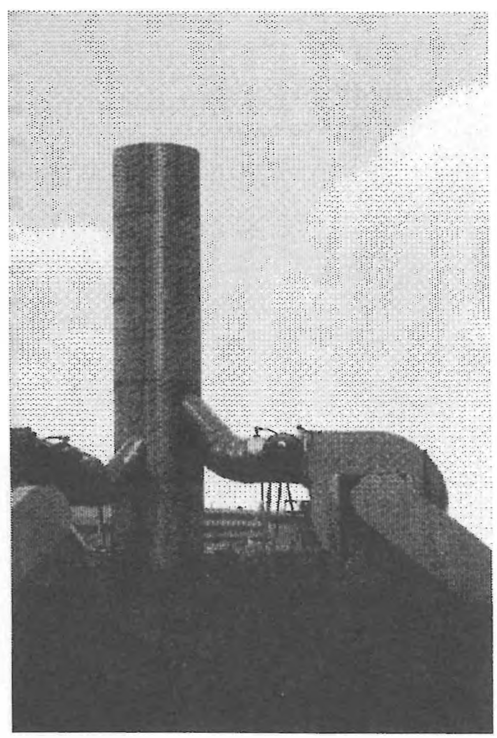

Figure #2: Field Installation of Active Silencer

Noise Control in Buildings Without Exposed Fiberglass

Susan H. Dineen, Lawrence J. Gelin, Steve Wise
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Fan noise in HVAC ducts has traditionally been attenuated with fibrous internal duct liner or with passive silencers constructed with porous fill material. Because these materials can absorb dust and moisture and provide a breeding ground for microbes, building designers and owners are reducing or eliminating the use of such materials.

With active noise control (ANC) it is possible to cancel low frequency fan noise without using porous or fibrous material. ANC systems use inert components such as microphones and loudspeakers to cancel broadband, ductborne fan noise from air handlers up to 25,000 cfm. They can achieve up to 20dB reduction of noise in the 63, 125, and 250Hz octave bands. Attenuation of up to 10dB in the 31.5 Hz band can also be achieved.

Active noise control was applied to two air handlers in an office building in Canada. Each unit is 9,000 cfm with a top horizontal discharge into a 20" diameter unlined duct. Sound levels as high as NC54 were measured in the office areas. The active systems achieved 5 to 20 dB reduction of the noise between 50 and 250Hz, resulting in an NC38. (Fig.1)

In another installation, active control was applied to attenuate broadband noise as well as a 165Hz tone. (Fig.2) The noise propagated through an unlined supply air duct and then broke out of the 34" x 12" flat oval duct over the noise -sensitive space. (Fig.3) The ambient noise - NC38 - in the room was a problem since the room is used for consumer research tests which require auditory discrimination. Active cancellation of the ductborne noise reduced the noise levels in the room to NC18. (Fig.3)

Active noise control solutions are also applied in clean rooms for semiconductor manufacturing spaces and pharmaceutical production and packaging facilities. In one installation which has been operating for nearly three years, active noise control was applied to the discharge ducts of four air handling units serving a 350sq.ft clean room for pharmaceutical production. (Fig. 5) The measured sound level in the room was an unacceptable NC68.

Approximately 10 ft. of discharge duct was treated with a 1-inch thick encapsulated fiberglass sheet along the internal perimeter. The encapsulation prevents erosion and moisture absorption while providing attenuation of mid and high frequency noise. Active components, mounted directly to the duct, attenuate low frequency noise. The result is a 10 NC point reduction in the clean room, without the penalty of pressure drop. (Fig.6)

In clean room applications which use a plug, or plenum fan design, much of the sound energy is dissipated within the discharge plenum. However, there often remains a dominant tone at the blade pass frequency, typically between 80Hz and 150Hz. In these applications it is possible to locate the active noise control system at the discharge of the fan and cancel the tone in order to achieve reductions of up to 10 NC points in the clean space.

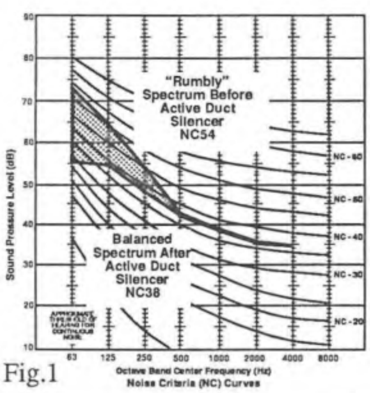

Fig.1

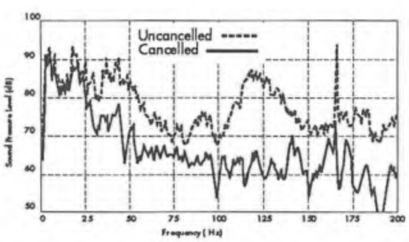

Fig.2

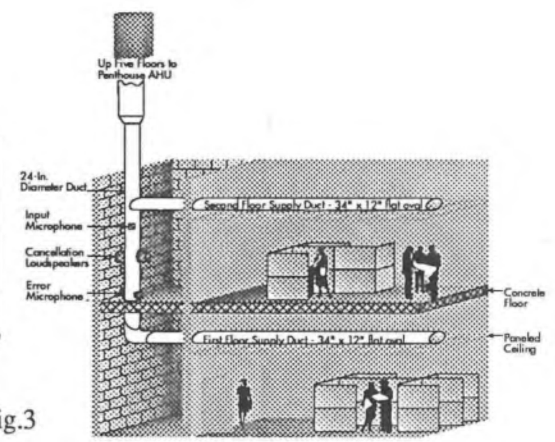

Fig.3

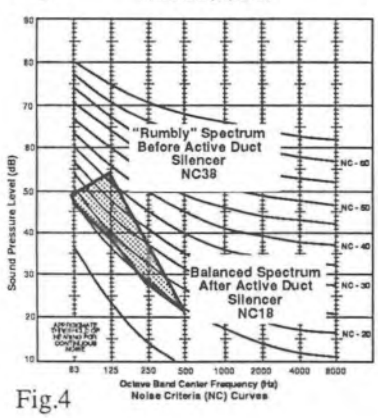

Fig.4

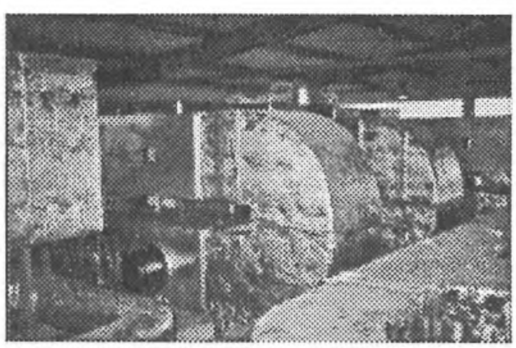
Fig.5

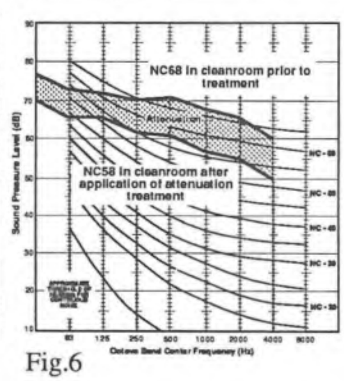

Fig.6

ACTIVE NOISE CONTROL AT THE O.H.S.C. CO-GEN FACILITY

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1 Introduction

Complaints about noise from the co-generation facility at the Ottawa Health Sciences Centre, OHSC, started when the unit went into operation in 1992. Owned and operated by TransAlta Energy Corporation, the non-utility power generating station supplies power and steam to the OHSC, and is located about 400m from a quiet residential area. Although active noise cancellation, in the past, has been applied with limited success for mitigating outdoor noise problems, AERCOUSTICS Engineering Limited designed and installed such a system in the summer of 1994, overcoming significant engineering constraints to complete a successful execution of this technology. Active noise cancellation is well suited to mitigating low frequency pure tone noise problems where sound is used to cancel the offending noise electronically by generating a mirror image or out of phase sound signal. The combination of noise and anti-noise results in an energy efficient solution that is one-fifth the cost of traditional silencer baffles.

2 Noise Source Description

AERCOUSTICS Engineering Limited was retained by the TransAlta Energy Corporation in 1993, to mitigate excessive low frequency noise from high speed operation of two wet surface air condensing or WetSAC, fans. These fans are axial, 26 ft. in diameter, have flow rates of 1,000,000 cubic feet per minute; the 15-foot exhaust stack is constructed of a stiff, lightweight quarter-inch fibreglass shroud. The low frequency noise was a result of a 23.8 Hz pure tone due to the bladepass frequency of the WetSAC exhaust fan(s). The resulting 23.8 Hz pure tone is evident in the exhaust sound spectra and in the sound radiated by the excitation of the stiff, lightweight shroud which radiates sound efficiently in this frequency range.

3 Permanent System Design And Installation

A permanent active noise control system 1 was engineered in the spring of 1994 and integrated into the overall operation at the OHSC facility that same summer. These include custom designed, innovative and unique loudspeakers as per Figure 1, and microphones which were both manufactured in Canada. They were designed to operate year round in the very humid WetSAC fan area over a temperature range of -40 to +40 °C.

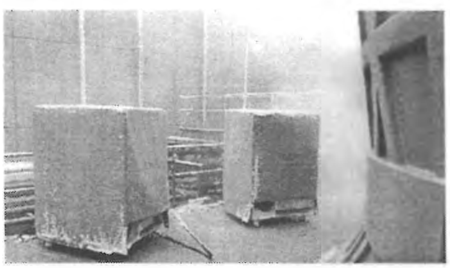

FIGURE 1: Active System Transducers. Speakers designed to withstand hostile environment.

Kevlar diaphragm loudspeakers are housed in heated enclosures built to withstand the hostile outdoor environment. Each loudspeaker system was designed and specifically tuned to the frequency range of interest to meet a target sound level of 125 dB at 1m at 24Hz. In total, two sets of four loudspeaker enclosures housing 16 18-inch drivers were placed circumferentially around each respective WetSAC fan to optimize cancellation. The active noise controller was designed to ensure that the offending noise source and cancellation sources are in anti-phase at the 23.8 Hz cancellation frequency. The system requires an absolute frequency reference for reliable operation which is supplied by a tachometer mounted at the fan drive shaft. There are also two reference microphones mounted on each fan shroud which provide the amplitude and phase reference of the fan noise. An error microphone mounted on each pair of loudspeakers monitors the amplitude and phase of the signal from the canceling noise source. The loudspeaker output amplitude is modulated with up to 9000 Watts of available amplifier power to optimize cancellation of the fan low frequency noise.

4 Active Noise Control System Performance

Sound measurements performed around the perimeter of the plant typically indicate an 11 dB reduction (as high as 17 dB measured) at 23.8 Hz as per Figure 2. This is about twice the reduction possible through the traditional methods previously mentioned. The AERCOUSTICS active noise cancellation system has been operating reliably since August, 1994, and there have been no noise complaints since its installation.

5 References

1. Nelson, P.A., and Elliott, S.J., "Active Control of Sound" *Academic Press Limited*, 1993.

NOTES: This project won the 1995 Award of Excellence as selected by the Association of Consulting Engineers of Canada and the Canadian Consulting Engineer Magazine.

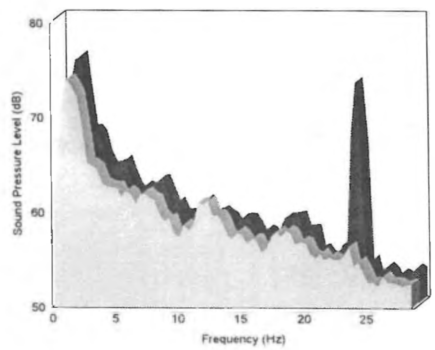

FIGURE 2: Narrowband spectra illustrating the reduction of the 23.8 Hz low frequency pure tone by up to 17 dB within the residential community.

A COMPARISON OF TWO TEST METHODS FOR MEASURING YOUNG'S MODULUS OF BUILDING MATERIALS.

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1. Introduction

For building noise control problems, solid materials such as sheet metal, wood, gypsum board, and oriented-strand-board (OSB) are frequently used. These materials are characterized by the following parameters: density, Young's modulus, Poisson's ratio, and the loss factor. The current ASTM standards for the measurements of Young's modulus are designed specifically for ceramics, bricks, carbon and graphite materials, and concrete[1]. The techniques used are based on measuring either the longitudinal wave velocity or the resonant frequency of the flexural mode of vibration of a specimen of the material. These techniques have been applied to the bar shaped specimen of some building materials.

2. Longitudinal wave velocity method

The technique used was an adaptation of the procedure specified in the ASTM C 769 - 80 standard. Two accelerometers were mounted one at each end of a 1.3 m long bar shaped specimen of a building material. An impulse was generated at one end of the bar by tapping it with a stick. The signals from the two accelerometers were captured simultaneously with a two channel A/D board operating at 50 kHz sampling rate. Knowing the exact length of the bar and the transit time of the impulse between its two ends, the longitudinal wave speed was computed. The Young's mudulus (E) was then calculated from the wave speed (C) and the density (ρ) of the material according to the following formula,

$$E = \rho\ C^2$$

3. Sonic resonance method

The method used was an adaptation of the procedure specified in the ASTM C 747 - 93 standard. The 1.3 m long bar shaped specimen was freely supported at its transverse fundamental nodal points (0.224 L from each end) using two slings. A light weight accelerometer was placed at one end of the bar to pick up the transverse vibration of the bar when it was tapped at the other end by the rubber head of a pencil. The signal was captured by an A/D board. The resonant frequency (f) of the fundamental flexural mode of vibration was determined from the frequency analysis of the captured time signal using FFT. The Young's modulus was then determined according to the following equation,

$$E = k\ M\ f^2$$

where M is the mass of the specimen and k is a constant which can be obtained from Table 8 of Ref 2.

4. Conclusions

Based on a number of measurements using different building materials, the following conclusions can be drawn. (1) Either methods can be used. (2) The sonic resonance method is better than the wave velocity method. It requires only one detector, where as the wave velocity method requires two and needs careful calibration for zero-time correction between the two detectors. Results also indicate that the sonic resonance method has better repeatability.

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MEASUREMENT OF TYPICAL SPEECH AND BACKGROUND-NOISE LEVELS IN UNIVERSITY CLASSROOMS DURING LECTURES

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Introduction

As part of a study on classroom acoustics [1], an experimental procedure has been developed for determining typical long-term speech and background-noise levels in university classrooms during lectures. A particular objective was to determine typical levels of student-activity noise as separate from ventilation noise; previous work had suggested that it can be surprisingly high [1]. The procedure involved recording lectures, digitizing and filtering the recordings, and determining long-term sound-pressure-level distributions. The procedure has been applied to lectures in UBC classrooms.

Classroom Recordings

A total of 18 lectures were recorded at up to four positions in 14 classrooms. The classrooms were seminar and lecture rooms which contained 10-291 seats and 7-290 students, and with unoccupied mid-frequency reverberation times that varied from 0.5-1.8 s. Both male and female instructors - teaching a wide variety of subjects at various academic levels - were involved. Recordings were made using calibrated Bruel & Kjaer 4145 microphones and a TASCAM DA30 Mk II digital tape recorder. Data were collected for 10-15 min - in three blocks at the beginning, middle and end of the class - at each position. As a cross check, ventilation-noise levels were measured at each test position in each classroom on a separate occasion using a sound-level meter.

Analysis Procedure

Recordings at each position in each class were digitized and filtered using digital 63-8000 Hz octave-band and A-weighting filters. Software processed the resulting pressure time histories as follows: the signals were squared; short-term (using intervals which decreased over the test frequency range from about 160 to about 1 ms) mean-square pressures were calculated; interval sound-pressure levels were calculated; sound-pressure-level frequency distributions were determined and plotted; the resulting distributions were best-fit by one, two or three normal-distribution curves in an attempt to identify peaks. Only the A-weighted results have been analyzed to date, and will be discussed here.

Results

Attempts to best-fit one or two normal-distribution curves to the results generally gave a poor fit; however, three curves usually gave a very good fit. These were identified with three components of classroom sound - speech signal, ventilation noise and student-activity noise. However, in most cases it was not immediately obvious which, if any, of the three best-fit levels corresponded to which sound component - nor what was the correct level order.

In order to cast some light on this, levels were compared with ventilation-noise levels measured independently, and with typical long-term speech levels published in the literature [2]. Regarding ventilation noise, the agreement was generally poor - suggesting that such levels vary significantly with time - but indicated that the lowest of the three levels was likely that due the ventilation. The highest of the three noise levels generally corresponded fairly well to expected speech levels. The middle level was, therefore, usually associated with student-activity noise.

Fig. 1 shows a typical A-weighted level-distribution curve. It apparently has two main peaks - at 41 and 52 dBA according to the best-fit results - apparently corresponding to long-term ventilation and speech levels, respectively, as discussed above. The best-fit procedure identified a third peak - which is not obvious in Fig. 1 - at 43 dB. In a few cases the frequency distribution appeared to have only a single peak, indicating low signal-to-noise ratio, and making identification of component levels difficult. Following is a summary of the results:

Background noise - Ventilation-system noise levels varied from 33-46 dBA. Student-activity noise levels varied from 38-49 dBA. Total background-noise levels varied from 39-50 dBA.

Speech level - Speech levels at individual positions varied from 41-60 dBA. Room-averaged levels varied from 47-57 dBA with an average value of 53 dBA. Differences between male/female speakers and front/back seat positions altered levels by ±1.2 and ±1.5 dBA, respectively. Speech levels did not vary significantly with room acoustical conditions. It can be hypothesized that instructors adjust their voice levels to compensate for the acoustical conditions and maintain a constant speech level. To test this, speech levels and diffuse-field theory were used to estimate long-term instructor sound-power levels. These varied from 56-68 dBA with male/female differences varying this value by ±1.2 dBA, respectively. There was a strong correlation with room size - average speech power levels of 61 dBA (<30 seats), 62.5 dBA (30-80 seats), 64 dBA (>80 seats) - supporting the hypothesis.

Signal-to-noise ratio - It is speech-signal to total-background-noise ratio which determines speech intelligibility. Thus, for each position in each classroom ventilation- and student-activity-noise levels were added energetically and the results subtracted from the speech level. The resulting individual-position signal-to-noise ratios varied from 2-13 dBA. This was in all cases lower than the signal-to-noise ratio of about 15 dBA considered necessary for excellent speech intelligibility for normal-hearing adults [3].

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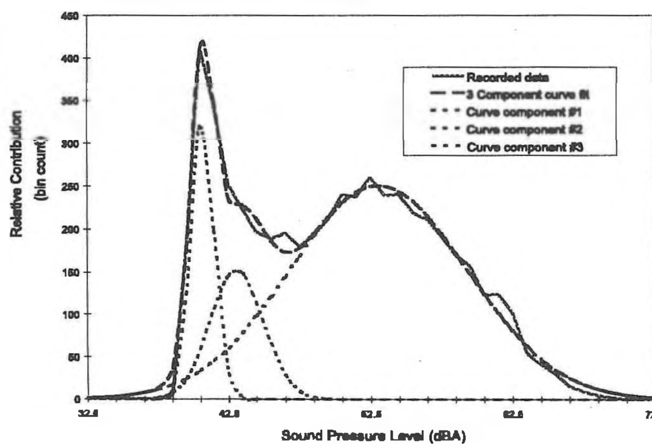

Fig. 1. A-weighted long-term sound-pressure-level distribution curve at a front seat in a 84-seat classroom with a male instructor.

IN SITU MEASUREMENT OF THE SURFACE IMPEDANCE OF FOAM AT OBLIQUE INCIDENCE USING PSEUDO-RANDOM SEQUENCES AND A SINGLE MICROPHONE

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Introduction

The surface impedance of materials can be determined experimentally in the free field or in rooms (*in situ*) from the technique of two- or single-microphone transfer functions. The purpose of this article is to extend the use of the pseudo-random sequence and a single microphone to ***in situ*** impedance measurement of materials at oblique incidence. First an experimental technique for measuring the acoustical impedances of materials is described. Then spherical-wave surface impedance of materials are estimated using the relationship between pressures at two field positions. Finally the experimental results are presented.

Experimental techniques

A sketch of the experimental set-up is shown in Fig. 1. The acoustical impedance at midpoint M is expressed by two transfer functions H_1 and H_2 measured sequentially at M_1 and M_2 (time dependent factor $e^{-i\omega t}$ is used here):

$$\tilde{Z}_M = \frac{p}{u} = \frac{i\omega\rho a}{2}\frac{\bar{H}_1 + \bar{H}_2}{\bar{H}_1 - \bar{H}_2}, \tag{1}$$

where ρ is the air density, ω is the angular frequency, $\bar{H}_{1,2} = (1/N)\sum_n^N H_{1,2_n}$. A digital pseudo-random signal called a Maximum Length Sequence (MLS) is used in the measurements. The measurement system was validated using the measured residual pressure-intensity index

Experimental Procedure, data acquisition and processing

The sound source was a loudspeaker in a wooden box enclosure driven by the pseudo-random sequence generator of the MLSSA system. The loudspeaker was supported by a stand which allowed the angle of incidence θ_0, and the distance from the source to the surface of the material r_0, to be varied. A microphone was mounted on a support allowing the position relative to the surface of the test sample to be adjusted with a precision of 0.025 cm. The test material was an industrial CONAFLEX foam ($1.2 \times 1.2 \times 5$ m^3). The test sheet was backed by an acoustically hard plane boundary. The measurements were made for different angles of incidence in a semi-reverberant room and in an anechoic chamber. Surface impedances Z_s and reflection coefficients $R(\theta)$ were calculated from $\tilde{Z}_M$ using the following relationships,

$$Z_s = \frac{\rho c}{\cos\theta_0}\,\frac{1 + R(\theta_0)}{1 - R(\theta_0)}\,\frac{1}{1 + i/kr_0}, \tag{2a}$$

$$R(\theta_0) = \frac{-i\omega\rho a A/2 + B\tilde{Z}_M}{C\tilde{Z}_M + i\omega\rho a D/2}, \tag{2b}$$

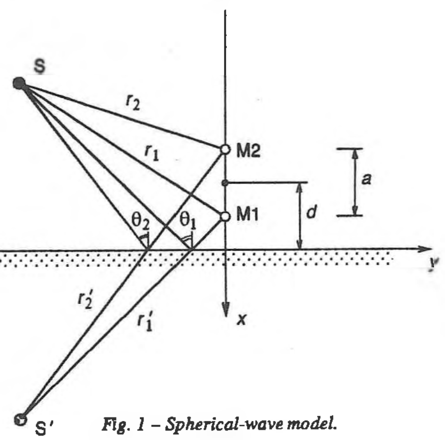

Fig. 1 – Spherical-wave model.

where $A = T_1 + T_2$, $B = T_1 - T_2$, $C = T_2' - T_1'$ and $D = T_1' + T_2'$ with $T_1 = e^{ikr_1}/r_1$, $T_1' = e^{ikr_1'}/r_1'$ for example.

Experimental results and analysis

Fig. 2 shows the absorption coefficients $(1 - |R|^2)$ at $\theta = 0°$ measured in the semi-reverberant room and those measured in an anechoic room. It is noted that the values measured in the two environments have good agreement in the frequency range 300 Hz to 7000 Hz. However there are big discrepancies at low frequencies below 300 Hz because of systematic errors. Fig. 3 shows the measured surface impedance $Z_s/\rho c$ versus angle of incidence at given frequencies with $r_0 = 31$ cm. It is shown that the surface impedance of foam is a function of the angle of incidence.

Conclusion

Measurements were carried out on a CONAFLEX foam at oblique incidence. The variations of the surface impedance with the angle of incidence demonstrate that the foam is of local reaction at the test frequencies. The comparison of results of experiments in a semi-reverberant room and in an anechoic room shows that the acoustical properties of materials can be measured in an arbitrary room using a pseudo-random sequence and a single microphone.

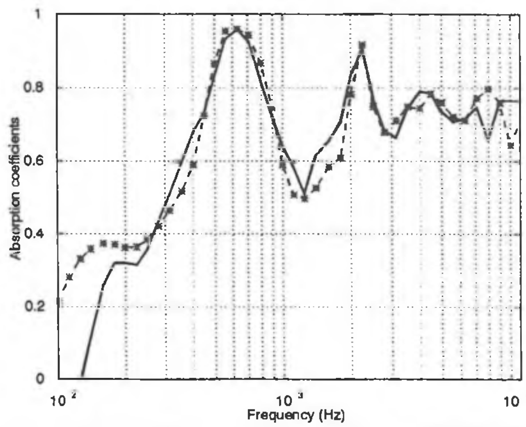

Fig. 2 Surface absorption coefficient of foam measured in an anechoic room (———) and in a non-anechoic room (* **).*

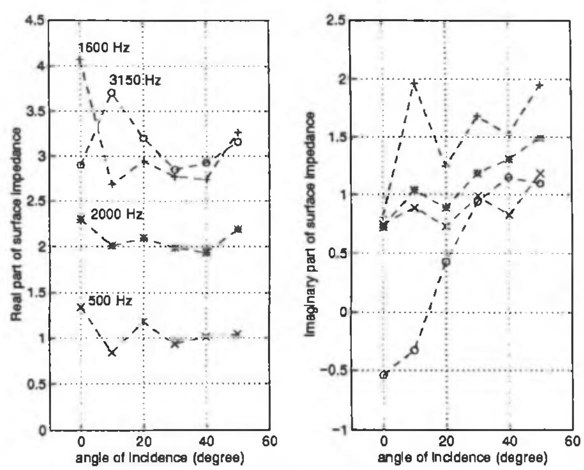

Fig. 3 – Surface impedance $Z_s/\rho c$ of CONAFLEX foam with a thickness of 5 cm at oblique incidence.

USE OF RESIDUAL PRESSURE-INTENSITY INDEX FOR EXAMINATION OF THE ACCURACY OF A SURFACE-IMPEDANCE MEASUREMENT SYSTEM

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Introduction

The surface impedance of materials can be determined experimentally in the free field or *in situ* by measuring sequentially the transfer function at two locations using a single microphone if the excitation signal can be controlled [1]. However as no experimental system is perfectly linear and repeatable, errors will be introduced. The objective of this work was to verify experimentally the measurement system of surface impedance using the residual pressure-intensity index. The method presented here can be used to examine the stationarity and repeatability of an experimental system.

Expression for residual pressure-intensity index

The biased impedance $\hat{Z}$ can be expressed as a function of equivalent phase errors defined in terms of the characteristics of microphones [2] . In the configuration of measurements adopted here, the microphone is oriented parallel to the surface of the material and the measurement is little sensitive to the vent effect in reactive field. The biased impedance $\hat{Z}$ depends only on the equivalent phase error ϕ_e between the signals measured at two positions:

$$\hat{Z} = \frac{Z \pm |Z|^2/K_0}{1 \pm 2\,\mathrm{Re}\{Z\}/K_0},\quad K_0 = |p_0|^2/2\rho c\,|I_0| \simeq ka/|\phi_e|. \tag{1}$$

The residual pressure-intensity index is given as

$$L_{K_0} = 10 \log K_0 \simeq 10 \log ka/|\phi_e| \qquad \text{(dB)}. \tag{2}$$

As only one channel is used in the impedance measurements, the equivalent phase error here does not include the electronic phase mis-match. However if the repeatability properties of a system are not perfect (non-linear effects for example), equivalent phase error will be introduced. This error can be represented as the difference in phases between two average transfer functions $\bar{H}_1$ and $\bar{H}_2$, which are determined using N time acquisitions without changing the position of the microphone $\phi_e = \mathrm{Arg}\{\bar{H}_1 \bar{H}_2^*\} = \bar{\varphi}_1 - \bar{\varphi}_2$, $\bar{\varphi}_1$ and $\bar{\varphi}_2$ are the phases of $\bar{H}_1$ and $\bar{H}_2$ respectively (for $a = 0$). As $\bar{\varphi}_1$ and $\bar{\varphi}_2$ are random variables and have Gaussian distributions, it is natural to determine ϕ_e by calculating the standard deviation of the phase of the transfer function φ. Since the two sets of measurements are independent of each other, $\mathrm{Cov}[\bar{\varphi}_1, \bar{\varphi}_2] = 0$, $\mathrm{Var}[\bar{\varphi}_1 - \bar{\varphi}_2] = 2\sigma^2[\varphi]/N$. The equivalent phase error can be defined statistically from the variance of $\bar{\varphi}_1 - \bar{\varphi}_2$ by $|\phi_e| \approx \sqrt{\frac{2}{N}}\,\sigma[\varphi]$. The standard deviation of the phase of the transfer function is estimated by $\sigma[\varphi] \simeq \sqrt{\frac{1}{M-1}\sum_{m=1}^{M}(\varphi_m - \bar{\varphi})^2}$, where $\bar{\varphi} = \mathrm{Arg}\{\bar{H}\}$ with $\bar{H} = (1/M)\sum_{m=1}^{M} H_m$. The residual pressure-intensity index becomes

$$L_{K_0} \simeq 10 \log \sqrt{\frac{N}{2}}\,\frac{ka}{\sigma[\varphi]} \qquad \text{(dB)}. \tag{3}$$

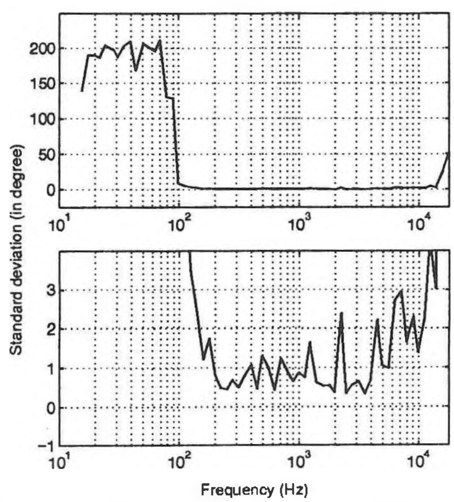

Fig.1 – Standard deviation $\sigma[\varphi]$ (in degree) estimated using 150 experimental samples $H_m(m = 1, 2, 3 \ldots 150)$.

Eq. (2) shows that the residual pressure-intensity index L_{K0} is dependent on the standard deviation of the phase of the measured transfer function, the separation distance of the microphones, frequency and the number of acquisitions.

Verification of experimental system using L_{K0}

In order to determine experimentally the variance $\sigma^2[\varphi]$ of the phase of the transfer function, a series of transfer functions H_m (m=1,2,3, ... M) were measured using a single microphone at the same position in the sound field. The measurements were done in a semi-reverberant room. The glassfiber sheet was put on the floor of the room. A microphone was placed 10 cm from the surface of the material. A loud-speaker used in the impedance measurement was driven by a determinist broad-band signal. Fig. 1 shows the standard deviation $\sigma[\varphi]$ estimated from 150 measurement samples. It is noted that $\sigma[\varphi] > 50°$ at frequencies $f < 100$ Hz. The measured values L_{K_0} were calculated using $\sigma[\varphi]$ in the case of $a = 10$ cm and $a = 0.5$ cm respectively. The results are presented in Fig. 2, for $N = 32$. It is shown that for $a = 10$ cm, $L_{K0} > 20$ dB at frequency range $300 < f < 20000$ Hz. Hereas when $a = 0.5$ cm $L_{K0} > 20$ dB in the frequency range $2 < f < 20$ kHz.

Conclusion

A technique using pseudo-random sequences and a single microphone for *in situ* measurement of the acoustical properties of materials was validated experimentally by measuring the residual pressure-intensity index L_{K0}. This verification allows us to obtain the impedance using a single microphone without phase-mismatch errors as in the two microphone method. It was shown that this technique has frequency-range limitations determined by the distance between the two microphone positions.

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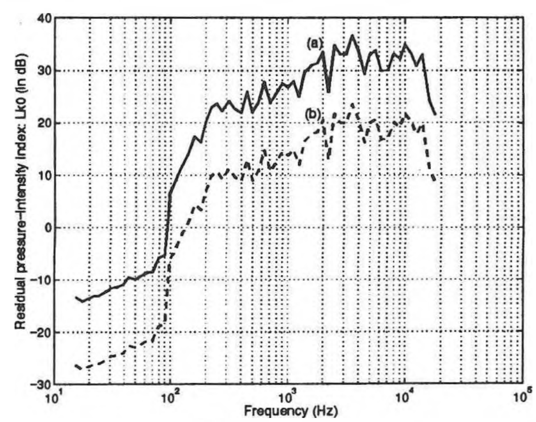

Fig.2 – Variation of L_{K_0} (in dB) with the frequency for two distances between microphones (a) $a = 10$ cm; (b) $a = 0.5$ cm.

Auralization of Speech-Communication Cues

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Introduction

Auralization is becoming a common technique in simulating different acoustical environments with laboratory control [1]. Furthermore, auralization provides a means of presenting binaural stimuli to a listener through some transducer. This study utilizes an auralization system to acoustically simulate a small test chamber (17.42'x12.92'x8.83') and present binaural speech babble to a listener as heard from this simulated room.

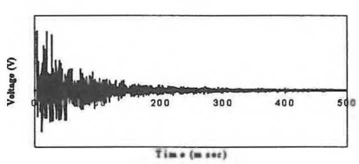
The room was simulated using a commercial auralization hardware and software package from Tucker-Davis Technologies (TDT). This system automatically calculates the direct sound and first-order reflections, assuming all walls behave as 100% reflective. Further components were supplemented to the software to increase the accuracy of the simulation. These components include the implementation of an approximate reverberant tail, and wall absorption filters.

Reverberant Decay

An approximate reverberant tail was created to represent the higher-order reflections that the TDT system ignored. To create this reverberant tail, the reverberant decay curve of a small test chamber was measured using the MLSSA system V9.0. From the decay curve, the reverberation times (RT) per octave bands were calculated. Table 1 presents the RT values for the different octave bands.

Table 1: Averaged Measured Reverberation Times for the Small Chamber per Octave Band

Octave Bands (kHz)	.125	.250	.500	1.00	2.00	4.00	8.00
RT (sec)	0.72	0.94	0.85	0.56	0.55	0.48	0.49



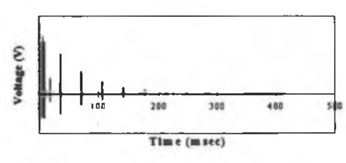


Figure 1a and b The measured and simulated impulse response for the small test room.

For simplicity purposes, the RT value of 0.6 seconds was chosen as the average RT over all octave bands accounting for the power in the low frequencies and the speech intelligibility contributed by the high frequencies. Using this RT value, the reverberant tail was made. The reverberant tail approximates an exponential decay curve using the decay processors built in to the TDT system. The set up consists of a feedback loop with one decay time variable, x, and one attenuation constant programmed on the delay processors. This particular arrangement creates a series of pulses separated by time x. Figure 1 displays both the measured test room impulse response and the simulated test room impulse response obtained by the TDT system.

Surface Absorption

In calculating the first-order reflections to simulate the wall properties, speech signals were convolved with several finite impulse responses (FIR) that describe the relative reflective characteristics of the surfaces of the room. The values for these characteristics were obtained using a room-prediction technique, which involves manipulating the absorption values for each surface until it yields the proper RT for the room. Table 2 shows a list of the calculated absorption coefficients per octave band for the different surfaces in the room.

Table 2: Absorption Coefficients per Octave Bands for the Room's Surfaces

Surface	.125 kHz	.250 kHz	.500 kHz	1.00 kHz	2.00 kHz	4.00 kHz	8.00 kHz
Floor	0.02	0.02	0.01	0.03	0.03	0.04	0.04
Ceiling	0.35	0.22	0.23	0.30	0.35	0.50	0.55
Wall	0.09	0.06	0.04	0.05	0.07	0.07	0.07

Simulation Validation

Once the room was simulated, the localization perception of listeners in the real and virtual environments were compared to validate the sound-field simulations [2]. In the real room, localization was accurate with feedback; without feedback, front-back confusions occurred. The performance of the localization tests in the virtual test room were comparable to the performance in the real room.

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AN ACOUSTICAL COMPARISON OF TWO ATRIUM SPACES

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Introduction

This paper reports the results of comparisons of the acoustical conditions in two architecturally different atrium spaces. The East atrium is a long and narrow atrium with open plan office areas opening directly onto the three levels of the atrium space. The Small atrium has a small octagonal plan, is three floors high and serves as an entrance to a large office complex. These studies were part of a larger project to examine various aspects of atria and included acoustical tests in 10 atrium spaces.

Acoustical impulse response measurements were made in each atrium using an approximately omni-directional source and a maximum length sequence signal. Various acoustical measures including decay times and sound levels were obtained from the impulse responses.

The East Atrium

Figure 1 compares average early decay times, EDT, and reverberation times, RT in the East atrium. The results are characteristic of many atrium spaces with the largest decay times at mid-frequencies. They decrease at lower frequencies due to the sound absorption of the glass and other materials, and at higher frequencies due to air absorption. EDT values are shorter than RT values indicating that the decay is not completely linear and that the sound field is not very diffuse.

Average values of the measured early-arriving and late-arriving relative sound levels showed the early energy to be several decibels stronger. This suggests that if background sound levels are low there will be reasonably good speech intelligibility within this space, which was confirmed from measures of speech intelligibility.

Figure 2 shows relative sound levels, G, at 1000 Hz versus source-receiver distance in this atrium. For comparison, the predictions of simple diffuse field theory and Barron's revised theory are also shown. At distances up to about 30 m there is some agreement with Barron's theory but with considerable scatter because some positions were partially screened. At larger distances, measured values consistently fall below predictions. These results are different than those in large auditoria because of the larger distances and more similar to measurements in industrial spaces.

The Small Atrium

The average measured decay times in the Small atrium peak at 250 and 500 Hz. This was because the medium and higher frequency decay times were reduced because of the presence of a large area of porous sound absorbing material.

Measured early relative sound levels were again a few decibels higher than later arriving sound levels. However, the total relative sound levels were 8 or 9 decibels higher in this smaller atrium.

The propagation of sound within the Small atrium was quite different than in the East atrium (Figure 2). For sources and receivers on the main floor, the variation of sound levels with distance was similar to simple diffuse field theory. For propagation to receivers on the upper two floors, sound levels decreased at approximately 6 dB per doubling of distance. This suggests that the direct sound is dominant due to the large areas of sound absorbing material on the balcony faces.

Conclusion

The common reverberant characteristics of atrium spaces are due to the low frequency absorption of the large areas of glass. The propagation of sound within atrium spaces depends on the geometry of the atrium and the placement of sound absorbing materials.

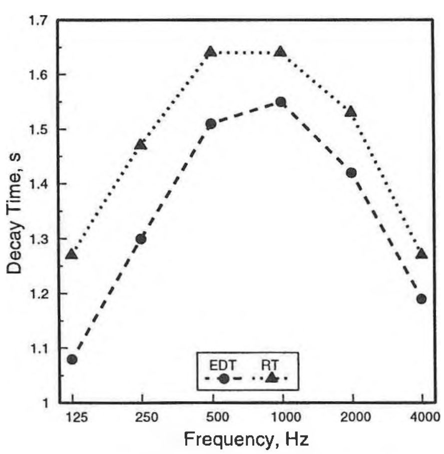

Fig. 1. Average decay times in the East atrium

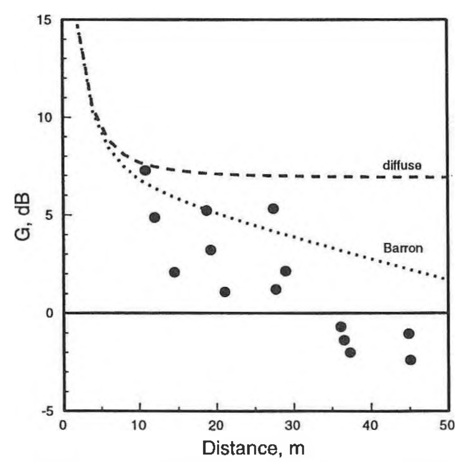

Fig. 2. Relative sound levels versus distance, East atrium.

IMPACT OF ACOUSTIC CRITERIA ON HVAC SYSTEM DESIGN FOR THE FRANCIS WINSPEAR CENTRE FOR MUSIC EDMONTON, ALBERTA

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The Francis Winspear Centre for Music will provide concert audiences with world class acoustic performance. The project construction cost will be 30 million dollars, and is scheduled to be complete in September, 1997.

The acoustic criteria for this project has substantial impact on the heating, ventilation and air conditioning systems design.

Heating systems for the theatre space were selected to ensure hydronic noise does not impact in the theatre zone.

Central plant equipment location was carefully positioned for noise to have the least possible impact through structural transmission.

Rotating equipment, such as fans and pumps, were selected with minimum noise generation criteria.

Air supply systems to the theatre space were designed with low air velocities and with careful consideration to air distribution patterns to avoid cold drafts, and to provide a comfortable environment for the audience and performers. Air distribution throughout the theatre is governed by sound trapped slot entry points into void spaces behind seating racks, and transferred through large collection plenums back to central plant systems.

Special systems were provided to cool high heat generation spaces, such as spot follow rooms and electronic equipment rooms.

Vibration isolation was provided for equipment, piping and ductwork.

Acoustic provisions were made for a unique problem of removing hydrocarbon vapours from contaminated soils.

In summary, the heating, ventilating and air conditioning systems design was carefully fashioned to respect the noise criteria performance that was established for the facility.

Speech Intelligibility Workshop - Acoustics Week in Canada 1996

At Acoustics Week in Canada 1996 in Calgary a speech-intelligibility discussion group will be convened to discuss the delivery of speech intelligibility in the architectural environment. This will include unamplified and reinforced speech in public-assembly spaces such as churches, schools, lecture theatres, drama theatres etc. In addition, speech-intelligibility delivery in life-safety and critical public-address functions - such as voice warning and fire page systems, swimming pools, airports, train stations, industrial plants, prisons, and others - will be discussed.

Participants will be invited to present background papers and case studies as bases for the discussion. The ultimate goal of the discussion group will be to determine what possibilities may exist for enshrining a minimum requirement for speech intelligibility in building regulations governing life safety, and public-assembly spaces where delivery of information through the spoken word is considered a prime requirement of the space. This new regulation would offer a parallel requirement to the barrier-free-access regulations outlining the need for a listening-assistance systems for the hearing impaired, and would finally establish a minimum guideline for the members of the audience with normal hearing.

At this preliminary discussion-group meeting, the possibility of establishing a new intelligibility standards committee will be discussed. Further reports will follow in a future issue of the CAA journal.

Barry McKinnon

ACOUSTIC EFFECTS OF VOCAL WARM-UPS

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1. Introduction

Singers, like others who undergo heavy physical exertion, benefit from a warm-up or light exercise prior to full use of the muscles involved in the exertion. The mechanics of vocal warm-ups, however, are not thoroughly understood. People who use vocal warm-ups know that they work but their scientific study has been somewhat overlooked.

The physiological responses to sports warm-ups are well documented [1] and it may be assumed that vocal warm-ups trigger similar responses: an increase in tissue temperature; increased blood circulation, etc. The purpose of this study is to examine whether these physiological changes affect the acoustic output.

2. Singer's Formant

The singer's formant is a peak of energy near 3 kHz found in singers' voices (except soprani) which helps project the voice. This is clearly seen as a wide band in the spectrogram [2,3]. As the voice warms up and settles into the "ideal" mode of production, the singer's formant should become more focused and the bandwidth of the singer's formant is expected to become narrower.

3. F_0 Range

Singers use a wide range of fundamental frequencies and this is generally considered to increase with a warm-up. The scattergram allows measurements of the extreme range as well as the range of controlled production. The extreme range is measured by means of a glissando or slide to the outer limits of the singer's range. The range of controlled production (the frequencies which can be sustained by the singer with "good vocal technique") is measured -- also by scattergram -- within the context of ascending and descending scales.

4. Average Energy

In musical terms, this section looks at control over dynamics; in acoustic terms, we are looking at energy. To measure the acoustic side of this, measurements of the singer's performance at three amplitudes (relative to each other) will be taken at three different F_0s. Measurements taken after the warm-up are expected to show a greater dynamic range.

5. EGG

The electroglottogram produces a glottal waveform by means of electrical impedance. The velocity of the closing phase is expected to increase as a result of the vocal warm-up.

6. Discussion

The singer's warm-up is designed to increase the efficiency of voice production. That efficiency increase should lead to observable changes in the acoustic output -- the most readily accessible aspect of the singer's production. The singer's formant is expected to narrow in bandwidth; the frequency and energy ranges are both expected to increase; and the glottal waveform should become more regular and "efficient". All this will help the singer perform more safely and more comfortably.

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PALATOMETRIC PATTERNS IN SPEAKERS WITH REPAIRED CLEFT PALATES OR SEVERE HEARING IMPAIRMENT

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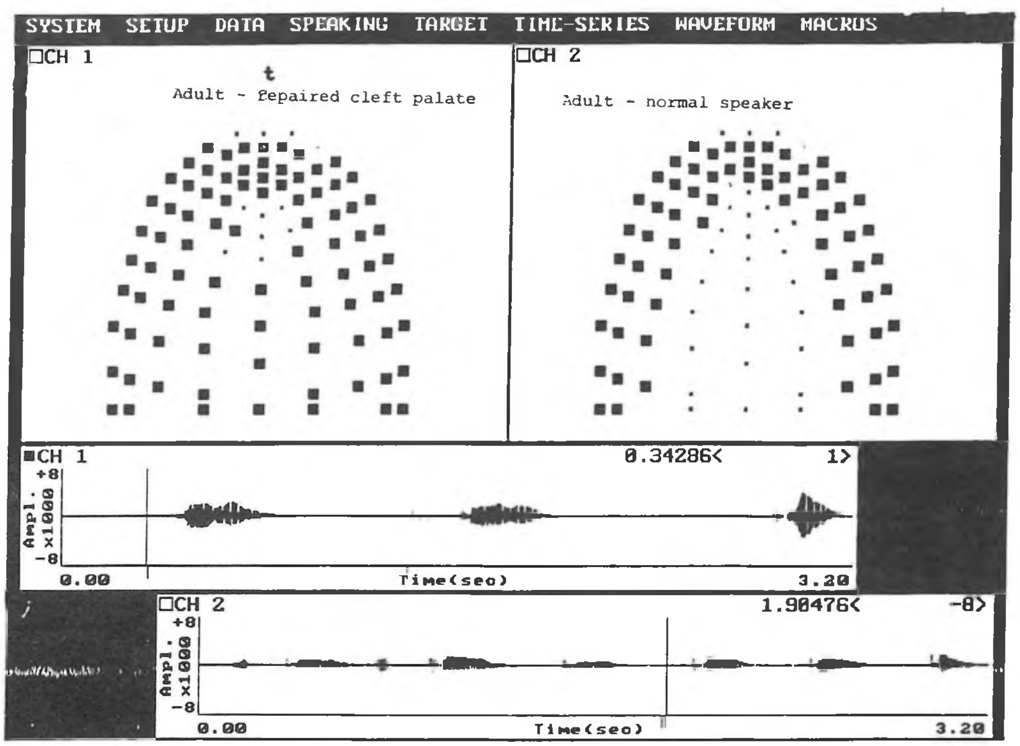
Palatometry (electropalatography) provides visual information about tongue-palatal contact during speech production. Subjects wear custom-designed artificial palates with up to 96 electrodes, and dynamic or static computer displays show tongue contact patterns. A pilot speech therapy project is currently being conducted at UBC with five adults and six children who have a variety of impairments or combinations of impairments: repaired cleft palate, sensorineural hearing loss, and/or motor speech disorders. In this symposium we will focus our discussion on similarities and differences between the speech of two adults (one with a profound hearing loss, and one with a repaired cleft palate), and two children (one with a profound hearing loss and a cochlear implant, and one with a hearing loss and partially repaired cleft palate). At assessment, all subjects tended to have the most difficulty with alveolar and palatoalveolar phones. Discussion will focus primarily on these speech sounds.

Data presented comes from assessment, treatment, and followup speech samples. Assessment and followup probes involved taperecordings and narrow phonetic transcriptions of about 100 words from single word, sentence, and paragraph elicitations, plus palatometric tracings of about 50 single words. The followup probes were conducted following each of two 8-session treatment blocks (of about one month each). At the end of each treatment session, a palatometric tracing of the best final production was also stored, allowing for longitudinal data collection over the 8-10-week period.

At assessment, tongue placement was typically retracted for subjects in this group. Sibilants, if produced at all, had lateral release or no/abnormal grooving. As the clients learned to use the visual feedback to change tongue placement, normalization of speech occurred over time.

Both palatometric and perceptual data (including subject self-ratings) will be presented. Comparisons will be made across subjects and with data from a normal adult speaker. The following figure shows production of [t] by a normal speaker (right of figure) and someone with a repaired cleft palate.



The Effects of Speaking Rate on the Comprehensibility of Native and Foreign-accented Speech

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1. Introduction

This study assessed the effects of speaking rate on local and global aspects of second language (L2) speech production. Since previous research has suggested that a reduced speaking rate may improve the intelligibility of L2 speech, it was proposed (Hypothesis 1) that the difficulty L2 learners often have in producing accurate phonetic targets is due in part to insufficient time to execute good productions. If so, learners might show more intelligible vowel targets when speaking at a rate slower than normal. The second proposal (Hypothesis 2) was that the perceived global comprehensibility and accentedness of L2 connected speech may improve at a slower speaking rate. To test these hypotheses, we carried out two experiments in which native English listeners identified and evaluated productions of L2 learners.

2. Methods and Results

Speakers. Recordings were made of 10 high-proficiency adult native speakers of Mandarin (5 female, 5 male). All had been born in China and had moved to Canada after the age of 21. Informal evaluation indicated that their accents ranged from moderate to strong. A comparison group of 10 native speakers of Canadian English was also recruited.

Recordings. Two types of speech samples, front vowel productions and a short reading passage, were collected from each speaker. The vowels were elicited in bVt words ("beat," "bit," "bait," "bet," "bat") in the carrier sentence, "Now I say ____." The speakers read the carrier sentences first at a normal speaking rate and then at a rate one half as fast as normal. The individual bVt tokens were digitized, the final [t] portion was removed from each token, and a randomized presentation tape was prepared. An ANOVA indicated that the CVs produced in the slow condition were significantly longer ($p<.05$) than the CVs produced in the normal condition (208.0 msec vs. 192.9 msec respectively). The effect was uniform across vowels.

The reading passage was also recorded at two speaking rates. From each production, the first two sentences (18 words) were digitized, and a second randomized tape was prepared. Duration measurements indicated that, while both groups slowed down significantly ($p<.05$), the English speakers produced the slow passages at a rate 69% of normal while the Mandarin speakers produced them at a rate 76% of normal.

Vowel Identification Task. Thirty-eight native English listeners listened to the stimulus tape, and circled which of the five words they heard on a response sheet. ANOVAs on the vowel identification data (%-correct ID) for each listener indicated that overall, the Mandarin speakers produced less intelligible tokens of all words except "bait." No effect of speaking rate was observed on the identification rates for "beat," "bait," and "bat." However, the Mandarin speakers' productions of "bit" and "bet" were correctly identified at significantly higher rates in the slow condition than in the normal condition. Thus, Hypothesis 1 was partially confirmed.

Rating Task (Reading Passage). Seventeen native English listeners evaluated the passages on two 9-point scales: comprehensibility ("1" = very easy to understand, "9" = very difficult to understand) and accentedness ("1" = no accent, "9" = very strong accent). An analysis of mean ratings for each listener revealed that the Mandarin speakers' utterances were judged to be less comprehensible and more accented overall than the utterances of the native English speakers. There was no overall effect of speaking rate on comprehensibility for either group of talkers. However, the Mandarin speakers' passages were judged to be significantly more accented in the slow condition than in the normal condition ($M = 6.1$ vs. $M = 5.8$, respectively). Therefore, Hypothesis 2 was not supported and was, in fact, partially contradicted.

3. Conclusions

While these results indicate that the intelligibility of local phonetic targets may sometimes improve when L2 learners articulate slowly, they do not support the proposal that slowing down improves global comprehensibility. In fact, the passages produced at the slow rate in this study were judged to be more accented than those produced at the normal rate. It appears then that, in the slow reading passages, any improvement in the production of phonetic targets was either not detected by the listeners or was offset by some other factor, such as reduced fluency. It is clear that research of this type needs to be extended to cover other types of speech samples and other types of rate adjustments.

AUDITORY CHANGE WITH PITCH AND FIBREOPTIC FILMING OF ARYEPIGLOTTIC ARTICULATIONS

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1. Introduction and Method

At issue is the distinction between "pharyngeal," "epiglottal" and "glottal" places of articulation, and the phonetic description of manners of articulation produced in the pharynx. Pharyngeals are characterized by a retraction of the tongue, bringing the epiglottis in close proximity to the back of the pharynx [1,2], but it is often difficult to discern visual details of manner of articulation to correlate with auditory and acoustic measures, because a deeper view is obscured by the epiglottis. The question remains: if pharyngeal stops are possible, how are they distinct from epiglottals and glottals; and is their production similar to the mechanism for producing fricatives and approximants?

Laryngoscopic images of the pharynx and larynx were obtained using a Kay Elemetrics Rhino-Laryngeal Stroboscope 9100 -- a computer-controlled system including a dual halogen (fixed) and xenon (strobe) light source, a Panasonic KS152 camera, a Mitsubishi S-VHS video cassette recorder BV-2000 and printer. In order to investigate the extent of view possible of the laryngeal and pharyngeal mechanisms behind the apex of the epiglottis during pharyngeal articulations, an Olympus ENF-P3 fibrescope was attached to the Kay system, using nasal insertion and a 28mm lens for wide-angle view. The subject in all nasendoscopic observations was the author, producing maximally contrastive phonetically controlled speech data. The view from the naso-pharynx was adjusted to peer behind the apex of the epiglottis as far as possible, filming each articulation in the environment of an [i] vowel (see [3,4]).

2. Pharyngeal Articulatory Categories

It appears under fibreoptic observation of the pharyngeal mechanism that the aryepiglottic folds and their related cartilages are the main, active "articulator" in the production of speech sounds in the category "pharyngeal," i.e., that "epiglottal" articulations can be treated as a category of pharyngeal manners of articulation. Evidence of trilling accompanying friction is presented; and the nature of the pharyngeal (epiglottal) stop is described. It is suggested that pharyngeal stop, fricative, approximant and trill share a common place of articulation, differing in manner of articulation; that "epiglottals" are not separate from pharyngeals in place of articulation; and that the behaviour of pharyngeals parallels the uvular series; as in Table 1.

Furthermore, all of these articulations may be produced with a raised or lowered larynx. Major differences in auditory/acoustic quality are achieved when the larynx as a whole is raised or lowered during the production of pharyngeals. Most salient differences in auditory quality are realized when pitch varies between high and low register.

These distinctions help to explain the phonetic variation in pharyngeal quality that occurs in Northwest Coast languages, /ʕ,ʕʷ/ and their "glottalized" counterparts /ʕ',ʕʷ'/ [5]; in Caucasian languages where the more constricted sound is labelled [ʜ] in contrast to [h] [6,7]; and in Mon-Khmer register tones involving larynx raising vs. lowering [7]. These larynx height settings, with their consequent acoustic effects, have been shown to be pitch-dependent [8].

Table 1. Pharyngeal Consonantal Distinctions (Place/Manner/Voiceless-Voiced)

[ʔ]	Glottal plosive
[h]	Voiceless glottal fricative
[ʡ]	Pharyngeal ("epiglottal") plosive ("massive glottal stop")
(ħ̞)	Voiceless pharyngeal approximant (not audible)
[ʕ̞]	Voiced pharyngeal ("epiglottal") approximant (the common value of /ʕ/)
[ħ]	Voiceless pharyngeal ("epiglottal") fricative
[ʜ]	Voiceless fricative with aryepiglottic trilling
(ʕ)	Voiced pharyngeal fricative ([ʕ̞, ʢ] more common)
[ʢ]	Voiced fricative with aryepiglottic trilling

Acknowledgements

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The Hierarchy of Timing Strategies in Connected Speech: Patterns and Consequences

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1. INTRODUCTION

The present study further explores the durational patterns observed in the spontaneous conversation of six speakers of Skolt Sámi (a Finno-Ugric language). The objective of the research reported on here is to examine the constraint of keeping characteristic durational ratios constant in relation to employing certain temporal strategies observed during the course of a controlled experiment. It will be argued that this constraint overrides timing strategies aimed at significantly reducing durations while keeping to a durational target in connected speech.

2. METHOD AND EXPERIMENT

The material analyzed here consists of recordings of the spontaneous conversation of six speakers of Skolt Sámi (all female). Only those sections of spontaneous conversation were considered in which certain durational targets (e.g. paragraphs) were identified. Durational measurements were made of the relevant segments within the disyllabics, and ratio values were analyzed with reference to those obtained by two of the speakers during the course of a series of controlled experiments. A total of 240 utterances was analyzed (84 and 56 respectively for the two speakers with whom the controlled experiments were conducted, and 15-30 utterances for each of the other four speakers). These utterances were selected from a much larger corpus on the basis of durations of disyllabics falling into the range of 177 to 302 msec (varying by morphological type). Those disyllabics with durations not falling into this range were excluded from the analysis in order to restrict variation in speech tempo. [1] The same types of disyllabics have durations between 550 and 720 msec in citation forms.

3. DISCUSSION AND RESULTS

The four timing strategies observed in larger grammatical units during the controlled experiment are as follows: (i) the employment (or not, as the case may be) of the optional vowel reduction or vowel drop rule, (ii) shorter word durations in paragraph-final sentences, (iii) shorter pause durations in paragraph-final sentences, and (iv) shorter absolute segment durations achieved by keeping the characteristic ratios constant. [2] While no particular hierarchy could be observed during the controlled experiments, it was noted that in the spontaneous conversations the speakers hardly ever resorted to the non-employing of vowel reduction or vowel drop option while keeping to a durational target. It is all the more interesting because this strategy appears to be consistently employed during the controlled experiment, and thus was considered as evidence pointing to the fact that speakers tend to conform to a durational target. In connected speech, shorter absolute word durations and shorter pause durations were consistently apparent in paragraph-final sentences. Because the relevant segments -- first syllabic vowel and the consonant(s) following -- also had considerably shorter durations in words occupying positions near paragraph boundaries, it was important to examine the consequences of significant decrease in absolute duration with regard to the segment ratios that have linguistic significance. A comparison of ratio values apparent in spontaneous conversation with those in the controlled experiment suggests that the tendency to maintain durational ratios appears to be an important goal even during spontaneous conversation. The durational measurements representing this tendency are summarized in Figure 1. These values confirm results of an earlier study using material produced by the two speakers referred to above.

4. CONCLUSIONS

The four timing strategies observed in the controlled experiment have a definite hierarchy in terms of their occurrence in connected speech. Acoustic analysis of durational patterns apparent in this speaking mode reveals that (i) the apparent constraint for maintaining characteristic durational ratios constant overrides the strategy aiming at significant durational decrease close to boundaries; (ii) only three of the four strategies attested in the controlled experiment were consistently employed, the option available for not employing the vowel reduction rule thus was not being utilized. The latter fact indicates that different temporal strategies may be associated with different speaking modes.

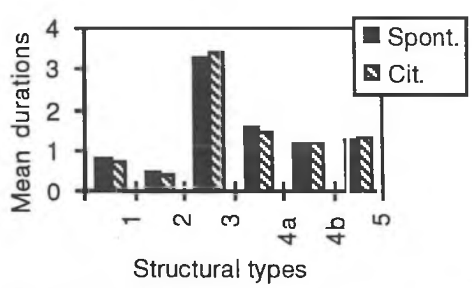

Figure 1. Durational ratios in two speaking modes

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NASALANCE IN SPEAKERS OF WESTERN CANADIAN ENGLISH AND FRENCH

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1. INTRODUCTION

Nasalance refers to the relative amount of oral and nasal acoustic energy in speech. It is affected by the amount of phonemic as well as assimilation nasality in an utterance. [1] Nasalance also has been shown to correlate with listener judgements of too little or too much nasality [2]. Thus its measurement has evolved as a clinical procedure that complements perceptual assessments of the resonance disorders known as hyper- and hyponasality. Disorders of resonance, particularly hypernasality, degrade speech intelligibility and may be present in a number of conditions, including neurological disorders and craniofacial birth defects. Resonance also may be affected by severe hearing loss or faulty sensorimotor learning patterns. To facilitate the valid and reliable use of nasalance data in the clinical practice of speech-language pathology, collaborative efforts among centres in North America have begun to accumulate language- and region-specific databases for nasalance in native speakers of English and French using standardized procedures, identical instrumentation and comparable spoken materials [3;4;5]. Healthy males and females aged 9-85+ have been included so that the effects of spoken-language history, regional dialect, sex and age on nasalance may be studied and appropriate normal ranges determined for clinical practice. *The nasalance data reported here were obtained from 458 normal speakers of western dialects of Canadian French and Canadian English.*

2. METHOD

The Nasometer 6200 (Kay Elemetrics Corp.) was used to record native speakers of western Canadian English or French as they read aloud three passages constructed for each language [3], one containing *no nasal phonemes* (#1, Table 1), one reflecting the *natural proportion of nasal phonemes* in everyday speech (#2, Table 1), and one *saturated with nasal phonemes* (#3, Table 1). Percent nasalance was computed according to the formula, [(nasal energy)/(oral+nasal energy)x100]. All participants rated their health as "good" at the time of recording, and were screened to exclude histories of neurological, respiratory, laryngeal or craniofacial disorders, sudden or congenital hearing loss and more than age-appropriate presbycusis. Subjects were recruited from schools, health units and seniors' centres in rural and urban communities in northern Alberta.

3. RESULTS

Results obtained thus far are reported in Table 1 for 306 western Canadian anglophones and 152 western Canadian francophones. The unbalanced nature of the data set renders interpretation of statistical analyses moot, especially for French, but some trends are detectable. Differences in nasalance are apparent in the data for children (9-13 yrs) and teens (14-19 yrs) versus those for adults; differences among nasalance values for young, middle-aged and old adults (20-44, 45-64 & 65-85 years) are not so noticeable. Small differences between nasalance scores for males and females are apparent in many of the cells.

Table 1: Mean % Nasalance (1SD) values for age, sex, passage and language.

ENG	Sex (#)	#1	#2	#3
9-13	M (27)	9.4 (3.2)	30.8 (4.2)	59.8 (5.9)
	F (31)	10.0 (2.8)	33.2 (3.7)	62.0 (5.2)
14-19	M (37)	10.8 (5.0)	32.9 (4.5)	62.1 (6.4)
	F (37)	10.7 (4.1)	34.6 (4.3)	63.0 (5.7)
20-44	M (31)	11.9 (6.0)	33.6 (6.0)	62.8 (7.4)
	F (31)	10.2 (3.4)	34.0 (3.2)	61.4 (5.0)
45-64	M (25)	12.7 (4.6)	33.9 (5.7)	62.0 (6.6)
	F (35)	12.9 (5.1)	35.3 (5.4)	63.7 (7.0)
65-85	M (21)	12.8 (4.7)	33.3 (5.7)	61.7 (6.6)
	F (31)	13.7 (5.0)	35.4 (5.6)	63.5 (7.6)
FRE	Sex (#)	#1	#2	#3
9-13	M (21)	9.4 (4.2)	24.2 (4.8)	33.1 (6.8)
	F (23)	8.5 (2.2)	24.8 (3.1)	34.3 (4.6)
14-19	M (6)	8.3 (3.6)	23.4 (3.0)	34.3 (2.6)
	F (9)	9.8 (2.4)	26.4 (4.4)	38.7 (6.5)
20-44	M (21)	13.9 (5.3)	28.3 (5.5)	38.6 (7.0)
	F (34)	14.7 (5.8)	30.2 (6.0)	40.5 (6.8)
45-64	M (7)	13.8 (5.5)	27.1 (4.1)	37.1 (5.7)
	F (17)	14.9 (5.1)	31.0 (5.4)	40.7 (6.7)
65-85	M (5)	10.5 (3.0)	24.6 (6.4)	32.0 (5.6)
	F (9)	12.7 (3.1)	27.1 (4.0)	37.2 (6.1)

#1=no nasals; #2=typical balance; #3=loaded with nasals. Passages 1, 2 and 3 for English are known as "Zoo," "Rainbow" and "Nasal sentences," respectively; the French counterparts are "La peur du tigre," "Le petit prince" and "Blanche Neige," respectively. [3]

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Perceptual and Acoustic Analysis of Word Initial Voicing Contrasts Across Speaker Age

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1. Introduction

There are several acoustic cues, including voice onset time (VOT), first formant frequency (F1), fundamental frequency (F0) and burst amplitude (BA) that influence how a listener perceives a plosive with respect to the feature of voicing. For a given speaker, these cues may be complementary, providing the listener with redundant information to identify the contrast accurately or they may be contradictory, leading to perceptual uncertainty. While previous investigators have studied the development of VOT in children, limited information is available regarding children's production of these multiple cues and how these influence adult listeners' perceptions of children's voicing contrasts [1]. This study explored the relationship among four acoustic characteristics of word-initial voiced and voiceless alveolar plosives produced by four age groups of normal speakers of Western Canadian English and was designed to answer the following questions:

1. Are there differences among speaker age groups in the frequency of correct and incorrect or ambiguous identifications of voiced and voiceless plosives by adult listeners?
2. Are there age group differences in VOT, F1, F0 and BA for correctly identified voiced and voiceless plosives?
3. What combination of these four acoustic cues best predicts listeners' identification of word-initial voiced and voiceless alveolar plosives?

2. Method

Ten female and 10 male subjects in each of four age groups (2.5-3.0 years; 4.5-5.0 years; 10-11 years; and adults) (N=80) were audio tape recorded as they produced five spontaneous repetitions of the minimal pair monosyllabic words "tot" and "dot". All subjects passed age appropriate hearing and speech screening tests. Computer files were made for each recorded test word using CSpeech. The recordings were low-pass filtered at 10.5 kHz, and sampled at 26 kHz. PERCEPT software operating with CSRE 3.0 was used to present the computer files of the recorded utterances to listeners for perceptual identification and to record and score their responses. Five speech-language pathologists judged the word initial plosive as /t/, /d/, or ambiguous with respect to voicing, for each token recorded from the subjects. Words were presented in blocks according to speaker age. The 10 words spoken by each of the 20 subjects in an age group were randomized across speaker and initial consonant.

CSpeech was used to obtain measures of VOT, F1 and F0 at vowel onset, and BA, for one token for each voicing condition per subject, randomly selected from the subject's perceptually validated tokens. Perceptually validated tokens were those where at least four of five judges correctly identified the voicing feature of the initial consonant. This provided 80 tokens for analysis for each voicing category. F0 and F1 values were transformed to log Hz so that differences between voicing conditions within an age group could also be compared across age groups (different vocal tract sizes).

3. Results

The youngest age group (2.5-3.0 years) had significantly fewer correct and significantly greater incorrect and ambiguous identifications, compared to the older groups. The youngest age group also had significantly fewer perceptually validated tokens than the older age groups. While all speakers produced some perceptually distinct voicing contrasts, listeners' reliability in judging the contrasts increased with speaker age.

For all age groups, VOT was longer, F1 onset frequency was higher, and burst amplitude was greater for /t/ than /d/, while F0 onset frequency did not differ significantly. No significant age effects or age by voicing condition interactions were found for VOT, F0 or BA. Variability on the acoustic measures decreased as age increased. A discriminant function analysis was conducted using one perceptually validated token for each voicing category from half of the speakers in each age group. To make meaningful spectral comparison across ages groups (who varied in vocal tract size), the F1-F0 difference in log Hz was entered as the "spectral variable". This analysis revealed that VOT was the primary cue for the perception of voicing while F1-F0 information at the onset of the vowel, and to a lesser extent burst amplitude, provided secondary cues. The ability of the function to discriminate between /t/ and /d/ was statistically significant (chi square (3) = 109.51, p=0.000). The classification function for the tokens that were used to derive the discriminant function prediction equation correctly classified 96.25% of these tokens. The classification for the cross-validation procedure which used a perceptually validated token from the remaining half of the speakers in each age group correctly classified 98.75% of these tokens.

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ACOUSTIC INTERPRETATION OF PHARYNGEAL ARTICULATIONS

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Introduction and Method

This investigation examines the acoustic contrasts present in pharyngeal articulations. Pharyngeal articulations include consonants found in Native North American, Caucasian and Semitic languages; and habitual postures such as pharyngeal voice quality settings in individual speakers. Our investigations, up until now have involved spectrographic examination focusing on behaviour of the first and second formants. It appears to be necessary to look at energy higher than the first two formants in order to distinguish between pharyngeal consonants. Currently, in this study spectral data is compared with videolaryngoscopic information obtained from a trained phonetician, focusing on viewing behind the epiglottis. This combination of examinations offers information that is helpful in defining place and manner categories that need to be distinguished phonetically (according to IPA standards) in this area of the vocal tract.

It appears that the pharyngeal mechanism used to produce pharyngeal consonants and to maintain a pharyngeal voice quality setting is the same [1]. That is, the epiglottis approximates the pharyngeal wall causing constriction. What happens to the cavity behind the epiglottis has been described videoscopically by Laufer and Condax [2], however, the acoustic details have not been addressed.

Given that pharyngeal articulations either in consonants or as voice quality settings involve the same 'place' of articulation, it is the aim of this study to distinguish the range 'manners' of pharyngeal articulations available from those used in language. The International Phonetic Alphabet (IPA) chart lists two pharyngeal fricatives and a series of epiglottal consonants. It has been proposed (Esling, this volume) that the epiglottal consonants are produced in the same place of articulation as the pharyngeal fricatives but involve varying manners of articulation.

In these investigations a trained phonetician has produced a series of pharyngeal articulations involving specific consonants [ʔ,ʡ,ʕ,ʢ,ħ,ʜ]. Vowels have also been produced with a pharyngeal voice quality setting and recorded at eight pitch intervals. The data collected include video images of the pharyngeal consonants and spectrographic analysis of pharyngeal consonants and pharyngealized vowels.

As noted by Laufer and Condax [2] the video information for pharyngeal consonants suggests the larynx is raised and approximates the epiglottis. It is the assumption here that the degree of approximation can result in such manners as frication and trilling. Consequently, it is suggested that the epiglottal consonants occupy the same place as pharyngeal consonants but vary in manner. The result of this investigation may support the addition of the epiglottals to the pharyngeal column in the IPA chart in a way that compares with the manners for uvular consonants.

According to past research the effects of raising the larynx and constricting at the pharynx, causes a raised F1 and a lowered F2 [1,3]. However, this does not fully describe the trilling, frication and variation in larynx height of the pharyngeal consonants that are observed in these investigations. The higher formants do not appear to be well described in the literature but seem to be significant from preliminary examinations. It is noted here that pharyngeal consonants produced intervocalically in an /ɑ/ environment both with raised and lowered larynx settings, have a dominance of energy in the region of the fifth formant. The /ɑ/ vowel exhibits very similar first and second formant characteristics with pharyngeal consonants.

Summary

A trained phonetician has produced pharyngeal consonants and voice quality settings that provide extreme, or peripheral examples. These examples can then be compared cross linguistically or within individual voice quality settings. A video examination provides a means to describe these consonants by observing interaction of anatomical features and, spectrographic analysis can help to support these observations. The video data show the epiglottis separating the vocal tract in the region of the pharynx with varying degrees of constriction. It is also possible to see the arytenoid structures of the larynx raise up toward the epiglottis and at times, the aryepiglottic folds trill laterally. It is assumed the degree that the larynx approximates the epiglottis from below has an affect on the perceived pharyngeal consonant. The effect of voicing on pharyngeal postures is also considered. Closer examination of spectrograms and comparison with uvular consonants may reveal a pharyngeal category that compares with uvular consonants.

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Acquisition of [r-l] phonemic contrast by English speaking children.

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1. Introduction

Multiple acoustical cues are a critical component of perceptual and productive distinction between phonemic contrasts. For example, usage of both the spectral and temporal information enables English listeners to discriminate [r] and [l] sounds with a high efficacy. Specifically, the difference in the frequency onset of the third formant (F3) frequency transition between [r] and [l] is used in the discrimination of this contrast. For [r] the onset of F3 is close to the second formant (F2) and has a rising transition. F3 of [l] is high relative to F2 and falls slightly towards the F3 formant of the following vowel. In addition to the spectral cue, the durations of both the steady state and the subsequent frequency transition of the first formant (F1) are used by English listeners in differentiation of prevocalic [r] and [l]. The frequency transition of F1 in [r] is much longer relative to the frequency transition of [l]. Discrimination of [r] and [l] sounds by English listeners and speakers is based on the integrated phonemic percept. It was found that the perceptual ability to integrate acoustical cues improves significantly with age. There is a change among children in the ability to integrate spectral and temporal acoustical cues during perceptual distinction of [r-l] contrast as a function of age. Specifically, with increasing age, children rely more on phonemic similarities than on acoustic dissimilarities. Thus, the attainment of perceptual distinction between phonemic contrasts (at least for [r] and [l]) is a gradual and progressive development in the proficiency of categorical perception.

This study examined development of the incorporation of spectral and temporal acoustical cues in the production of the phonemic contrast between [r] and [l] sounds in the initial prevocalic position for 3 to 6 years old English children.

2. Method

Ninety-six monolingual English children, divided into four age groups (3, 4, 5, 6 years old), participated in this study. All children were native speakers of Canadian English, had normal peripheral hearing (10 dB HL or better for 250 Hz to 6000 Hz) and normal articulation according to age. Each subject was asked to produce 6 words with [r], [l], or [w] in the prevocalic position in two vowel contexts ('rake', 'lake', 'wake', 'rock', 'lock', 'walk'). Three repetitions of each word were recorded using the microphone B&K 4165 and a DAT recorder, SONY DAT-75ES. The recorded speech samples were digitized at sampling frequency of 40 kHz with 16-bit amplitude accuracy.

3. Results and Discussion

Mixed 3 factor repeated ANOVAs with age as the between subjects variable (4 levels) and vowel (2 levels) and consonant (3 levels) types as the within variables were conducted on dependent variables. Separate analyses were conducted on several acoustical features: F1 transition duration; F2 transition duration; F2 onset frequency; F3 onset frequency; and the temporal cue defined as the ratio of the F1 transition to the F2 transition duration. Within each level of consonant type and vowel context, simple main effects were conducted to examine age-related changes in the production of [r], [l] and [w]. For all follow up tests Bonferroni adjustments were used to control the Type 1 error rate. There is a significant change in the use of acoustical cues for distinction of [r-l] contrast by English children as a function of age. Thus, spectral acoustical cues such as the difference between F3 and F2 onsets of formant, and the onset of third transition formant change with age. Specifically, with increasing age, the difference between F3 and F2 onsets of the formant transition of the [r] phoneme decreases in both vowel contexts ($F(3,92)=3.67$, $p<.01$), while the same difference for [l] and [w] phonemes does not change with age (Figure 1). Similarly, the onset of F3 formant transition for [r] phoneme decreases rapidly with age ($F(3,92)=13.52$, $p<.0001$), while that for [l] and [w] phonemes decreases only slightly. The temporal cue such as the F1 transition duration or the F1/F2 transition duration does change significantly with age in the [a] vowel context ($F(3,92)=7.17$, $p<.0001$ for [l]; $F(3,92)=8.27$, $p<.0001$ for [r] and $F(3,92)=5.16$, $p<.002$ for [w]). In this context, the ratio of F1 and F2 transition durations for all of three consonants decreases with age. However, this ratio for the [l] phoneme is lower than that for the [r] and [w] phonemes independent of age.

In conclusion, the results obtained in this study demonstrated that in order to produce phonemic contrast it is necessary to perceptually distinguish phonemes. The results revealed that children with lower perceptual scores produce [r] sounds with greater variability and higher values of F3 and 'F3-F2' as well as higher values of temporal cue in [l] phoneme than children with higher perceptual scores. Moreover, the lack of ability to integrate temporal and spectral cues in the perceptual task demonstrated by the younger groups of children was reflected in their poor productive distinction between the [r] and [l] sounds: the duration of the F1 transition and the F3 onset frequency did not differ in the produced [r] and [l] sounds. Thus, it seems that the attainment of productive distinction of a phonemic contrast is closely related to perceptual ability to focus on similarities and/or dissimilarities between a perceptual percept derived from the integrated information of spectral and temporal cues. The mastery of phonemic categorization is acquired during the process of exposure to spoken English. An effective category formation is a lengthy process that depends on the amount of environmental variability necessary for generalization, as well as on the ability to pay attention to a few acoustical cues at the same time. Thus, from the results discussed here and elsewhere, apparently there exists a strong relationship between the perceptual ability to process information about phonemic percept, and its productive execution.

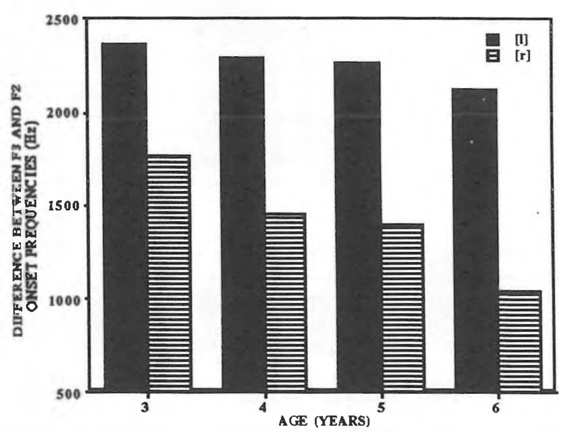

Figure 1. Difference between F3 and F2 onset frequencies for [r] and [l] as a function of age.

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TUNING PAST AND PRESENT

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In Western music the dominant tuning system is Equal Temperament, which evolved, largely empirically, along with the developement of musical composition. It was not fully realized until the twentieth century when the technology became known. Previously, many attempts were made to expand musical possibilities, gradually using more and more of the remote keys, resulting in what we regard today as historic tunings.

It is intended to demonstrate, via recordings and piano tunings , a few of these earlier systems, including Just Intonation as a starting point, particularly Meantone tuning, which was used as late as the early part of the nineteenth century. These tunings will have been performed aurally in order to give authenticity to the exercise.

The Sanderson Accu-tuner is an electronic tuning aid used by many piano tuners; Denis has developed a method of employing one to predict the beat rates of the most used musical intervals in tuning. This could have significant use in the aural training of tuners. This will be demonstrated.

Other aspects of tuning include the effects of inharmonicity of piano strings on tuning and the arithmetic of Meantone and Equal Tempered tuning.

Piano tuning is essentially a craft and very much a way of life, and while individual experiences vary widely, we feel sure a few anecdotes will provide a glimpse of what it is sometimes like to be a piano tuner -technician.

MEMORY FOR POPULAR MUSIC IN ELDERLY AND YOUNG ADULT LISTENERS

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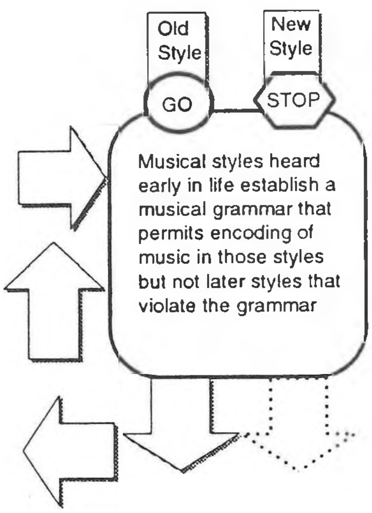
Background

Although research on music perception has accelerated over the last two decades, most of this research has involved young adult listeners. The present research, however, compares young and older adults on a musical memory task. Its purpose is to test a theory of the acquisition of musical knowledge or grammar. Consistent with evidence that the brain is more plastic early in life, we propose that musical grammar or knowledge about musical style can be acquired best early in life. Moreover, this grammar provides a framework for encoding music throughout the rest of life. It follows that music which is consistent with this style is easily retained in memory but music which violates this style is less easily retained (see Figure below).



Much research on music perception has used highly controlled tone sequences which may not "begin to do justice to the range of patterns and relationships inherent in real music" (Sloboda, 1985). The present study avoids this problem by using excerpts from actual music.

Present Study

Fifteen undergraduate students (mean age=19.8 years) from the University of Prince Edward Island and eleven senior adults (mean age=70.5 years) who were attending an Elderhostel conference participated in an experiment which investigated familiarity and memory for music which differ in decade of popularity from 1900's to 1990's. Subjects listened to 40 short musical excerpts and rated their familiarity of the tunes. This was followed by a test of immediate retention for 20 tunes (two old and two new tunes from each decade, permitting an unbiased recognition measure--*d'*--for each decade). All subjects completed the Otis-Lennon verbal intelligence test, and a questionnaire concerning musical experience.

Results and Conclusions

For young adults, familiarity and recency of the excerpt were positively correlated (.74) and differentiation between old and new song excerpts was greatest for music from the 1980's. For senior adults, familiarity and recency were negatively correlated (-.29) and differentiation of old and new songs was greatest for songs of the 1940's.

Thus, for both age groups, the priority in memory is shown for music associated with styles popular early in life. Performance, it is argued, was thus independent of the excerpts themselves and instead dependent on having considerable exposure to music of that style early in life. Listeners have difficulty assimilating tunes from less familiar styles because they do not fit early acquired frames of reference. This idea is also consistent with research on the mental plasticity associated with youth and the decline of mental plasticity with age. However, it does not contradict the fact that much musical learning can nevertheless continue with age.

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Note: The paper is based on an Honours thesis of Lisa D. Clyburn, Memory for Popular Music in Seniors and Young Adults: How Does Memory for Style Change with Age? Department of Psychology, University of Prince Edward Island, August, 1996.

A COMPARISON OF A MUSIC-BASED AND VERBAL-BASED REMINISCENCE INTERVENTION PROGRAM IN ENHANCING PSYCHOLOGICAL WELL-BEING AMONG ELDERLY NURSING HOME RESIDENTS

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RATIONALE

Long-term care residents are among the most depressed in the developed world (Stones et al., 1995a). They experience a deep sense of loss as they must leave the familiar surroundings of home and become accustomed to living in an environment where everyone around them is old and unhealthy. It is understandable that this experience has profound effects on their sense of independence, dignity and self-identity.

Often, nursing home residents find more comfort in memories than in their present situation and search for opportunities to regress to the past (Rattenbury & Stones, 1988). Erik Erikson proposed that an essential aspect of later adulthood entails the evaluation of one's life through a life review process (Erikson, 1963). Reminiscence, or the reviewing of one's past, has become an increasingly popular activity in the nursing home context. Reminiscence programs have had success in alleviating depression (Rattenbury & Stones, 1989), increasing self-esteem (Lappe, 1987), self-identity and self-worth (Baker, 1985) and life satisfaction (Cook, 1991). A recent long-term study of the effects of a reminiscence intervention program reported that participants had significantly higher psychological well-being scores and postponed morbidity and mortality (Stones et al., 1995b).

Music therapy has also become an increasingly popular activity with the aged. Music has had success in improving the self-esteem of the elderly (McCloskey, 1985); increasing social interaction (Wylie, 1990); and reducing maladaptive behaviour (Gibbons, 1988). Music is also effective in creating an atmosphere of the past and is often capable of rousing various emotions within us (Karras, 1983). Familiar music can be used as a powerful stimulus for invoking memories of the past for the elderly, particularly songs from their young adult years (Clyburn, 1994). We have a tendency to relate certain songs to different aspects or periods of our lives and music from a particular era is often aligned with certain historical events. It would seem reasonable, therefore, that music would be an effective means of stimulating reflections of the past among the elderly.

THE PRESENT STUDY

Only one known study has investigated the beneficial effects of a musical intervention program with the elderly (Bennett & Maas, 1988). They reported that a reminiscence program using familiar memories to invoke memories was more effective in promoting life satisfaction and ego integrity than a program that elicited memories through verbal questions. The present study extended Bennett and Maas' (1988) evaluation of the music-based evaluation program by including no-treatment control groups that had not been used in Bennett and Maas' (1988) design. Fourteen residents (mean age= 82.4 years) at each of two nursing homes were randomly assigned to either a no-treatment control group or a reminiscence group. At one nursing home, popular music selections from the residents' young adult years were played throughout each session. Memories and thoughts invoked by these musical selections were discussed. At the other nursing home, a general-questions approach was adopted. Prior to and following a six-week intervention period, subjects completed the Memorial University of Newfoundland Scale of Happiness (MUNSH), the Life Satisfaction Index A (LSIA), and the Memorial University Mood Scale (MUMS). In accordance with findings of Bennett and Maas (1988), there were greater positive changes in the MUNSH (happiness) scores for those in the music-based group (5.0) than for the verbal-based group (1.4); MUNSH scores did not increase for either control group (0.0 and -0.9). The interaction approached significance ($p > .09$). The average enjoyment rating was also slightly higher for the music-based group (4.5) than the verbal-based group (3.5), indicating that the music seemed to add a more entertaining, sociable atmosphere. In general, the results of the present study were consistent with the view that music is effective as a catalyst for reminiscence in therapeutic contexts.

Note: This research is based on an honours thesis by Leah Clyburn entitled A Comparison of a Music-based and Verbal-based Reminiscence Program in Enhancing the Personal Well-being of Elderly Nursing Home Residents. Dept. of Psychology, U.P.E.I., 1995.

The French Horn vs The Concert Hall

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Summary

The focus of work of this type is to bridge the communication gap between the arts and the sciences (in this case: music and acoustics) in order to bring about effective solutions to difficult problems which neither side can solve independently. The sound reflection problem experienced by the French horn player in a concert hall is an example of this kind of problem. Solutions to the reflection problem presented by the acoustician are most often judged to be inadequate by the musicians. In acoustics, as in most sciences, the common response to criticism from the arts is that until the concerns are expressed in scientific terms there can be no response. A more productive approach is to find the scientific basis behind the criticism and use this new information to both address the concerns of the artist and develop a better solution than could be had without the input of the artist. This paper is an example of the effectiveness of this approach.
The French horn is the only instrument played with the hand placed in the bell and with the bell pointing to the back of the hall. The sound of the horn is therefore affected by the player, his clothing and the back wall of the concert hall before it reaches the audience. The horn player must therefore deal with many unusual variables which are difficult to control. The result is that the horn player has a unique set of playing problems to overcome which are poorly understood by acousticians.

The problems arising from this unique playing position can be separated into three parts: the effect on the standing wave, the problem of reflected vs direct sound and the obvious time delay.

The Jack Singer Concert Hall in Calgary, Alberta was built in 1984. The horn players in the Calgary Philharmonic Orchestra have been complaining of acoustical problems related to excessive time delay, poor tone quality and loss of sound power since the orchestra first moved into the new hall. The attempts of the technicians to remedy the problems have met with little success and seem only to create new problems for the horn players which the technicians are unable to comprehend. The acoustical basis of the concerns of the horn players can be identified as follows:

1. The primary reflections must be early enough to provide a manageable time delay and avoid a loss in sound power.
2. The sound reflectors must return an acceptable balance of high and low frequencies.
3. The sound of the horn must be reflected away from the bell and dispersed into the audience to avoid interference with the standing wave in the horn and to provide the characteristic diffuse sound of the French horn.

An effective solution to the problem resulted from the construction of five reflectors specifically designed to address the concerns of the horn players. With the reflectors in place, the sound of the horn is reflected early but reflected away from the bell of the horn to avoid interference with the standing wave. As well the reflected sound is diffuse. The material and thickness of the reflector gives a well balanced spectrum which is acceptable to the horn players. The improved low frequency reflections eliminate the double attack problem and the loss of sound power by keeping the sound out of the area below the choir loft. The reflectors have been favourably received by the players, the conductor and the recording engineers who all feel that the problems have been solved.
An important by-product of this research is the development of a wavelet based sound analysis technique which uncovers aspects of tone quality that are unavailable by any other method.

The aXiØ and Live Electroacoustic Music [Presentation/Demonstration]

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Until very recently, electroacoustic music was created predominantly for fixed media such as tape. For many this is an ideal situation which allows the composer to fix precisely and permanently all aspects of a work. The composition becomes an object in time. The lack of a visual element heightens listening acuity, we listen with the ears of the blind.

In a concert, music is essentially a performing art and electroacoustic tape music is not always successful. With the advent of MIDI (Musical Instrument Digital Interface) keyboards and synthesizers, it seemed possible to bring electroacoustic music back into the sphere of live music. However, commercial MIDI keyboards are not ideal controllers of electroacoustic sounds which need greater control of the spectral evolution of sound in time.

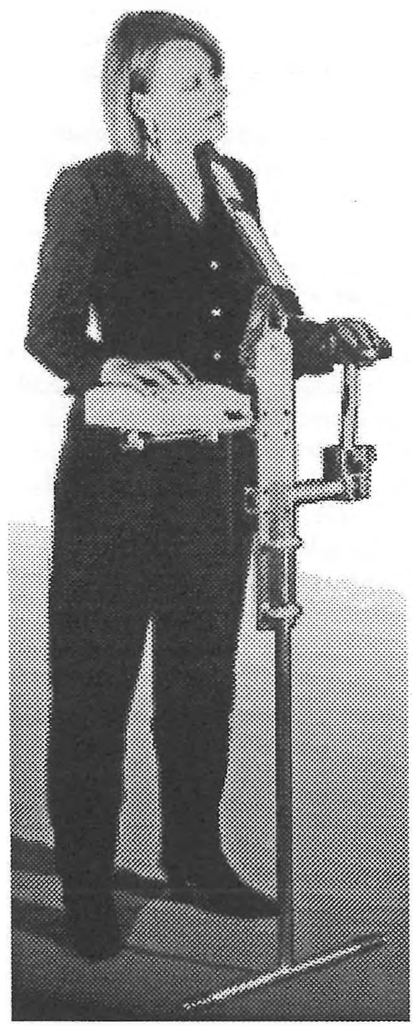
The **aXiØ** - 'alternative expressive input object' - is a new electroacoustic instrument/controller which gives the musician a broad range of expression and multi-dimensional control of MIDI synthesizers and samplers. It was designed and built by Brad Cariou (brad_cariou@nt.com) at the University of Calgary and conceived to provide digital artists with intimate control and flexibility for work in various new media. Using a Macintosh computer running the MIDI program MAX, it is completely user-programmable.

The aXiØ exists in two versions - the original prototype (see picture opposite) and the current version which has several design improvements. With a cross-like structure, the aXiØ stands about body height, vaguely resembling a stream-lined robot. It is played with both hands on three distinct playing surfaces:

a sophisticated joystick for the left hand,
a two-octave, velocity-sensitive keyboard for the right hand,
and, running up the musician's shoulder, is an array of assignable buttons used to change voices or trigger musical sequences. Ergonomic considerations were critical factors in its design.

In performance, the right hand selects and plays (with aftertouch) the notes or events while the left hand provides more expression and transformation of the sound. For instance, the left hand joystick could be used to move a sound in space, to change its overall volume, to sustain a sound, to change its timbre, and to do it all simultaneously in 'real time'.

The aXiØ has the potential to transform electroacoustic music from a predominantly studio art to a performance art. The presentation will demonstrate some of the expressive capabilities of the new aXiØ.



The original prototype of the aXiØ.

USING DIGITAL TECHNOLOGY IN A VOICE LESSON

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25 min.

The following presentation will demonstrate through analysis, (1) a front {i}, neutral { ɜ }, and back vowel { ɔ } and the imbalances in their formants as required for a professional singing performance, (2) how resonance balancing equates the imbalances {resonance balancing} in the three chosen vowels, (3) the required formant/vibrato relationships in a professional voice necessary for carrying power, (4) how poor posture and various breath support {management} systems alter the sung sound, (5) the effects of vocal tract stress and how its removal drastically alters-improves the sound spectrum.

These factors are important because (1) a professional singing voice must fill a hall of circa 3,500 seats, and be heard above a 90 piece orchestra, (2) stress, improper posture, and a non-flexible breath management system reduce quality and agility, (3) performer reliability is a prime factor enabling frequent performances. Therefore, the technique must fulfill these requirements. The voice suffers physical changes during lasting phonation and requires a rest period, hence the effects which could increase this trauma must be kept to a minimum. A stressed voice does not warrant attentive listening.

Digital analysis of the singing voice is recommended for the following reason: Not everyone has the same hearing. Some auditory defects can cause disabilities in judgement. By having them pointed out analytically, compensations can be made in the teaching and in the performing area. Sound analysis leaves little room for subjective arguments and can therefore become a binding tool in the field of pedagogy. Sound analysis points out important missing elements such as irregular vibrato, poor harmonic structure, noise and low energy, etc. The screen becomes a strong biofeedback device for both the student and the voice instructor. Many things go unnoticed in the heat of the moment but a machine is impartial and records everything. The lack of vocal energy and a sameness of sound {a non-distinctive singing style} in singing might limit the success of our young singers.

THE PHONEME AND THE GEOMETRY OF DECISION REGIONS IN SPEECH PERCEPTION

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INTRODUCTION

Various architectures have been proposed to account for human speech perception. This paper focuses on a non-parametric method for comparing them. Comparison of distinct architectures is not as easy as it may seem. One basic distinction hinges on whether a system is interactive, (i.e., has feedback between levels). Massaro (1989, Cognitive Psych. **21**, 398-421) demonstrated that certain predictions of the TRACE model were not compatible with empirical results, while those from his feed-forward fuzzy logical models were. McClelland (1991 Cognitive Psych, **23**, 1-44) showed that the failings of TRACE and of a simpler interactive activation model (IAC) may be due to the choice model employed and not to the presence of feedback.

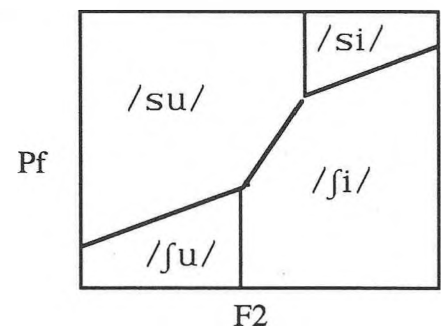
In experiments where more than one stimulus property is varied, simple geometric properties of decision regions may differentiate perceptual models. This criterion has previously been used to help characterize differences in computational power among neural networks with zero, one or two hidden layers by Lippmann (1987, IEEE ASSP Magazine **4**, 266-278). For evaluation of perceptual models, decision regions are defined by the modal response category, i.e., the category that is predicted to have the maximum number of responses on repeated presentation of the same stimulus. Such a characterization is independent of such complex issues as the choice process and the location of noise sources.

2 A CASE STUDY

Consider a problem in speech perception involving the words 'see, sue, she, shoe.' These may be synthesized by varying two stimulus parameters. 1) Pf, the frequency of a fricative pole. Lowering Pf can change an isolated /s/-sound to a /ʃ/-sound. 2) F2, the frequency of the second formant of the vocalic part of the syllable. Lowering F2 changes an isolated /i/ to /u/. It is well known (Whalen, 1989, Percep. and Psychophys., **46**, 284-292) that the boundary between /s/ and /ʃ/ on the Pf axis is lower when the following vowel is /u/ rather than /i/. Furthermore, the boundary between /i/ and /u/ is lower on the F2 axis when the preceding vowel is changed from /s/ to /ʃ/.

There are several ways such shifts might occur. One would be through an interactive-activation (IAC) model where the output of consonant units feeds back to adjacent vowel units and output from vowel units feeds back to adjacent consonant units. Results at keast partly compatible with those described in the previous paragraph could be obtained if /i/ output fed back with a larger positive weight to the input of /ʃ/, while /u/ output fed back more positively to /s/ input. Similarly, /s/ output could feed back more positively to /u/ , and /ʃ/ output more positively to /i/.

However, the apparent compatibility breaks down in detail. Several purely feed-forward (non-interactive) alternatives were examined by Nearey (1990, J. Phonetics **18**, 347-373) to model the results of the experiment of Whalen (1989) involving a two dimensional (Pf, F2) continuum. The results of this analysis indicated that the most appropriate model had decision regions depicted schematically:



The line separating /su/ from /ʃu/ is parallel to that separating /si/ from /ʃi/. These lines are not perpendicular to either axis. The boundaries of these regions are simultaneously modified in a continuous fashion by both stimulus properties. The /s/ response regions are favored by either increasing Pf or by decreasing F2. This is also grossly compatible with an IAC model. For example, higher F2 favors /i/ responses and the output of the /i/ unit feeds back more strongly to the input of the /ʃ/ unit, so raising F2 continuously (assuming output units have not saturated) increases /ʃ/ response area continuously.

The situation is quite different for the vowel boundaries. Both the /su/ - /ʃu/ boundary and the /si/-/ʃu/ boundary are unaffected by the frequency of Pf. The vowel choices are affected by the consonant choices by only what amounts to a bias effect. The /ʃi/ region is relatively larger than the /ʃu/ region, while the /su/ region is larger than the /ʃu/ region, but the shift in preference *does not vary continuously* with the frequency of Pf. However, the activation levels of the fricative units, are varying continuously and the usual richly interconnected architecture of IAC or TRACE would predict continuous change in vowel activations because of feedback.

Additional examples of applications of decision region geometry to issues in perceptual models will be discussed.

SPEECH PERCEPTION AND THE PHENOMENON OF FOREIGN ACCENT

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Although the phenomenon of foreign accent is commonly attributed to the learner's inability to articulate target language sounds, recent studies suggest that an important cause of foreign accent is a faulty perception of the sounds of the target language by non-native speakers of that language. This paper will review the results of recent experiments that document the important contribution of speech perception to the phenomenon of foreign accent.

The first experiment dealt with the pronunciation of a three-way vowel contrast (French /i/:/y/:/u/) by speakers of English and Portuguese where only two of these vowels occur (/i/ and /u/) [1]. Anglophones usually replace French /y/ with an [u]-like vowel, while Portuguese speakers replace it with an [i]-like vowel, with no apparent articulatory reason for this different outcome. A psychoacoustic test using a continuum of synthetic high vowels differing in F2 frequency and consisting of a forced-choice identification task revealed that the explanation to this phenomenon is to be found in the perceptual domain: The results of the test indicate that anglophones perceive French /y/ as belonging to their /u/ category, while Portuguese speakers perceive it as belonging to their /i/ category.

The second experiment focused on the actualization of a two-way consonantal contrast (French voiced and voiceless stops) by speakers of Mandarin Chinese, for whom the same contrast rests on different physical attributes [2]. A psychoacoustic test using a continuum of synthetic bilabial stops differing in VOT duration and consisting of a forced-choice identification task provided a description of the perception of these categories by speakers of Mandarin Chinese. These perception results, in conjunction with objective measurements of production data on the VOT values of French and Chinese plosives, provide an explanation for the replacement of French [p] with [b] and of French [b] with [p] by speakers of Mandarin Chinese, and make it possible to understand the variation observed in imitations of French plosives by native speakers of Mandarin Chinese [3].

The results of a perceptual study in which native speakers of Italian and anglophones were asked to identify single and double (geminate) intervocalic consonants suggest that Italian listeners are sensitive to differences in consonant duration and not to co-occurring differences in vowel duration in the process of distinguishing between words like fato and fatto [4]. On the other hand, it appears that anglophones are not sensitive to differences in consonant duration and that they rely instead on differences in vowel duration. This brings to light the importance of the perceptual expectations of the speakers of the target language as a component of the phenomenon of foreign accent: Although English learners appear capable of distinguishing between Italian single and geminate consonants on the basis of concommitant vowel duration differences, their continued reliance on the latter in production is likely to prevent them from being understood, or perceived as native or near-native by Italian listeners, because the latter appear not to be sensitive to vowel duration differences but only to consonant duration differences.

Because of the important role of speech perception in shaping accented productions of target language sounds, it seems that any accent reduction program must include a well-structured auditory training component based on a good understanding of how non-native speakers perceive the sounds of the target language, and keeping in mind the perceptual expectations of the speakers of the target language. The results of recent laboratory experiments in which auditory training led to a better perception and production of non-native sound contrasts ([2], [5]) reveal the value of this approach and suggest that it deserves to be integrated systematically into foreign language pronunciation instruction.

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THE EFFECT OF CONSONANTAL ENVIRONMENT ON FRENCH LISTENERS' PERCEPTION OF ENGLISH [u]

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1. INTRODUCTION

This paper challenges the traditional view of the French high rounded vowels, according to which French [y] is a difficult sound for English-speaking learners because their L1 inventory has no equivalent, and French [u] is less difficult because English does have an [u]. Current theory claims that L2 sounds are perceived by learners as members of L1 categories and that foreign accent is the result of their producing those L2 sounds according to L1 phonetic realization rules [1]; and English listeners have been found to categorize both French [y] and [u] as exemplars of English [u] [2]. It is known that English [u] has much higher F2 values in alveolar than in labial context [3], and that English listeners compensate for this in categorizing vowels [4]. One may expect, then, that English-speaking learners will produce both [y] and [u] with higher F2 values in alveolar than in labial environment. The *density hypothesis* [5] predicts that the 3 French high vowels, [i,y,u], will vary less as a function of context than the 2 English vowels, [i,u]. It was hypothesized, then, that the contextually-induced F2 shift of English [u] in production would be greater than the perceptual shift in French listeners and that, consequently, the latter would hear more English [u]s as French [y] in alveolar than in labial environment.

2. METHOD

2.1 Production of English [i,u]

Speakers of Canadian English recorded tokens of [bib], [did], [bub], [dud]. An F2 value was taken from each item.

2.2 French Listeners' Perception of English [i,u]

The above samples were played to European French listeners, who identified the vowel of each as [i], [y], or [u].

2.3 French Perception of Synthetic Vowels

The 20 members of a high-vowel continuum varying in F2 were attached to transitions appropriate for initial [b] and [d]. The [bV] and [dV] syllables were played 10 times in random order to French listeners for vowel identification.

3. RESULTS [6]

3.1 Production of English [i,u]

Analysis of speaker means for each vowel in each context revealed significant differences between [bib] and [did] ($p=.001$) and between [bub] and [dud] ($p<.0001$). In line with previous findings [3], F2 was higher in alveolar than in labial context for both vowels but the difference was much smaller for [i] (60 Hz) than for [u] (379 Hz).

3.2 French Listeners' Perception of English [i,u]

English [i] was heard as [i] in both contexts by French listeners. Analysis of [y] identifications given to English [u] showed the mean proportion for [bVb] environment (0.19) to differ significantly ($p<.0001$) from that for [dVd] (0.61). These findings support the hypothesis that French listeners would hear more English [u]s as [y] in alveolar context.

3.3 French Perception of Synthetic Vowels

For each listener F2-crossover values were computed for the [u-y] and [y-i] boundaries in each environment. Analysis revealed significant difference between the [bV] and [dV] contexts for [i-y] ($p=.001$; Mean=85 Hz), but not for [y-u] ($p=.5167$; Mean=14 Hz). While the former contrast was significant, this boundary shift was quite small in comparison to the production shift found for English [u].

4. DISCUSSION AND CONCLUSION

The perceptual results from synthetic vowels, along with the analysis of English production, explain the perception of English vowels by French listeners. In both contexts English [i] lies well within the French [i] category. By contrast, as is illustrated in Figure 1, English [u] production shows a very large contextually-induced shift where French perception shows none. These two functions place English [u] close to the boundary, and within the French [u] category when it is in labial environment, but within the [y] category when in postdental environment.

An assessment of the relative difficulty of French [y] and [u] for English-speaking learners must be context-sensitive if it is to be accurate. French [u] is not a less difficult sound than [y]. Rather, [y] is particularly difficult in labial context, while [u] presents its greatest challenges in postdental context.

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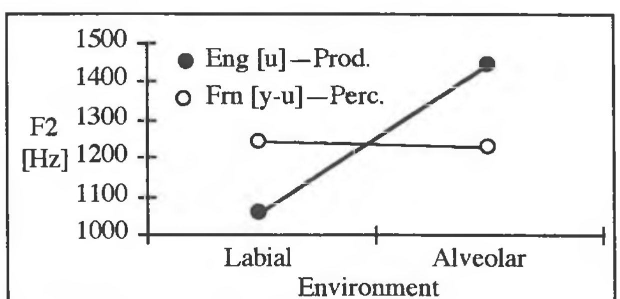
6. FIGURE 1: English F2 production means for [u] (male speakers) and the mean French [y-u] perceptual boundary.



THE COMPLEX INTERACTION OF LANGUAGE COMPREHENSION IN WORKING MEMORY

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Working memory has been found to be more complex than earlier research had supposed. The earliest view was one of simple capacity and any observed effects were interpreted as reaching the limits of that capacity. Baddeley (1990) presents a more dynamic model. The model that Baddeley has evolved is working memory consisting of three components: phonological loop, spatial-visual sketchpad, and central executive. The issue of interest is the phonological coding of visual language material and the interference that certain acoustic stimuli exert on that coding. Certainly, this model is not the only one, but by looking at some of the interesting effects observed in divergent research, it continues to be one that at the very least has heuristic value.

Salamé and Baddeley (1982) investigated the interference effects of acoustic similarity. These effects were on the recall of visually presented digit sequences suggesting an encoding process that makes them acoustic in nature. A masking process of this acoustic version of the visual information is suggested. Though white noise did have a significant masking effect, unattended speech, i.e., speech in a language not familiar to the participants, had a more significant effect. Varying the intensity of noise was not nearly as disruptive as the pattern of the noise, most particularly unattended speech. This result is contrary to simple masking explanations.

Salamé and Baddeley (1987) investigated patterned interference with the phonological store and found broadband noise failed to have a significant effect. However, the unattended speech sounds did have a significant deleterious effect. Experiment 3 added the feature of covert and overt rehearsal, while overt rehearsal had a significant effect, the noise level did not. A filtering mechanism that allows speech sounds into the phonological store while excluding non-speech-like noise is suggested (Salamé & Baddeley, 1987, p. 1192).

To investigate the filter concept more fully, Salamé and Baddeley (1989) experimented with various types of music, both instrumental and vocal. In digit symbol recall tasks, vocal music produced the most recall errors followed by instrumental music which was significant as compared to the quiet condition. In experiment 3, unattended Arabic speech was added and also pink noise that had its amplitude modulated by the Arabic prose. The unattended speech had the most significant effect on errors, whereas the modulated pink noise did not have a significant effect. Variance in amplitude did not distract attention, indicating acoustic similarity as the factor at work with the unattended speech.

The phonological loop is seen as two structures, the phonological store and the rehearsal loop (Baddeley, 1990). Articulation was considered earlier as the mechanism of the rehearsal loop, but it has now been rejected. Baddeley and Wilson (1985), in testing persons with dysarthria, found that acoustic confusion of phonemically similar material occurred when read, just as it does with persons once they have learned to read silently. This confusion indicates that the code conversion process occurs at a higher level than the articulation mechanism. Campbell and Dodd (1982, 1980) also concluded that articulation cannot account for cross-modality effects. Normal hearing participants produced modified recency and suffix effects when a speaker was seen, but not heard. Normal hearing participants use lip-reading to a certain degree. The recency and suffix effects are associated with acoustically presented lists, rather than visually presented lists.

Gathercole (1986) used a distracter task to determine if the modality effects are the result of articulation or not. The result was a significant recency effect for read aloud lists, regardless of the suppression task. Mouthed suppression only had a non-significantly disrupt recall of both silently read and read aloud list. The act of articulation is not the disruptive factor. Experiment two of Gathercole (1986) replaced the mouthed suppression task with a spoken suppression task. The spoken suppression task significantly reduced the recency effect of the read aloud lists. Total recall accuracy was also significantly reduced for both the silently read and read aloud lists in the spoken suppression condition. The acoustic nature of the suppression task, rather than its involvement with articulation, appears to explain the result the best.

Morris and Jones's (1990) exposed participants to sounds prior to performing digit recall tasks. When conditioned with unattended speech and recall tested under these conditions, the recall errors were reduced. Pink noise produced no such reduction. Speech similarity seems to be the requirement to condition a person against speech interference in recall tasks, rather than exposure to comparable noise levels. Morris and Jones (1990) suggested a combination of both the filter rejection and speech detection paradigm to be at work; Salamé and Baddeley (1989) suggested it might be either or. Clearly there is still much evolution occurring regarding the functioning of the phonological loop and the central executive deemed to control this interference.

Short-term memory is more dynamic than once thought. Acoustic coding of visual language material occurs to some degree. The code is speech related. The conditions of interference are complex, indicating some level of processing. Attention could play a role, but the mechanism of interference seems to be related to acoustic similarity with speech, rather than variations in intensity causing distraction. Articulation is not the answer in producing the code conversion of visual into acoustic. Further investigation into this acoustic coding and the mechanisms of interference is on going.

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Identification of bilabial plosives: Integration of VOT and burst intensity information.

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1. Introduction

Multiple acoustic parameters contribute to the perceptual distinction of many phonetic contrasts. The importance of timing of voice onset relative to the plosive release and as well of other acoustic cues to the voiced/voiceless distinction for stop consonants has been studied in a number of phonetic environments. Many of voiced/voiceless phonetic distinction correlates for stop consonants in initial position have been examined using synthetic speech, including voice onset time, a pitch change prior to voicing onset, the presence of a voiced transition or the frequency of the first formant at the onset of voicing. All of these studies indicate that the duration of Voice Onset Time (VOT) is a dominant and decisive phonetic correlate of the phonemic (voiced) contrast for stop consonants in a word-initial position. Thus, the difference in the duration of VOT in naturally produced plosive consonants in the initial position of words serves to distinguish voiced and voiceless tokens spoken by native talkers of eleven languages.

Moreover, it was demonstrated that produced voiced and voiceless stop-consonants vary also in the peak's intensity and in the duration of the burst of frication noise. The frication noise is of a longer duration and of a higher intensity at the release of a voiceless plosive. Thus, the perceptual categorization of bilabial stop-consonants in word-initial position, while mostly relies on a difference in the VOT, might also depend on an acoustical cue such as intensity of the noise burst (plosive release). Therefore, a goal of the present study was to examine the influence of the intensity level (Sound Pressure Level) of the initial burst on the categorization of bilabial stop consonants in the initial position of words into voiced and voiceless phonemes.

2. Method

2.1 Stimuli. Ninety nine stimuli differing in the duration of VOT and the intensity of initial frication burst, and ranging from [ba] to [pa] were generated using a synthesizer Monet (Trillium Sound Inc.) implemented on a NEXT computer using a 44 kHz sampling frequency. The duration of noise burst was constantly maintained for all stimuli. The first formant (F1) started at 200 Hz and increased in 40 ms to 760 Hz, the second formant (F2) started at 900 Hz and increased in 40 ms to 1600 Hz, the third formant (F3) started at 2000 Hz and achieved a steady state of 2400 Hz in the same time as F1 and F2. The fourth and fifth formants were constantly maintained at 3600 Hz and 4500 Hz, respectively, across the entire stimulus. Each synthetic syllable was 175 ms in duration. Change from an initial voiced to an initial voiceless stop consonant was achieved both by delaying the onset of energy in F1 relative to higher formants (F2 and F3), and by exciting F2 and F3 with an aspiration noise prior to the onset of a periodic source. Change from an initial voiced to an initial voiceless stop consonant was accomplished by changing VOT value from 10 ms to 30 ms in 2 ms steps. Moreover, the intensity of the burst varied from 0 dB SPL to 30 dB SPL in 3 dB increments.

2.2 Participants. Twenty eight young adults with normal hearing (<10 dB HL for 250 Hz to 8 kHz range) participated in the experiment. All listeners were phonetically naive and had never participated in a speech perception experiment.

2.3 Procedure. Listeners were tested individually on an identification task in an anechoic room. The stimuli were presented to the listeners via headphones AKG-240 at 70 dBA. Listeners were asked to identify syllables [ba] and [pa] and were instructed to press one of two buttons labeled either "ba" and "pa". Listeners were exposed to randomized 792 stimuli (each of 99 stimuli presented 8 times). Stimuli presentation and a collection of responses were controlled by a NEXT computer.

3. Results and Discussion

The mean percentage of the responses labeled "ba" for each stimulus and each participant were calculated. ANOVA with VOT and burst intensity as factors and the percent of [ba] responses as dependent variable indicated that the main effect of VOT ($F(2, 2755)=1137.32$, $p<0.0001$) and the main effect of intensity ($F(2, 2755)=66.006$, $p<0.0001$) were significant. Thus, both acoustical cues, VOT and burst intensity play roles in a distinction between [b] and [p]. Furthermore, the stimuli differing in the intensity were grouped into three classes: low (0 dB - 9 dB), moderate (12 dB - 21 dB) and high (24 dB - 30 dB). Similarly, stimuli that differ in VOT were categorized into three groups: short (10 ms - 16 ms), medium (18 ms - 24 ms), and long (26 ms - 30 ms). Separate two-factorial ANOVAs on the percent of [ba] responses were conducted for each listener. These analyses revealed that 25% of all listeners (Group 1), while categorizing tokens into [b] and [p], relied mostly on VOT information. However, majority (75%) of listeners (Group 2) in addition to VOT information relied very strongly on burst intensity's information. The 'Group 2' of listeners was more proficient in classifying stimuli, differing in their VOT, into two categories ([ba] and [pa]) than the 'Group 1' (Figure 1). Figure 1 shows that listeners of the 'Group 2' were more likely to assign stimuli into ba-category in a presence of a low burst intensity, and were less likely to identify stimuli as [ba] when the burst intensity was high than were the listeners of the 'Group 1'.

This finding suggests that all listeners are using information about VOT and burst intensity while categorizing initial bilabial stop consonants into voiced and voiceless phonemes, however, they place a different weight to available information.

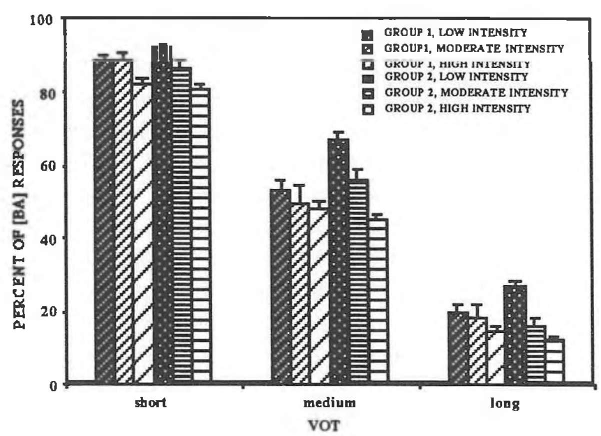

Figure 1. Percent of [ba] responses as a function of VOT for different parameters of the burst intensity and group of listeners. Standard errors are presented as error bars.

SPEECH SCIENCE AND SPEECH TECHNOLOGY

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1 INTRODUCTION

Speech synthesis technology has maintained a relatively close relationship with speech science since the beginning. Speech recognition technology has had a more volatile relationship with speech science. The victory of statistical pattern recognition methods (documented by Klatt 1977, JASA **62,** 1345-1366) in the ARPA sweepstakes and the subsequent success of template based and HMM systems had much to do with the development of a gulf between the two disciplines. In the face of failures of expert-system recognizers compared (e.g.) to Bayesian learning automata, many in the engineering community have concluded that it is more productive to accommodate uncertainty than to incorporate knowledge. However, the extension of speech recognition to large-vocabulary, speaker-adaptive systems has lead statistical modelers to develop architectures and heuristics that accommodate phonetic context and speaker variation in ways that are quite interpretable within a phonetic and speech science framework. This trend, if properly appreciated by both camps, may lead to a renewal of ties between the speech recognition and speech science communities.

2 SYMBOLIC CONTEXT

Consider a straw-man Model A that is a perfectly legitimate HMM, but one that no one actually uses. It assumes that the symbolic elements are phoneme-like units, represented by a single family of allophones that has no relation to its context. (This is "free variation" in linguistics 101 vocabulary.) Model A further assumes that each of these phones is realized by a sequence of one or more observation frames. These are typically 10-40 ms spectral sections or measures derived from them. Each observation frame is assumed to be independent of the others. This is the type of naive model phoneticians meet, when they are first exposed to HMMs. But the distance between HMMs in practice and phonetic theory is much less than this.

Although valid as an HMM, no one actually uses Model A because it won't work. Instead, major concessions are made to accommodate *contextual variation* that speech scientists have always insisted on. A rather standard model involves two kinds of concessions to speech science. The first concession uses what amounts to the Linguistics 101 strategy of context-conditioned allophones, in the form of what have been called "Wickelphones" (after Wayne Wickelgren's model of speech production in the 60's.)

For a full triphone implementation, each phoneme has a separate allophone for each combination of left and right phone contexts. Complete triphone sets are rarely used. Often, many triphones are not frequent enough in the training data to allow reliable estimates of their distributions. Some kind of data sharing is imposed between elements, smoothing over elements of a phoneme family. Typically this is done by numerical clustering, pooling estimated distributions over similar allophones. However, more principled, knowledge-driven methods are also sometimes used. In a recent paper by Jouvet, Bartkova and Stouff (ICSLP 94 283-286), a phonetically motivated clustering scheme out-performed a number of standard statistical clustering schemes for triphone smoothing.

Some dialog is clearly possible and it can cut both ways. The importance of clustering in triphone models may bear on the perceptual issue of exemplar-based versus prototype models. In exemplar models, every example is stored and new tokens are classified on the basis of distance to previously learned examples. In a prototype model, only an abstract summary of a category is stored. Speech recognition research demonstrates that enumerating contexts may exact the heavy price of inadequate generalization. Heavy smoothing of triphone models moves them further from exemplar-like toward prototype-like behavior.

3 STIMULUS CONTEXT

Apart from symbolic context (allophones), which are fully legitimate additions to HMMs, there other concessions to context. HMM theory requires that successive observations are conditionally independent of each other. But this assumption (essential to the strict Bayesian interpretation of HMMs) is deliberately violated for better performance. Two "standard violations" are 1) to allow massively overlapping analysis frames 2) to code *delta coefficients*, involving rate of change of properties. These can span up to 50 ms and often cross phoneme boundaries, so that the last state of a consonant HMM gets to preview information about a following vowel and *vice versa*.

Speech perception research provides a rich source of information on how human listeners use context in decoding speech signals. Careful study of existing evidence may help develop perceptually motivated accounts of context that can be engineered more forthrightly into stochastic models. There are already a fair number of hybrid neural-net plus "relaxed" HMM models that make more elaborate attempts to deal with context in a manner that seems more plausible to researchers in speech perception (e.g., Afify, Gong and Haton ICSLP 94, 291-293). This is one of many hopeful signs. Real progress in both camps is likely to be accelerated if the gulf between speech science and speech technology is actively bridged by workers on both sides who are willing to critically consider the others' insights rather than dismissing them as scientifically naive or as ivory-tower dreams.

Modes in Layered Media

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Introduction: Mode models of sound propagation in layered media are now well established. Owing to its complexity, however, the theory of modes can only be pressed to completion for simple layered media consisting of just one or two homogeneous layers, while illustrating one or another subset of the many different types of modes that occur in realistic geoacoustic media. My objective is to outline a general approach for all modes that is ultimately the same as that used for analyzing modes in any linear system, and to highlight some essential rules of mode behaviour that apply in many-layered, fluid and solid media.

Modes in linear systems: The partial differential equations (PDE's) governing a linear system—whether acoustic, electronic, or mechanical—can often be simplified using integral transforms (such as the Fourier transform), that together with boundary conditions reduce the mathematical problem to a system of linear equations L, in selected physical variables **v**, subject to some forcing **f**;

$$L\mathbf{v} = \mathbf{f}.$$

A mode in such a system is by definition a solution that persists in the absence of forcing, when **f**=**0**, which occurs when the determinant $|L|=0$. The oscillations of the mode are given by the corresponding null space solution **v**.

Modes in Layered Media: Likewise, a Fourier-Bessel transform and boundary conditions reduce the PDE's for elastic wave motion, where **v** now holds the complex valued down (+) and up (-) going compressional (P) and shear (S), plane wave strengths at the layer interfaces, for plane waves at frequency ω and horizontal wavenumber k; $L(\omega,k)$ represents the transmission and refraction of those waves through layers and interfaces; and **f** represents the source excitation. Here again, a mode exists when $|L(\omega,k)|=0$. Using a computer, we must search for the roots ω and k where this condition is satisfied. For a given ω there is a series (possibly infinite) of modal wavenumbers k.

Inhomogeneous Plane Waves: The propagation of plane waves through layered media is governed both by Snell's law and the boundary conditions applied a the interfaces between layers. In effect, Snell's law states that the horizontal wavenumber k for a plane wave must be the same in all layers. To include all possible modes and energy absorption by the media, we must consider *inhomogeneous* waves, whose horizontal and vertical wavenumbers, $k = k_r + ik_i$ and $\gamma = \gamma_r + i\gamma_i$, are complex; the wave's time t, horizontal range r, and depth z dependence within a homogeneous layer going as

$$e^{i(\pm\gamma z+kr-\omega t)} = e^{(\mp\gamma_i z-k_i r)}\, e^{i(\pm\gamma_r z+k_r r-\omega t)}$$

Unlike k, the vertical wavenumber γ changes in each layer and according to the wave type, going as $\gamma^2(z) = \omega^2 / c^2(z) - k^2$, where $c(z)$ is the P- or S-wave phase speed.

Proper Modes in the complex *k*-plane: $\gamma_i \geq 0$ for proper (physical) modes whose vibrations remain bounded as $z\rightarrow\pm\infty$. We can deduce where most of these proper modes must lie. If the vibrations of a mode are significantly large and span a layer (or band of similar layers) of significant thickness H, then γ_i must be small, otherwise the vibrations vanish because $e^{-\gamma_i H} \rightarrow 0$. k for such modes therefore lie close to the line defined by $\gamma_i = 0$ —a line that follows a portion of a hyperbola, in the first quadrant of the complex k-plane, asymptotic to the real and imaginary axes. The vibrations of proper modes whose k are far from this line must in depth be confined close to an interface. These may be Rayleigh or Scholte interface modes, or plate modes in a thin band of solid layers.

Other Properties: Continuing this way, we gain insight into the classification of modes as propagating or evanescent, proper or leaky, predominantly P or S, duct or interface modes, which in turn enables us to judge by inspection what modes are likely to exist in realistic media. Helpful analogies with modes in other linear systems can also be drawn.

RAYTRACING IN ANISOTROPIC MEDIA

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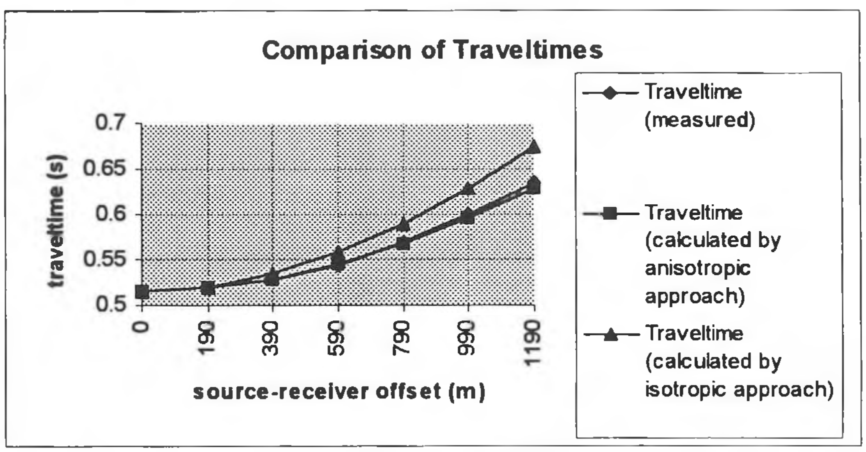
An analytic method relating incidence, reflection and transmission angles at an interface between anisotropic media is elaborated. The method relies on the continuity conditions relating tangential components of phase slowness across the interface, and on the fact that the ray is perpendicular to the phase-slowness surface. The rather familiar concepts of vector calculus are used to create a template for calculating phase and group angles across discontinuities. The angles involved in wave propagation through layered anisotropic media are, at times, significantly different than their isotropic counterparts. Thus the trajectories derived in raytracing by the isotropic versus the anisotropic approach differ considerably if the medium is significantly anisotropic. More importantly, measurable quantities, e.g., traveltimes, differ depending on the approach taken.

This template is used to derive analytic expressions for phase and group angles, and to elaborate a raytracing scheme for acoustic waves in transversely isotropic (TI) media. A relative simplicity and clarity of equations is achieved by using expression for phase velocities as a function of phase angle under the assumption of weak anisotropy.

The weak-anisotropy approach assumes that the difference in speed between the fastest and slowest propagation directions does not exceed 20%. Consequently, the expression for phase velocity is obtained by developing a rather cumbersome expression into a Taylor series and ignoring higher-order terms. The raytracing method can be used to calculate traveltimes for layered, weakly anisotropic media. In the process of raytracing, the bending of rays at discontinuities is taken into account using a derived anisotropic equivalence of Snell's law.

The results of a physical laboratory experiment, which involved propagation in the symmetry plane of an orthorhombic material with known characteristics, have been compared with theoretical calculations. The comparison indicates that the anisotropic approach predicts reasonably well the experimental results and yields a significantly better prediction than an isotropic one. It also suggests that weak-anisotropy assumptions can be useful in practical applications as long as one remains within the intended limits of approximation.



Comparison of measured and calculated traveltimes for a laboratory experiment involving a transmission of acoustic signal across a two-layer anisotropic medium separated by a planar interface. The source-receiver offset corresponds to the separation measured parallel to the interface. The actual dimensions of the physical sample used have been scaled to allow data processing using software developed for seismological studies.

A finite difference scheme for wave propagation through absorbing media

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1. Introduction

Because the earth is not a perfectly elastic medium, oscillatory motion is damped; as a result seismic waves propagating in the earth suffer absorption. The intrinsic anelasticity of the earth is usually modeled by the linear theory of viscoelasticity, which generalizes the theory of elasticity and that of viscous fluids. The general viscoelastic equation of motion (VEM) is an integro-differential equation, and in most cases of interest must be solved numerically. We present a finite difference scheme based on the reformulation of the VEM as a higher than second order partial differential equation (PDE).

2. Elasticity and Viscoelasticity

The basic equation of motion for a solid continuum is

$$\frac{\partial \sigma_{ij}}{\partial x_j} = \rho \frac{\partial^2 u_i}{\partial t^2}, \quad i = 1,2,3, \tag{1}$$

where u_i is the particle displacement vector, σ_{ij} the stress tensor, and ρ the density; and where the summation convention is assumed. Elasticity theory follows by assuming a stress-strain relation of the form

$$\sigma_{ij} = c_{ijkl} e_{kl}, \tag{2}$$

where $e_{kl} = 1/2\left(\partial u_k / \partial x_l + \partial u_l / \partial x_k\right)$ is the strain tensor, and c_{ijkl} is the tensor of elastic constants (Aki and Richards, 1980). Viscoelasticity generalizes the above relation by allowing the stress to depend on the entire past history of the strain:

$$\sigma_{ij}(t) = \int_{-\infty}^{t} G_{ijkl}(t-\tau) \frac{de_{kl}(\tau)}{d\tau} d\tau \tag{3}$$

(Christensen, 1971).

For simplicity, we consider SH waves, transverse waves polarized in the horizontal plane, which for a vertically inhomogenous medium are decoupled from the two other possible wave polarizations (P and SV). For SH waves, equation (3) becomes a scalar relation:

$$\sigma(t) = \int_{-\infty}^{t} R(t-\tau) \frac{du'(\tau)}{d\tau} d\tau, \tag{4}$$

where the prime denotes differentiation with respect to the spatial coordinate.

The VEM is an integro-differential equation and in general is tractable only by numerical methods. To make even a numerical solution practical, an arbitrary relaxation function $R(t)$ is approximated by a finite sum of decaying exponentials (the generalized Maxwell model) (Emmerich and Korn, 1987). By introducing auxiliary "memory" variables, the VEM may then be reformulated as a system of coupled PDEs and solved, for example, by the finite difference method.

3. Differential Formulation

Alternatively, the viscoelastic stress-strain relation (equation (4)) may be formulated using differential operators:

$$P\left(\frac{d}{dt}\right)\sigma(t) = Q\left(\frac{d}{dt}\right)u'(t), \tag{5}$$

where P and Q are polynomials of degree N in the differential operator d/dt, and where the whole of the spatial inhomogeneity is contained in the coefficients of Q. Instead of an integro-differential equation, the VEM becomes a PDE of order $N+2$ in time:

$$\frac{\partial}{\partial x} Q\left(\frac{\partial}{\partial t}\right) \frac{\partial u}{\partial x} = \rho P\left(\frac{\partial}{\partial t}\right) \frac{\partial^2 u}{\partial t^2}. \tag{6}$$

The solution of the associated initial value problem requires the values of the function u and its time derivatives through order $N+1$ to be specified at the initial time $t = 0$. However, the requirement that equation (5) be equivalent to (4) supplies constraints on the initial conditions so that only $u|_{t=0}$ and $\partial u / \partial t|_{t=0}$ may be freely specified. This is reasonable, since the VEM is an expression of the (second order in time) Newton's Second Law.

4. Conclusion

By replacing both space and time derivatives in equation (6) by centered finite difference expressions (while preserving the operator ordering of (6)), a stable finite difference scheme is obtained. This algorithm, including constraints on initial conditions, was then implemented in FORTRAN code. The program has been checked using a number of test calculations with known analytical or semi-analytical solutions.

While the approach via the differential formulation of the stress-strain relation does not appear to offer a computational advantage over existing approaches, it is of theoretical and conceptual interest. In particular, in this approach the time evolution of the wavefield is governed by a single PDE. By constructing a working computer code we have shown that this approach is practical for computation.

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REFLECTIONS ON BOUNDARIES

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Acousticians studying propagation in air and in water need to account for reflections at boundaries—both amplitude and phase effects—in order to model realistic scenarios. The typical assumptions made about boundaries in each discipline are somewhat different, however. This paper considers how acousticians in different disciplines cope with boundary effects and attempts to compare these methods.

A typical simplified boundary assumption for an acoustician working in air (either outdoors over ground or indoors with a porous wall covering) is the "locally reacting" boundary; that is, regardless of the incident field, the ratio between the pressure at the boundary and the normal component of the particle velocity is a single (although frequency-dependent) complex quantity called the surface impedance. In this case, the complex plane wave reflection coefficient $R(\theta)$ is related to the surface impedance z (normalized by the impedance of air) through the relation

$$R(\theta)=(z\sin\theta-1)/(z\sin\theta+1), \tag{1}$$

where θ is the grazing angle. Fig. 1 shows a calculation of the reflection loss $(-20\log|R(\theta)|)$ from such a surface for a typical case of normalized impedance $z=2.5-2.5i$. Note the large values of reflection loss at all angles away from grazing incidence, the almost-linear relation between loss and angle at near-grazing incidence, and the maximum of reflection loss near 16 degrees.

One can invert Eqn. (1) to define an angle-dependent surface impedance in terms of a general reflection coefficient:

$$z(\theta)=\frac{1}{\sin\theta}\frac{1+R(\theta)}{1-R(\theta)}. \tag{2}$$

Using this relation, we can compare the "impedance" of a surface having a general reflection coefficient with a surface having a constant impedance.

Underwater acousticians commonly assume that the seabed can be modelled as a semi-infinite elastic solid, the reflection from which has been worked out by Brekhovsikh[1], among others. Consider the reflection from a "sand" layer having the following properties: density 1.8 (relative to water), compressional wave speed 1850 m/s, shear speed 300 m/s, compressional attenuation 0.46 dB/wavelength, and shear attenuation 0.23 dB/wavelength. Assuming a water sound speed of 1500 m/s, the reflection coefficient can be calculated using the standard formula, and from Eqn. (2) the effective impedance follows. Fig. 2 shows the angle-dependent impedance for the sand layer and Fig. 3 shows the reflection loss.

In contrast to the locally-reacting (constant impedance) surface, the impedance of the visco-elastic solid shows a strong dependence upon the grazing angle of the incident plane wave, but this divides naturally into three regions: at low grazing angles the impedance is nearly constant, mostly negative imaginary (reactive); in the region of the critical angle for transmission of compressional waves into the seabed, the impedance changes rapidly from imaginary to real; at large grazing angles the impedance is again nearly constant, but is now positive real (resistive). The reflection loss curve also shows a large change at the critical angle. Note (again) the almost-linear relation between loss and angle at near-grazing angles.

The simplest boundary reflection models used by air acousticians (on the one hand) and underwater acousticians (on the other hand) have quite different characteristics; however, they are similar at near-grazing incidence, in that they both show near-linear relations between reflection loss and angle. In other words, in both cases the impedance is nearly constant at near-grazing angles.

[1] L.M. Brekhovskikh, *Waves in Layered Media* (Academic, New York, 1980), 2nd ed.

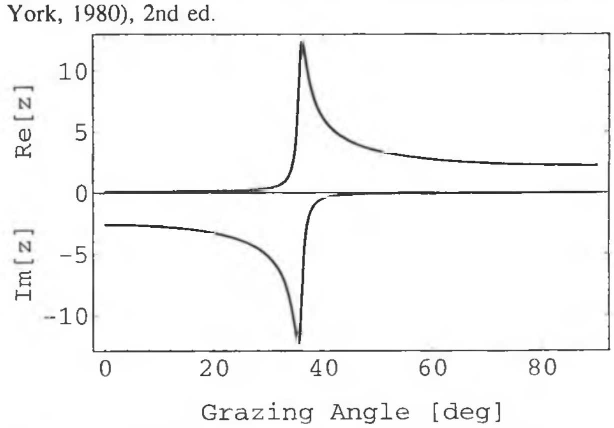

Figure 2 *Real (+) and imaginary (–) components of the normalized surface impedance (z) of the sand layer.*

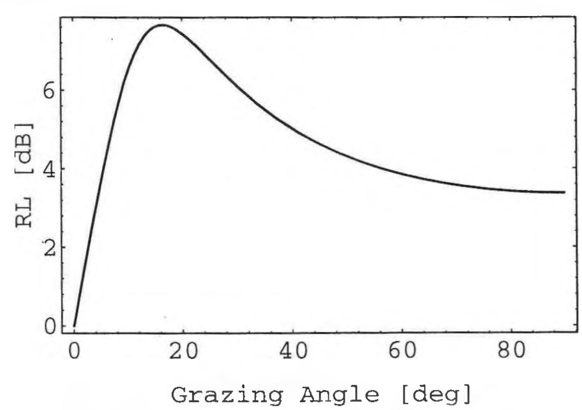

Figure 1 *Reflection loss (RL) of a locally-reacting surface with normalized impedance 2.5 2.5i.*

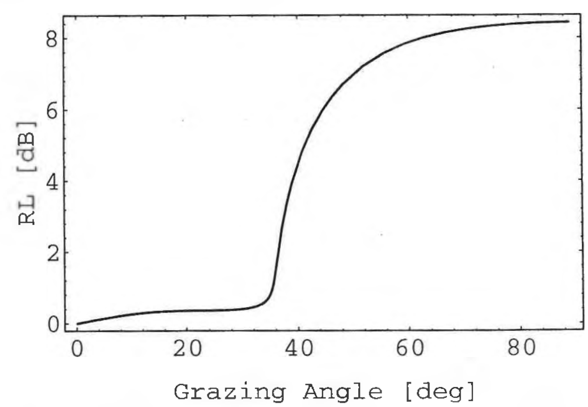

Figure 3 *Reflection loss (RL) of a sand layer, calculated using a visco-elastic solid model.*

SHALLOW WATER INVERSION BASED ON THE VERTICAL COHERENCE OF THE AMBIENT NOISE FIELD

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Past studies have shown how the spatial structure of the underwater ambient noise field can be linked to the acoustic properties of the ocean bottom. For example, Buckingham and Jones[1] have devised a way to estimate the compressional sound speed of the surficial sediment from the structure of the vertical directionality of the underwater noise field. Similar information can be obtained using the vertical noise coherence pattern.

This paper will present vertical coherence estimates from a shallow water model[2]. The model derives the noise intensity within the ocean from wind and wave noise sources, which are modelled as a distribution of dipole sources at the ocean surface. The coherence between two vertically separated sensors is obtained with a Fourier transform of the noise intensity as a function of vertical angle. The model uses the bottom reflectivity equations of Brekhovskikh[3] for a layered bottom. The ocean has an isovelocity profile.

The purpose of the study is to determine which ocean bottom parameters have a greater effect on the vertical noise coherence. For example, Fig. 1 shows the variation in reflection loss (as a function of grazing angle) for different bottom types. Bottom type A (solid line) represents a sand bottom typical of the Western Bank area south of Nova Scotia [compressional speed = 1650 m/s, shear speed = 260 m/s, compressional attenuation = 0.46 dB/λ, shear attenuation = 1.3 dB/λ, density = 1.8 g/cm^3]. The type B bottom (dashed line) is the underlying tertiary bedrock [compressional speed = 2000 m/s, shear speed = 800 m/s, compressional attenuation = 0.08 dB/λ, shear attenuation = 2.7 dB/λ, density = 2.2 g/cm^3]; type C (included for comparisons) is the same bottom as B, but with zero shear speed (long dashed line). All three bottoms were considered as half-spaces to produce Fig. 1.

The effect of the reflection loss on the vertical coherence pattern (real and imaginary part) is seen in Fig. 2 (the imaginary part is offset by -1). The line types are matched to those of Fig. 1. The coherence is plotted as a function of kd, where k is the wave number and d is the separation between sensors. The real part of the coherence is linked to the symmetrical component of the noise field, while the imaginary part is an indicator of large asymmetry in the vertical energy flux.

The effect of a change in compressional sound speed is seen by comparing the curves for bottom types A and C, for which the shear speeds are low. Since the harder bottom has a higher critical angle, the acoustic energy will be distributed over a wider range of angles in the vertical. At low shear speed, the imaginary part of the coherence is small, and the real part will decorrelate more quickly (over a shorter vertical distance) with a hard bottom.

The effect of the shear speed is seen by comparing the bottom types B and C. The difference is greatest over a specific range of low grazing angles (see Fig. 1). Over this range, part of the energy is converted from p-wave to s-wave, and the reflection loss gets stronger. This loss translates to a higher imaginary part of the coherence, and the real part decreases more quickly with vertical distance (or higher kd).

Changes in compressional or shear speed can have a strong effect on the vertical coherence pattern. Changes in attenuation, not shown here, have a much smaller impact on the coherence. The presence of layers in the bottom introduces a frequency dependence in the reflection loss and coherence patterns.

Some experimental data will be presented at the conference. It will be shown how the modelled coherence can be matched to the data to infer bottom structure.

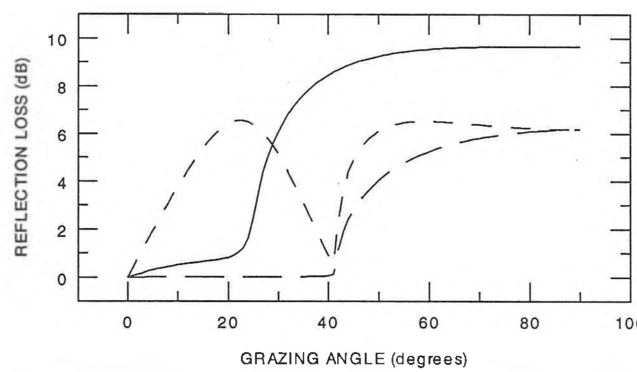

Figure 1 *Reflection loss versus grazing angle for bottom type A(————), B(— — —), C(—— ——).*

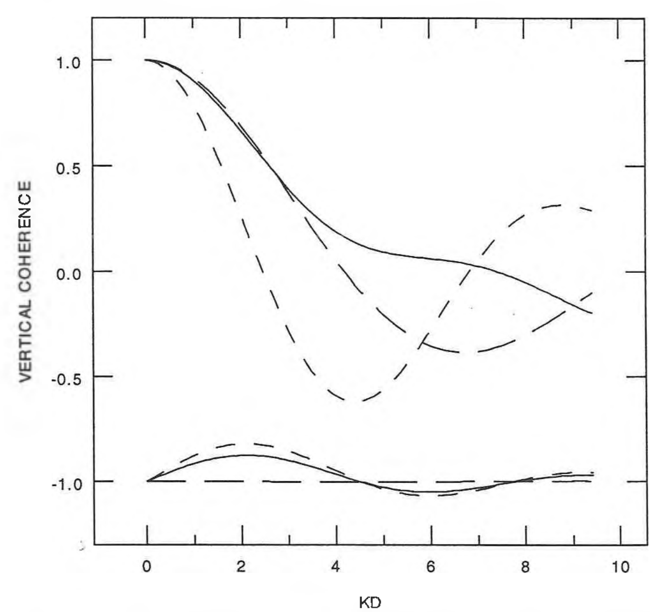

Figure 2 *Real part and imaginary part (offset by -1) of the vertical coherence for bottom type A(————), B(— — —), C(—— ——).*

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[1] M.J. Buckingham and S.A.S. Jones, "A new shallow-ocean technique for determining the critical angle of the seabed from the vertical directionality of the ambient noise in the water column," J. Acoust. Soc. Am. **81**, 938-946 (1987).

[2] D.M.F. Chapman, "Surface-generated noise in shallow water: a model", Proc. Inst. of Acoust. 9(4), 1-11 (1987).

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Matched Field Inversion in Underwater Acoustics

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Matched field inversion is a full wave inversion method based on matched field processing and efficient global search techniques. Matched field processing (MFP) is a model-based signal processing method that was originally introduced in underwater acoustics as an inversion technique for localization of a sound source in the ocean [1]. In its simplest form, MFP determines the source location by comparing measured acoustic fields with replica fields that are calculated for a known waveguide environment and a specified experimental geometry. The best match is obtained for the correct source position, if the environment is accurately modelled. Errors in the model of the enviroment are manifested as mismatch in the MF comparison process. In practice, a search procedure is carried out over a regular grid of possible source ranges, depths and bearings. However, MFP is a powerful full-wave inversion method, and it has more recently been applied to the general inverse problem of estimating the properties of the ocean waveguide itself. Matched field inversion has been applied to estimate quantities such as the sound speed profile in the water, under-ice roughness, and geoacoustic properties of the ocean bottom. The emphasis of this paper is on the last application, inversion for geoacoustic properties of the bottom.

Matched field inversion for geoacoustic properties is formulated as a global optimization problem, rather than a pure inverse problem. This approach is amenable to the addition of any prior knowledge of the environment. The components the MF inversion algorithm are very simple, consisting of: (1) a specific form for the geoacoustic model itself; (2) a numerical model for calculating the acoustic fields for a specific experimental geometry; (3) a cost function for assessing the measured and modelled fields; and (4) an efficient global search method. The form for the geoacoustic model is usually a layered earth model, with parameter value bounds that are based on any prior knowledge of the specific environment. The model parameters include the layer depths, and the density and sound speeds and attenuations (compressional and shear) for each layer. The replica acoustic fields can be calculated using normal mode methods for range independent problems, or parabolic equation techniques for range independent environments; in some cases, adiabatic normal mode methods may also apply for weak range dependence. The cost function for the global search process is based on a linear Bartlett matched field processor; this function is robust to mismatch and is more effective for use in a random search. Successful application of the method depends on the use of an efficient global search algorithm to search the multidimensional parameter space of candidate geoacoustic models. In practical problems, the number of model parameters can be large (8-10), so that regular grid searches are precluded. Two efficient global search algorithms are currently in use: simulated annealing [2] and genetic algorithms [3]. Both these methods have the ability to escape from local minima, so that the search process is not trapped by a false 'solution'. Although the inversion is sensitive to mismatch in the parameters that define the experimental geometry, these uncertainties can be tolerated in the inversion by including parameters such as range and array depth in the search process.

In this paper the MF inversion method is applied to data that were obtained using a vertical line array of hydrophones. The experiment was carried out in shallow water on the continental shelf off the west coast of Vancouver Island. Estimates were obtained for the parameters of a 2-layer elastic geoacoustic model, inlcuding the depth of the upper layer and the compressional and shear speeds of both layers. The estimated values were in good agreement with ground truth data that were obtained by conventional seismic methods.

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MEASUREMENTS OF SOUND FIELD STATISTICS NEAR THE GROUND WITH A LARGE OUTDOOR ARRAY

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Introduction

The statistics of a sound field propagating through the turbulent atmosphere outdoors near the ground depend upon the prevailing meteorological conditions, the ground topology and impedance, as well as the dynamics of the sound source. An understanding of these statistics is necessary to obtain effective sound field measurements, to realistically assess propagation models and codes, and design and predict the performance of acoustic remote sensing systems.

We explore some of these principles based on theory and measurements obtained with a large acoustic array developed at NRC. The array is unique measurement and research tool because of its large size (64 elements) and flexible geometry. The investigation distinguishes between line-of-sight propagation and the sound field within an acoustic shadow.

Coherence for Line-of-sight Propagation

Typical values for temporal coherence can exceed 10 seconds and tend to follow the relationship

$$t_0=(0.75C_n^2k_0^2R)^{-3/5}/V_\perp .$$

where C_n indicates the turbulence strength (and is of the order of 10^{-3}), R is the propagation range, and $V_\perp$ is the component of the wind velocity across the propagation path. Spatial coherence in the direction of propagation may be hundreds of meters but, transverse to this direction, it is on the order of a meter. The coherence between two tones separated in frequency by ω_0 and propagating simultaneously is essentially independent of the mean signal frequency. The frequency separation at which the coherence drops below 1/e varies as $R^{-1/2}$ at short ranges and as 1/R at longer ranges.

Coherence in an Acoustic Shadow Region

Temporal coherence in a shadow region follows

$$t_0=\sqrt{6}/2k_0\sigma_v\sin\theta$$

where k_0 is the source wavenumber, σ_v is the standard deviation of the wind speed (approximately 0.1 - 0.3 times the wind speed), and θ is half the scattering angle ($\sin\theta$ is typically about 0.1). Spatial coherence is much shorter in an acoustic shadow than for line-of-sight propagation. Within a shadow region, the coherence length in the direction of propagation may be as little as 10 meters and transverse to the direction of propagation it may be much smaller. Transverse coherence tends to be about twice the signal wavelength for higher frequencies. Frequency coherence decreases rapidly within the shadow zone boundary; signals separated by less than 20 Hz show little correlation at frequencies between 200 and 1000 Hz.

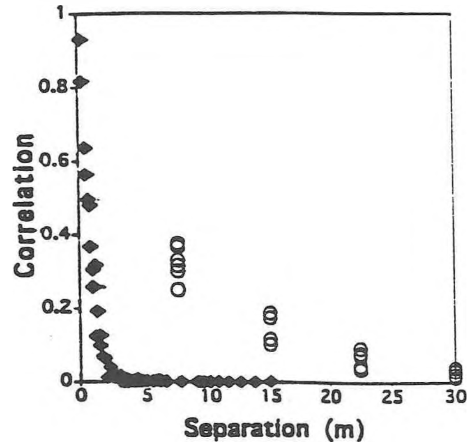

Figure 1 Spatial coherence in a shadow at 500 Hz, transverse (diamond) and longitudinal (circle) to the propagation direction.

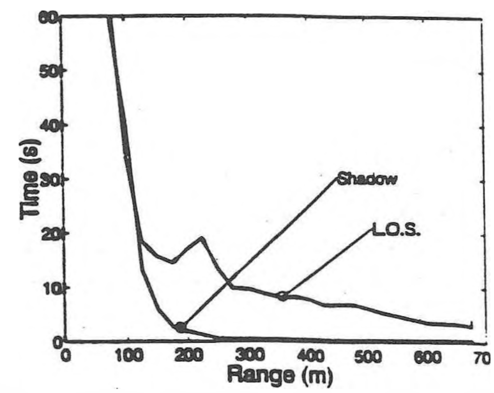

Figure 2 Temporal coherence for a 500 Hz tone for line-of-sight (L.O.S.) and acoustic shadow conditions.

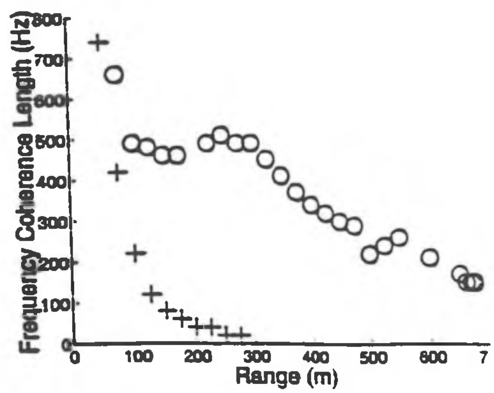

Figure 3 Frequency coherence length (bandwidth) for line-of-sight (circle) and acoustic shadow conditions (cross).

FACTORS AFFECTING ATMOSPHERIC SOUND PROPAGATION ABOVE AN IMPEDANCE SURFACE

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Predictions of sound propagation through the atmosphere are important in many areas. These areas include prediction of community and environmental noise from road, rail and air vehicles, and the detection and localization of acoustic sources at long distances. Many factors influence the propagation of sound through the atmosphere. Some of these factors are indicated in Fig. 1.

Heating of the ground by the sun leads to variations in the mean air temperature at different heights. The speed of sound depends on temperature, so the temperature profile will cause rays of acoustic energy to refract upward or downward, depending on the shape of the profile. The mean wind speed also varies with height, causing acoustic rays to bend upwards. These refractive effects lead to enhanced sound pressure levels near the ground in the case of downward refraction and reduced levels (acoustic shadows) for upward refraction.

The driving forces of the sun and wind, particularly when coupled with flow over varying terrain and uneven heating of the ground, generate random fluctuations in both temperature and wind flow, i.e., turbulence. Atmospheric turbulence scatters sound energy. Within the acoustic shadow arising from upward refraction or behind a barrier, the scattered energy can dominate the sound field.

When source and receiver are above the ground and not too far apart, the interference between direct and reflected sound paths can produce substantial cancellation at some frequencies. The resulting spectrum is controlled by the ground impedance. At longer ranges with near grazing incidence, the pressure reflection coefficient approaches $R = -1$: the reflected wave will interfere destructively with the direct wave over a broad range of frequencies, producing a diffractive acoustic shadow. For certain ground impedance regimes, surface waves can arise, spilling additional acoustic energy into shadow regions.

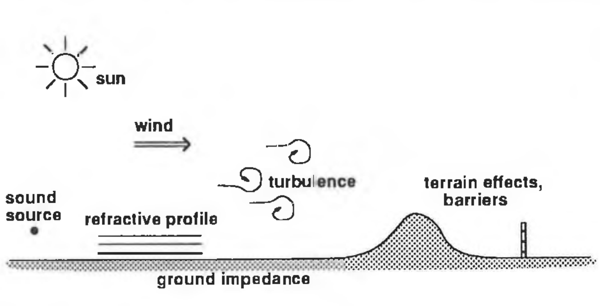

Figure 1. Sketch showing some of the factors that influence the propagation of sound through the atmosphere.

Noise barriers are often used to reduce the level of noise exposure. Procedures for computing insertion losses have been developed to deal with various barrier geometries. Propagation over real 3-dimensional terrain (e.g., hills, buildings), though, is a considerably more challenging problem for prediction.

Atmospheric attenuation reduces sound pressure levels beyond that expected solely because of geometrical spreading. Higher sound frequencies are attenuated much more strongly, so long range propagation is essentially a low frequency phenomenon. Variations in ground impedance, e.g., the transition from roadway to shoulder, are common and will affect propagation.

Different computational procedures can treat various combinations of factors; no procedure handles everything. The Greens function parabolic equation (GF-PE) approach can accommodate a refractive profile, turbulence, and ground impedance. Barrier and terrain effects are excluded, though. Fig. 3 shows a calculation for a 500 Hz tone propagating in an upward refracting sound speed profile. The source is 0.3 m above the ground (flow resistivity 200 c.g.s. rayl/cm) and receivers are on the ground. Calculations made with and without turbulence demonstrate the filling of the acoustic shadow with scattered energy.

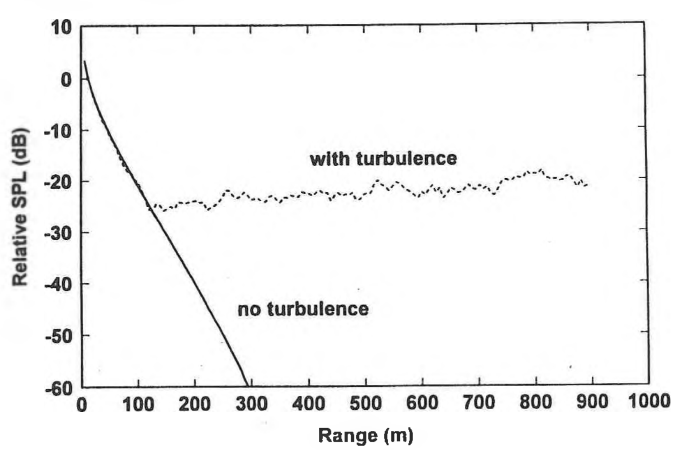

Figure 2. GF-PE prediction for a 500 Hz signal in upward refraction conditions, showing the effects of turbulence..

Acoustic image compression with the wavelet transform.

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Introduction
This investigation was motivated by acoustic images collected by a portable, autonomous, echosounding device and a Furuno echosounder. Marine and freshwater data were used to examine the feasibility of improving standard compression with the wavelet transform. A feature of these acoustic images is that they consist of reflected signals from both point and volume scatterers. The point scatterers appear as transient signals in the image, typically large, rapidly decaying spikes. The wavelet transform has generated much interest due to its ability to represent transient signals sparsely in the wavelet domain.

Discrete wavelet transform
The discrete wavelet transform is analogous to the fast Fourier transform although there are a few major differences. For example, the wavelet transform provides the optimal time-frequency window at all frequency scales. As well, the basis functions are not limited to the familiar sine and cosine functions of the Fourier transform. A number of wavelets have been derived which are efficient at representing all square-integrable functions [1]-[2]. Of these, two were chosen for this study: the DAUB4 and DAUB12 wavelets. The DAUB4 wavelet models linear, piecewise-continuous functions and DAUB12 provides continuity in higher derivatives.

Compression
The compression ratio is the size, in bytes, of the compressed image divided by the size of the original. Standard compression schemes fall into two categories. Lossless compression, a reversible process, is defined by the ability to recover the original image intact. Lossy compression sacrifices this requirement to provide a lower compression ratio. The choice of scheme is dependent on the intended use of the acoustic data.

Investigation
The pixels of the digital acoustic images were described by 16-bit integers. However, in each image, less than 16 bits of information was present. This implied an inherent compressibility. To allow a standard comparison, the amplitude range of each image was first shifted and scaled to cover the entire 16-bit range (0-65535). Each image was transformed into the wavelet domain with the DAUB4 and DAUB12 wavelets. The results were subsequently rescaled to cover the full 16-bit range, then compressed with a standard compression program. A greater degree of compression was also achieved through the removal of the least significant coefficients after the transform. These values were chosen arbitrarily at a factor below the peak value and ranged from 0.0 to 0.1, with the former corresponding to lossless compression (see sample, Figure 1). The value 10^{-5} is approximately the resolution of the 16-bit image.

The mean square error (MSE),

$$\bar{e} = \sum_{i,j} (f_{i,j} - o_{i,j})^2 \Big/ \sum_{i,j} o_{i,j}^{\,2} ,$$

where $f_{i,j}$ and $o_{i,j}$ are the pixel values in the uncompressed and original images, respectively, was calculated to provide a crude estimate of the amount of information loss. Compression ratios and MSE's for the different schemes are summarised in Table 1, below. The MSE for the original image is the round-off error associated with shifting and scaling.

Summary
The wavelet transform appears to be effective at compressing the acoustic images in this study for the reason that the signal is made "sparse" by the wavelet transform. This stems from the fact that these acoustic images consist of dominant information returned from point reflectors in the acoustic domain. These signals are "wavelet-like" and therefore may be amenable to modelling by wavelets.

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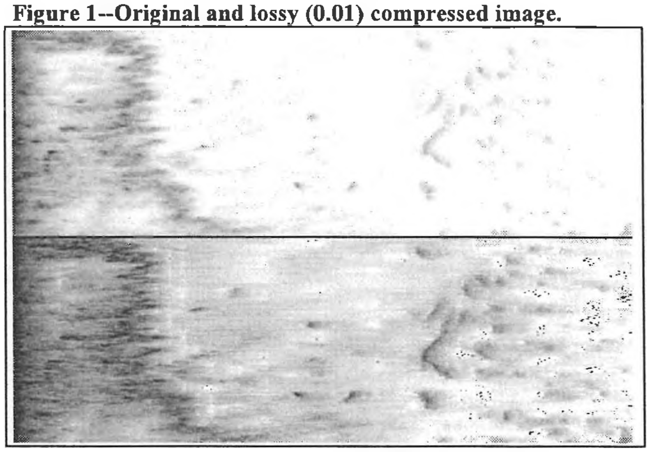
Table 1--Compression ratio and mean square error.

	DAUB4		DAUB12	
Image	Ratio	MSE	Ratio	MSE
Original	69.2%	6.39E-05	69.2%	6.39E-05
0.0	56.7%	2.79E-04	57.6%	2.69E-04
0.00001	56.6%	2.78E-04	57.6%	2.72E-04
0.0001	43.7%	7.66E-04	46.9%	6.59E-04
0.001	22.5%	5.25E-03	25.1%	4.47E-03
0.01	9.0%	3.46E-02	10.5%	3.05E-02
0.1	2.9%	1.98E-01	3.5%	1.75E-01

Figure 1--Original and lossy (0.01) compressed image.



INCREASING THE DEPTH CAPABILITY OF BARREL STAVE PROJECTORS

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1. Introduction

The Barrel Stave Projector (BSP), an underwater acoustic transducer, has been under development at DREA since the mid-eighties[1]. It is a flextensional transducer where the driving element is a stack of piezoelectric ceramic rings. The extensional vibrations of the ceramic stack are coupled to a concave shell that vibrates in flexure. The leverage action of the shell amplifies the motion of the stack. To operate in the low frequency regime (<2 kHz) it is necessary to reduce the hoop stiffness of the shell by forming the shell from staves separated by small gaps, 1 mm wide or less. The shell is covered by a neoprene boot to waterproof the transducer.

All flextensional transducers with air-backed shells have a common problem - their performance is very depth dependent not to mention the higher tendency of being crushed when submerged to great depths.

2. The pressure compensation system

In the BSP the volume behind the shell is usually occupied by air at atmospheric pressure. As the projector is lowered to progressively greater depths, the hydrostatic pressure on the outside of the shell increases. This may cause the rubber outerlayer to be pushed in the gaps between the staves and/or a deformation of the shape of the stave. Either of these mechanisms would cause a change in acoustic performance of the projector. When the differential pressure across the shell exceeds a certain threshold, the flexural mode of the staves becomes clamped. To overcome this, the BSP can be filled with a fluid to equalize the interior with the exterior ambient pressure. The compressibility of the fluid used must be about the same as air or better to maintain the performance of the projector. Otherwise the driver's force is expended more in compressing the interior fluid than generating the exterior water motion to produce radiation. Pressurized air is the natural choice. The added air inside the projector will still have some side effects on the performance of the projector. The compliance of the pressurized air is given by: $C=V/\rho_{air}c^2$ where V is the volume of air in the projector, ρ_{air} is the density and c is the sound speed in air. The density is related to the pressure P and temperature T by: $\rho_{air} = \rho_0(T_0/T)(P/P_0)$ where ρ_0 is the density at absolute temperature T_0 and pressure P_0. As the depth is increased, ρ_{air} will increase causing the compliance to decrease or the stiffness to increase and causing an increase in resonance frequency.

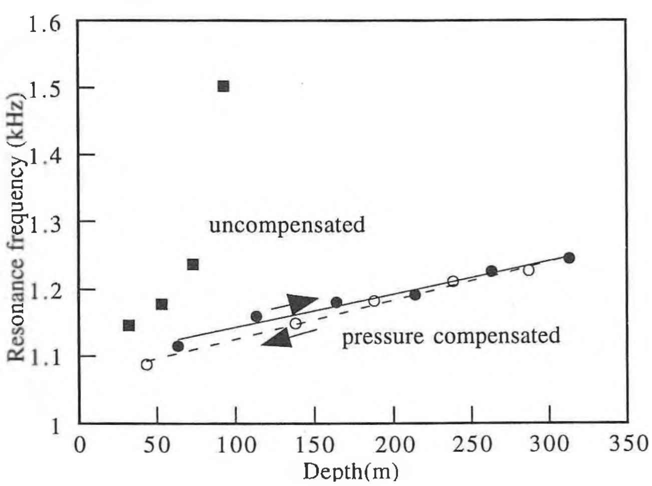

Figure 1 *The resonance frequency as a function of depth for a Barrel Stave Projector with and without pressure compensation.*

3. Calibration and Results

Several Sparton of Canada BSP Model 03BA1100[2] were refurbished to accommodate an active pressure compensation system. The system is essentially modified scuba gear as used by divers. High pressure tubing connects the air bottles to a dual-stage regulator which in turn is linked to the BSP via Tygon™ tubing. The projector has been equipped with a spigot on the bottom end plate. This end plate also has a channel milled out to allow the pressurized air to fill the area between the ceramic stack and the staves.

The transmitting voltage response and the admittance were measured for several Barrel Stave projectors without any compensation at various depths. The dependence of the resonance frequency, f_R, on depth is shown by the filled squares in Figure 1 for one of the projectors tested. As the depth increases f_R increases quickly. In fact at the deeper depths the lowest mode is no longer the flexural mode of the shell but the higher frequency longitudinal mode of the stack. For the BSP this dependency had previously been observed[3].

The same measurements were repeated with the projector equipped with pressure compensation. The second curve in Figure 1 shows the dependence of f_R on depth. The arrows indicate whether the projector was lowered to deeper depths or brought back up. The flexural mode of the projector remains active but increases at a rate of 0.57Hz/m as the depth is increased. This frequency increase is due to the increase of the density of air inside the projector. As the BSP is moved to shallower depths, f_R decreases at a rate of 0.43Hz/m. The difference in the rate of change of f_R is attributed to the inherent hysteresis of the regulator. The peak of the transmitting voltage response changes very little as the depth is increased. All the projectors tested showed a rate of change of f_R between 0.4 and 0.6Hz/m.

4. Conclusions

For many applications underwater acoustic projectors are required to work in the low frequency regime and at great depths. The Barrel Stave Projector is a compact high power low frequency projector and we have shown that by pressure compensating it with compressed air it can also be used in deep water applications.

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PE-ING IN AIR AND UNDERWATER

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1 INTRODUCTION

Numerical predictions based on the parabolic equation (PE) approximation are routinely used to model sound propagation in air and underwater. The main rationale for this is that accurate full-wave solutions to the PE can be computed efficiently using marching algorithms for both depth- and range-dependent inhomogeneous media. The development of the PE method has reached the point where finite-difference implementations derived from Padé series expansions can provide accurate solutions to one-way wave propagation for realistic geoacoustic conditions, e.g., over variable-depth bathymetry in the sea or variable-elevation topography in air. Moreover, with the introduction of both exact and approximate PE procedures for handling elastic media and rough-surface boundaries, the physics of shear wave propagation and forward-scattering can readily be accommodated.

2 THEORY

In two dimensions (r, z), (z positive down), the outgoing spatial component of the acoustic pressure $p\exp(-i\omega t)$ can be recovered from the field $\psi = p\exp(-ik_0 r)\sqrt{k_0 r}$ that satisfies the higher-order Padé PE [1]

$$\frac{\partial\psi}{\partial r} = ik_0 \sum_{j=1}^{J} \frac{a_{j,J}(\varepsilon+\mu)}{1+b_{j,J}(\varepsilon+\mu)}\psi. \tag{1}$$

Here $\varepsilon = N^2 - 1$, $\mu = k_0^{-2}\rho\partial_z\left(\rho^{-1}\partial_z\right)$, $k_0 = \omega/c_0$, $N = n(1+i\alpha)$, $n = c_0/c$ and ρ, c and α denote the density, sound speed and absorption, respectively. Although real-valued Padé coefficients $a_{j,J}$ and $b_{j,J}$ are known in analytical form [1], it is convenient for some applications to use complex-valued coefficients which must be determined numerically [2]. Using the method of fractional steps and the Crank-Nicolson finite-difference procedure, Eq. (1) is efficiently solved at each range step Δr as a sequence of J systems of tri-diagonal equations.

3 EXAMPLE

To illustrate the capability of Eq. (1), we consider the deterministic rough-surface test case examined at a recent Reverberation and Scattering Workshop [3]. Instead of forcing the PE to accommodate a non-flat pressure-release boundary, we modified the original problem by appending an air-layer backing to the region above the rough surface. By this maneuver, scattering by an external pressure-release boundary was replaced with scattering by an internal fluid/fluid interface across which the usual boundary conditions on the acoustic field apply. The large impedance drop across the ocean/air interface ($\approx 2\cdot 10^{-4}$) results in nearly perfect, out-of-phase reflection of sound for a water-borne source. A gaussian-tapered beam ($f = 400$ Hz) was steered upwards toward the surface at an angle of 10° to the horizontal. The full-field result for $|p|$ obtained using Eq. (1) is shown in Fig. 1 for a 20-m air layer backing and $J = 2$. The rough surface clearly scatters sound to steeper angles. This forward-scattered PE result agrees almost exactly with a reference solution obtained using an integral equation method [3].

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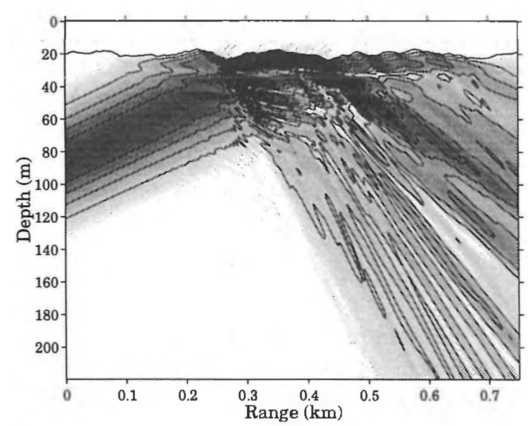

Figure 1: Forward scattering from a rough surface.

Modal Decomposition of Ocean Acoustic Fields Using Damped Least-Squares Inversion

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Long-range propagation of acoustic pressure fields in the ocean is often well modelled as a discrete set of propagating normal modes

$$p(r,z) = b \sum_{j=1}^{M} \phi_j(z)\,\phi_j(z_s)\,\frac{e^{ik_j r}}{\sqrt{k_j r}}\,,$$

where $p(r,z)$ is the (complex) pressure at range r and depth z, M is the number of modes, ϕ_j and k_j are the mode functions and wavenumbers, respectively, z_s is the source depth, and b is a complex constant. In this case, the acoustic field measured at an array of sensors can be decomposed into its modal components providing the basis for matched-mode processing techniques. The modal summation can be written as a linear matrix equation

$$\mathbf{A}\,\mathbf{x} = \mathbf{p},$$

where $\mathbf{A}$ is the mode matrix, $\mathbf{x}$ represents the modal excitations, and $\mathbf{p}$ is the pressure measurements. For example, for a vertical array of N sensors

$$\mathbf{p} = [p(z_1), \ldots, p(z_N)]^T,$$

$$\mathbf{A} = b \begin{bmatrix} \phi_1(z_1) & \ldots & \phi_M(z_1) \\ \vdots & \ddots & \vdots \\ \phi_1(z_N) & \ldots & \phi_M(z_N) \end{bmatrix},$$

$$\mathbf{x} = \left[\phi_1(z_s)\frac{e^{ik_1 r}}{\sqrt{k_1 r}}, \ldots, \phi_M(z_s)\frac{e^{ik_M r}}{\sqrt{k_M r}}\right]^T.$$

The corresponding expressions for a horizontal array are somewhat more complicated and are range dependent.

For an overdetermined system ($N > M$), the least-squares solution is obtained by minimizing the squared error

$$\psi_{\mathrm{ls}} = [\mathbf{A}\,\mathbf{x} - \mathbf{p}]^\dagger [\mathbf{A}\,\mathbf{x} - \mathbf{p}]$$

to yield

$$\mathbf{x}_{\mathrm{ls}} = [\mathbf{A}^\dagger \mathbf{A}]^{-1} \mathbf{A}^\dagger \mathbf{p},$$

where † indicates conjugate transpose. For a vertical array which densely samples the water column, the mode matrix $\mathbf{A}$ is approximately orthogonal, and the inversion is straightforward. However, for vertical arrays which poorly sample the water column or for horizontal arrays, $\mathbf{A}$ is non-orthogonal, and $\mathbf{A}^\dagger\mathbf{A}$ can be ill-conditioned, leading to instablility and poor results for noisy data. This difficulty is sometimes addressed by carrying out a pseudo-inversion of $\mathbf{A}^\dagger\mathbf{A}$ using singular value decomposition and deleting the smallest singular values in an ad hoc manner.

The method of damped least-squares (DLS) provides a regularized inversion with a rigorous approach to controlling the level of misfit. In its most general form, the method is based on minimizing a functional

$$\psi_{\mathrm{dls}} = [\mathbf{G}\,(\mathbf{A}\,\mathbf{x} - \mathbf{p})]^\dagger\,[\mathbf{G}\,(\mathbf{A}\,\mathbf{x} - \mathbf{p})] + \theta\,(\mathbf{H}\,\mathbf{x})^\dagger(\mathbf{H}\,\mathbf{x}).$$

The first term represents the data misfit, the second is a regularizing term, and θ is an arbitrary parameter which controls the trade-off between the two terms. $\mathbf{G}$ and $\mathbf{H}$ represent weighting matrices for the data residuals and modal excitations, respectively. Typically, for data with uncorrelated noise, $\mathbf{G}$ is taken to be

$$\mathbf{G} = \mathrm{diag}\{1/\sigma_1, \ldots, 1/\sigma_N\},$$

where σ_j is the standard deviation of the jth datum. $\mathbf{H}$ can be chosen arbitrarily to minimize different combinations of the excitations (or differences between excitations), providing flexibility in determining the character of the solution. The DLS solution is given by

$$\mathbf{x}_{\mathrm{dls}} = [(\mathbf{G}\,\mathbf{A})^\dagger \mathbf{G}\,\mathbf{A} + \theta\,\mathbf{H}^\dagger\mathbf{H}]^{-1}\mathbf{A}^\dagger\mathbf{G}^\dagger\mathbf{G}\,\mathbf{p}.$$

The trade-off parameter θ is chosen so that the (noisy) data are fit to a statistically meaningful level, e.g., to achieve a χ^2 misfit of

$$\chi^2 = [\mathbf{G}\,(\mathbf{A}\,\mathbf{x}_{\mathrm{dls}} - \mathbf{p})]^\dagger\,[\mathbf{G}(\mathbf{A}\,\mathbf{x}_{\mathrm{dls}} - \mathbf{p})] = 2N$$

for N complex equations. Since χ^2 is a monotonically increasing function of θ, an appropriate value of θ can be determined efficiently using Newton's method. DLS can also be modified to determine the smallest-deviatoric solution, i.e., the solution $\mathbf{x}_{\mathrm{sd}}$ which deviates minimally from an arbitrary reference vector $\mathbf{x_O}$. Defining $\mathbf{x} = \mathbf{x_O} + \delta\mathbf{x}$, the modal equations can written

$$\mathbf{A}\,\delta\mathbf{x} = \mathbf{p} - \mathbf{A}\,\mathbf{x_O} \equiv \mathbf{p_O}.$$

Applying the DLS formalism leads to

$$\mathbf{x}_{\mathrm{sd}} = \mathbf{x_O} + [(\mathbf{G}\,\mathbf{A})^\dagger\mathbf{G}\,\mathbf{A} + \theta\,\mathbf{H}^\dagger\mathbf{H}]^{-1}\mathbf{A}^\dagger\mathbf{G}^\dagger\mathbf{G}\,\mathbf{p_O}.$$

The characteristics and potential advantages of DLS inversion for modal decomposition will be illustrated and discussed in this paper.

Acoustic Localization of Hydrophone Array Elements in the Arctic Ocean

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BACKGROUND

In the Spring of 1996, the Esquimalt Defence Research Detachment of the Defence Research Establishment Atlantic and the US Naval Research and Development laboratory deployed an acoustic research array (ARA) through 4 m of polar pack ice near the edge of the continental shelf of the Arctic Basin, north of Ellesmere Island [1]. The ARA is configured as a 2.4-km horizontal line array (HLA) of 80 hydrophones on the seafloor, and two vertical line arrays (VLA's) of 20 hydrophones which span the water column (~560 m), one located at each end of the HLA. A 240-m secondary horizontal array of 8 hydrophones is also included perpendicular to the main HLA to resolve ambiguities about the main array axis. Acoustic data from the ARA are transferred 180 km via a seafloor fibre optic cable to a land-based recording recording facility at CFS Alert. Data are to be recorded continuously for the battery life of the system (approximately five years).

The three-dimensional nature of the ARA is ideally suited to advanced acoustic signal processing methods such as matched-field inversion. These methods require precise knowledge of the location of individual sensors in the arrays. This paper describes and analyzes a series of acoustic measurements carried out shortly after deployment of the ARA to localize the VLA and HLA sensors. Because of the difficulty of making acoustic measurements through the pack ice and the continual possibility of ice motion, only a minimal data set was recorded. Inversion methods are applied to obtain the most precise localization possible from this limited data.

VLA ELEMENT LOCALIZATION

Due to the effects of ocean tides and currents, the shape of a VLA suspended in the water column is not fixed but can change with time. The shape of each VLA of the ARA is continuously monitored using three high-frequency engineering hydrophones positioned at intervals along the VLA cable. The engineering hydrophones are localized acoustically using travel-time measurements from four transceivers (acoustic receive/transmit units) located on the seafloor, one in each quadrant about the VLA. For optimum array-processing performance, the position of the engineering sensors are required to a precision of 1 m. To meet this requirement, the position of the transceivers must be known to within < 1 m. The transceivers themselves were localized by recording their signals on hydrophones deployed just below the ice in the VLA and transceiver deployment holes. The transceivers were activated from a control transducer co-located with one of the surface hydrophones to provide a reference for absolute timing. Ideally, transponder localizations are based on a vastly over-determined data set from a symmetric set of source-receiver configurations to minimize the effect of experimental errors, e.g., [2]. In the present case, only five independent (non-symmetric) travel-time measurements were available to determine three spatial coordinates for each transceiver. A numerical sensitivity study was carried out prior to the experiment to appraise possible sources of error, and care was taken to minimize these errors in the field.

HLA ELEMENT LOCALIZATION

A simple experiment was carried out to localize the HLA hdyrophones based on deploying small impulsive sources (imploding glass light bulbs) at a series of locations surrounding the HLA. Shot instants were not independently measured, so the recordings represent relative rather than absolute travel times. Hence, shot instants must be included as unknowns in the inverse problem; this results in a coupled problem which must be solved for multiple hydrophone locations and shot instants simultaneously. Since refraction effects due to variations in ocean sound speed with depth are significant over the maximum propagation ranges, these coupled equations are non-linear. The system is solved using linearization and iteration, with ray tracing methods employed to calculate numerical partial derivatives.

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SINGLE AND PARALLEL BARRIER INSERTION LOSS BY MEANS OF IMPROVED DIFFRACTION BASED MODELS

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Introduction

Lam[1] recently introduced an improved diffraction based method for calculating the insertion loss of a finite length, three dimensional barrier. Comparisons of this new method to experimental scale modeling and wave based boundary element method showed good comparisons over a broad frequency range. This preliminary work of Lam is extended in this paper to include the modeling of two dimensional geometries and other diffraction models, namely the models of Kurtz and Anderson[2], and Pierce[3]. The work is also extended to include the consideration of parallel barriers and the modeling of finite ground impedance.

Lam's diffraction based method for calculating the insertion loss of a finite barrier considers the phase interrelation of each of the minimum diffraction paths from the source to receiver. The diffraction model used with Lam's method was Maekawa's curve. This method considers only the amplitude change of sound at the edge of the barrier. The same can be said for Kurtz and Anderson's diffraction model. Pierce's diffraction model, on the other hand, predicts the amplitude change and a phase shift at the edge of the barrier, hence providing a more accurate prediction model. Figure 1 compares insertion loss of a finite width barrier as predicted by BEM and the diffraction models.

2D Modeling

Lam's method was extended to consider two dimensional geometries for single and double barriers. Results compare well with the boundary element method. In modeling parallel barriers, Pierce's diffraction model was once again found to be superior. An impedance plane model was included for two dimensional geometries. The model incorporated was that of Hothersal's[4]. Comparisons with finite element modeling are shown in Figure 2. The results compare well.

Discussion

Accurate 2D and 3D boundary element models require extremely large amounts of computer memory and computational time. Simple diffraction based models, like those discussed here, take only a fraction of the computational time and require minimum computer resources. This enables these methods to be used as an accurate, efficient design tool to conduct frequency response tests, both discrete and octave band and even full contour mapping over large regions.

Continuing Work

To this point, all the models have considered a uniform atmosphere. This greatly overpredicts the performance of barriers. Work is currently being conducted to consider the effects of a non-uniform atmospheric medium, such as a wind/temperature gradients and turbulence on barrier performance.

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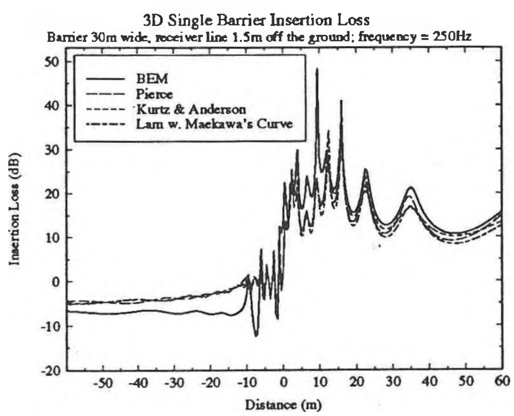

Figure 1.

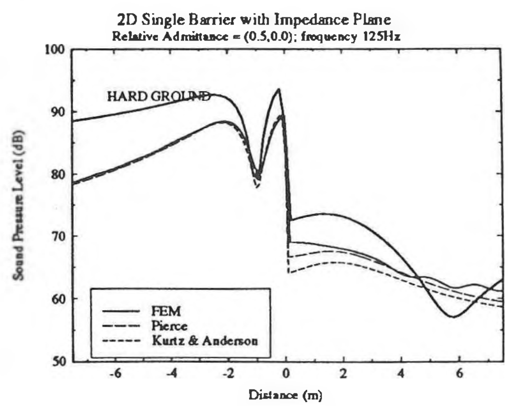

Figure 2.

Efficient tracking of a moving source

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1 INTRODUCTION

Matched-Field Processing (MFP) has been used to process underwater acoustic data for many years and has recently been proposed for use in radar. In MFP the measured data is matched against predictions to produce an "ambiguity" surface which contains the matches for all possible source positions in a search region. Consequently the value of the ambiguity surface at any point reflects the likelihood that the source is at that position in the search region. This paper describes an efficient algorithm for tracking an acoustic or electromagnetic source in scenarios in which the source position is ambiguous during any single look. The source is assumed to be moving at constant speed in a two or three dimensional region and its track is a concatenation of linear and circular subtracks. Such an algorithm enables low signal-to-noise (SNR) ratio targets to be detected or localized by the tracker when the source is not detectable or can not be localized on a single snapshot.

2 THEORY

The algorithm finds the statistically significant subtracks formed by joining the highest peaks on pairs of ambiguity surfaces. Such an algorithm is much more efficient than exhaustive ones which find tracks through all possible source positions [1]. For the scenario in this study, 20 km $\times$ 20 km $\times$ 100 m, 10^{34} tracks would be examined with an exhaustive algorithm. It has been shown for an efficient algorithm that detectable source tracks will be examined with high probability [2]. Weighting the matches at each position in proportion to the expected signal level enables detection of sources with track SNRs as low as 10 dB [1].

3 EXAMPLE

During PACIFIC SHELF 93 narrowband acoustic data at 45 Hz was collected on a 16 element vertical line array off Vancouver Island in range-dependent shallow-water a few hundred meters deep. This data was employed to form ambiguity surfaces in a range and depth wedge on 30 bearings covering the ocean volume through which the source passed. A 3-dimensional tracker that examines linear tracks, including diving tracks, was applied to the 3D ambiguity volumes formed from the 30 range-depth surfaces at each time interval. As seen in the figure the highest tracks, in the range-depth wedge 0-16 km $\times$ 34° $\times$ 100 m, approximated the source track from GPS and radar which has a 200 m range and 10 m depth uncertainty. This agreement was obtained despite a very limited knowledge of the geoacoustic properties of the environment and bathymetry.

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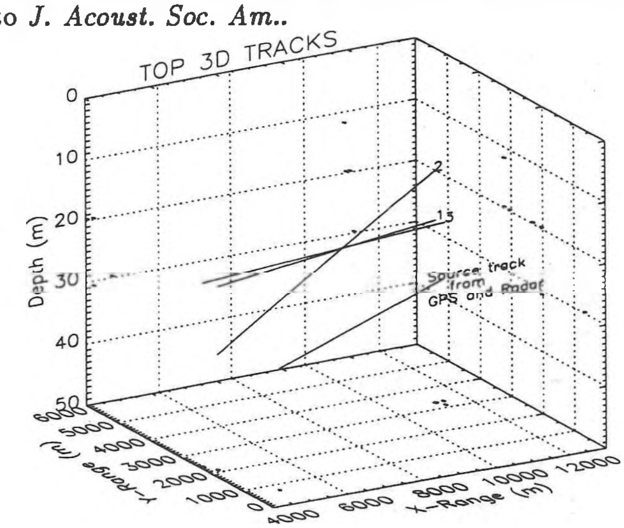

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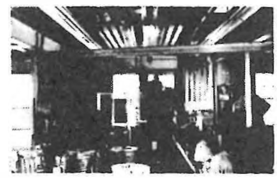

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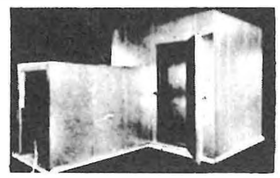

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NOISE AND VIBRATION CONTROL FOR A NATURAL-GAS-FUELED ELECTRICAL COGENERATION SYSTEM LOCATED DIRECTLY ABOVE PENTHOUSE RESIDENTIAL SUITES IN A HIGH-RISE APARTMENT BUILDING

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INTRODUCTION

In some buildings it has become economical to provide continuously operating electrical power generators, driven by internal combustion engines fueled with natural gas, to reduce the demand for electrical power from hydro utilities. This paper describes the remedial work that was undertaken during commissioning of one such cogeneration system, located at the top of a residential apartment building, to reduce the transmitted noise to an acceptable level in the penthouse suites located directly below. The work consisted of retrofitting concrete inertia bases under the generator sets and supporting these bases on large open springs. Noise and vibration test data from measurements taken before and after completion of the remedial work are presented to document the success of the project.

SYSTEM DESCRIPTION

Two 65 kW cogeneration units operating at 1800 RPM were installed in a mechanical room at the top of an apartment building. Noise from the engines was clearly audible in two residential penthouse suites located directly below. A site inspection revealed that the the units had been provided with acoustic enclosures and the motor/generator assemblies were vibration-isolated from the base of the enclosure with housed-spring isolators having a nominal static deflection of 1 inch. There was no vibration-isolation between the base of the acoustic enclosure and the floor slab. The adjustable snubbers in the housed-spring isolators were snug and as a result the isolation effectiveness of the springs was reduced. The measured noise level in the bedroom of one of the penthouse suites, before any remedial work, is shown in Fig. 1. The maximum measured vibration level on the mechanical room floor near the units was -52 dBg at 60 Hz. The owner requested that the existing isolators be properly adjusted and that neporene isolation pads be inserted under the acoustic-enclosure base frames. The noise level was reduced noticeably in the penthouse suite after the isolators were adjusted but was still audible. There was no further improvement when the neoprene pads were inserted under the units.

DESCRIPTION OF REMEDIAL WORK

The remedial work consisted of raising each of the acoustic enclosures and installing large open springs under brackets fitted to the sides of the steel bases. To make the isolation efficiency of the modified system higher, each steel base was filled with concrete before being raised. The new spring isolators were selected for a nominal 3-inch static deflection under the actual load. Fig. 1 shows the results of the noise measurements taken after completion of the remedial work. The NC-30 noise criterion curve, which was used as the design goal for an acceptable noise level in the bedrooms, is also shown in Fig. 1. The vibration level on the mechanical room floor after the remedial work was -69 dBg at 60 Hz.

CONCLUSIONS

From the case study described in this paper, it can be concluded that cogeneration systems driven by internal combustion engines can be successfully installed above residential suites if the proper noise and vibration control methods are implemented.

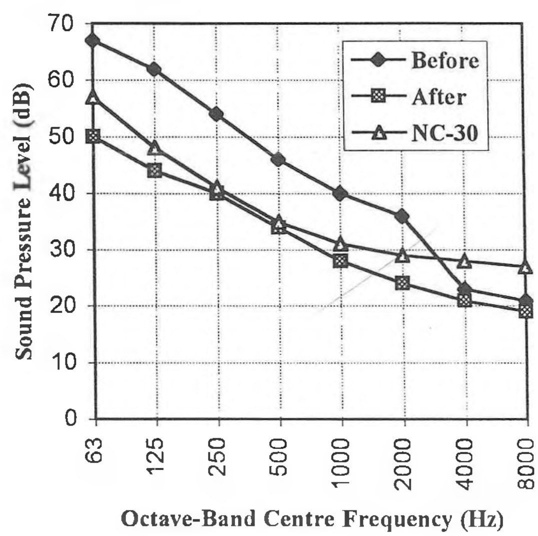

Fig. 1: Sound Measurements Taken Before And After Remedial Work.

EXPERIMENTAL EVALUATION OF SIMPLIFIED MODELS FOR PREDICTING NOISE LEVELS IN INDUSTRIAL WORKROOMS

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Introduction

Simplified models for predicting industrial noise levels have the advantage of simplicity and negligible calculation time over more generally-applicable and comprehensive methods such as ray tracing. Simplified models predict octave-band and/or A-weighted steady-state sound-pressure level as a function of distance from a single omnidirectional sound source of known output power level. The industrial workroom is typically described by model parameters describing the room geometry, surface absorption and contents (fittings - the numerous machines, stockpile, benches etc. in the workroom).

In recent work [1], existing simplified models for predicting noise levels in industrial workrooms were reviewed and critiqued. Models developed by Embleton & Russell (E&R - this is a Canadian Standard), Friberg, Kuttruff, Sacerdote, Sergeev, Thompson *et al*, Wilson and Zetterling were considered. Full references to the papers presenting these models, full details of the models themselves and a conceptual critique of the models are given in [1]. Most models were found to be conceptually inadequate; for example, some ignored a key parameter - the fittings. Preliminary attempts were subsequently made to develop an improved model (the Hodgson model)[2].

In the present work, these nine simplified models were evaluated by comparing predicted sound-propagation curves - SP(r), the variation with distance r of the sound pressure level L_p minus the source sound power level L_W - with those measured in empty and fitted industrial workrooms with and without absorptive surface treatments.

Workrooms

The study involved thirty industrial workrooms. All were of typical modern construction, with a steel-deck roof, concrete floor, masonry/glazing/metal-cladding walls, and horizontally-uniform fitting distribution. The workrooms were in four categories: empty (10), empty with absorbent surface treatment (5); fitted (10); fitted with absorbent surface treatment (5). The absorbent treatments consisted of various sound-absorbing materials applied to or suspended from the ceiling and/or all or part of the walls.

Test and Prediction Procedures

In each workroom sound-propagation curves were measured in octave bands from 125-4000 Hz. A calibrated omnidirectional loudspeaker array was placed near one end wall at half width. Steady-state sound-pressure levels were measured at convenient distances along the workroom at half width. Reverberation times were also measured. All models were programmed using spreadsheets. Parameters originally presented in the form of curves were predicted using equations determined by regression techniques.

Input data was estimated for each workroom. Surface absorption coefficients were estimated from measured reverberation times using diffuse-field theory. Fitting parameters were estimated from a knowledge of the fittings involved and from experience.

Each of the 9 models was used to predict SP curves for each of the 30 workrooms. 125-4000 Hz octave-band and/or A-weighted total levels were predicted as applicable. Predicted and measured results were compared using plots and statistics.

Results

Figs. 1 and 2 show the A-weighted or 1000-Hz octave-band results for untreated typical empty and fitted workrooms, respectively. The main conclusions can be generalized as follows:

In empty workrooms, the Embleton & Russell, Friberg, Sacerdote and Wilson models significantly underestimate levels in most cases. The Hodgson, Sergeev, Thompson et al and Zetterling models perform quite well.

In fitted workrooms the Embleton & Russell, Kuttruff, Sacerdote, Sergeev, Thompson *et al* and Wilson models significantly underestimate levels in most cases; the Thompson *et al* model overestimates levels. The Friberg, Hodgson and Zetterling models perform quite well.

Only the Hodgson model performed very well in most cases; this is not surprising since it was developed from the same workroom SP(r) data used in this evaluation work.

[1] M. R. Hodgson, "Review and critique of existing simplified models for predicting factory noise levels", *Canadian Acoustics* **19**(1) 15-23 (1991).

[2] M. R. Hodgson, "Preliminary simplified models for predicting sound-propagation curves in factories", *Canadian Acoustics* **20**(3) 37-38 (1992) and Erratum, **20**(4) 19 (1991).

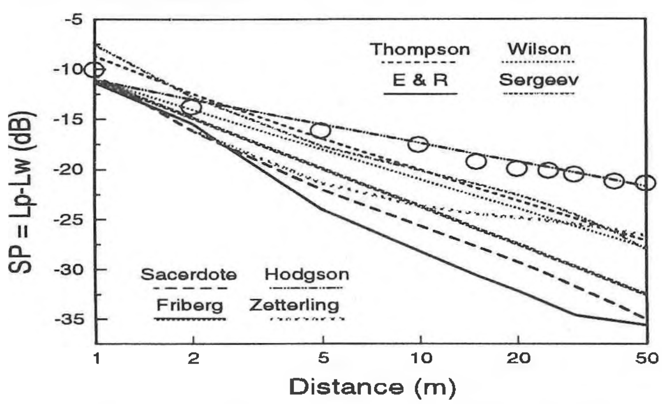

Fig. 1. A-weighted or 1000-Hz sound-propagation curves in an untreated empty workroom as (O) measured and as predicted by eight simplified models.

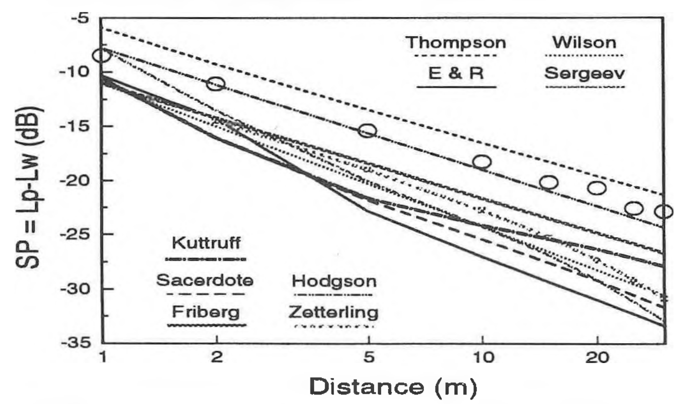

Fig. 2. A-weighted or 1000-Hz sound-propagation curves in an untreated fitted workroom as (O) measured and as predicted by eight simplified models.

IMPROVED METHODS FOR ESTIMATING FITTING DENSITY AND FITTING ABSORPTION COEFFICIENT IN INDUSTRIAL WORKROOMS

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Introduction

When predicting noise in industrial workrooms, a major factor that must be taken into consideration is the presence of 'fittings' - obstacles such as machines and stockpiles - in the room. Besides the fitting spatial distribution, there are two important parameters used in prediction models to describe the fittings. One is the fitting density - a measure of the number of fittings and of the average fitting scattering cross-sectional area - and the other is the fitting absorption coefficient. While ranges of typical fitting densities and absorption coefficients are known, no reliable method exists for measuring or estimating them in a given case. Furthermore, theoretical expressions for calculating fitting density assume small fittings and high frequency. In particular, Kuttruff proposed - assuming spherical fittings - that:

$$Q_0 = S_{tot} / 4\,V \qquad (1)$$

in which Q_0 is the Kuttruff fitting density in m^{-1}, V is the volume of the fitted region in m^3, and S_{tot} is the total surface area of the fittings in m^2. Lindqvist [2] corrected for the possibility of overlap of fittings of dimension D_f; the Lindqvist fitting density is:

$$Q_L = Q_0\,[1 + (8\,Q_0\,D_f / 3\,\pi) - Q_0^2\,D_f^2 / 2] \qquad (2)$$

When the fittings are large, fitting density will increase by 5-10 % after correction by the Lindqvist formula. The aim of the research discussed here was to develop and test improved methods for determining fitting density and absorption coefficients in industrial workrooms.

Correction for Large Fitting Dimensions

By considering the possibility of a third fitting blocking the path between two others, a formula was derived for calculating the fitting density in the case of large fitting dimension:

$$Q' = [(1 / a\,Q_0) - 2\,D_f + 2\,Q_0\,D_f^2]^{-1} \qquad (3)$$

in which $a=1+(8Q_0D_f/3\pi)-Q^2{}_0D_f^2/2$ is the Lindqvist correction factor in Eq. (2) and Q_0 is calculated by Eq. (1). The effect of the new correction depends on D_f and Q_0. If Q_0=0.1 m^{-1} and D_f=1 m, the fitting density will increase to 0.122 m^{-1}, which is 1.22 Q_0. Again, Eq. (3) is only valid at high frequency.

Frequency-Varying Fitting Density

By considering the various contributions to the total energy at a receiver position in empty and fitted regions when the source/receiver line is either blocked or not blocked by fittings, the fitting density is found to depend on three measurable quantities which all vary with frequency:

$$Q(f) = -\,(1/r)\,\ln\,\{1 - [(E_{nb}(f) - E_t(f)] / E_{Q=0}(f)\} \qquad (4)$$

in which r is source/receiver distance, E_{nb} is the sound energy for the case when there is no fitting blocking the direct sound, E_t is the total sound energy, and $E_{Q=0}$ is the sound energy in a free field.

Measurements were made, in the 125-4000 Hz octave bands, of average values of these quantities at a number of receiver positions in an anechoic chamber fitted with 81, 162 or 343 non-absorptive mineral-water bottles, all considered as 1:8 scale models. These data showed that a non-linear model must be used to express the relationship between fitting density and frequency. After considering several models and applying regression techniques to the experimental data, the best-fit variation of Q with frequency was found to be (with f_0=c/D_f):

$$Q(f) = Q(\infty) / (1 + 1.2\,f_0/f) \qquad (5)$$

Fitting Absorption Coefficient

A formula for fitting absorption coefficient was derived:

$$\alpha_f = [\alpha_{fr}\,(S_r + S_f) - \alpha_{er}\,S_r / \exp(-Q\,D_r)] / [S_f / \exp(-Q\,D_r)] \qquad (6)$$

in which α_{er} and α_{fr} are the average effective absorption coefficients of the empty and fitted rooms, respectively, calculated from measured reverberation times using diffuse-field theory, and D_r is the room mean free path $4V_r/S_r$. S_r and S_f are the room and fitting surface areas, respectively.

Experimental Validation

To validate the new expressions, experiments were done in a fitted 1:8 scale-model room with an acoustically treated ceiling. The fittings consisted of 31 mineral-water bottles placed on the floor giving Q_0=0.10 mFS^{-1} and Q'=0.18 mFS^{-1} (FS=full scale). The sound-propagation curves predicted by ray tracing [3] using Q(f) were in excellent agreement with the measured curves at all distances; Q_0 overestimated levels at larger distances.

Comparisons were also made with data from work by Hodgson [4], which involved ray-tracing prediction of sound-propagation curves in a fitted machine shop. By comparing measured sound-propagation curves with those predicted by ray tracing, Hodgson found that - using a fitting absorption coefficient of 0.1 - the best-fit fitted-region density of 0.23 m^{-1} gave excellent agreement with experiment at all frequencies.

Let us now apply Eq. (3) to the above data. The mean fitting dimension was calculated to be 1.15 m corresponding to Q'=0.255 m^{-1}. This is similar to the value of 0.23 m^{-1} found by the best-fit method. The 125-2000 Hz frequency-varying fitting densities calculated from Eq. (5) were 0.10, 0.14, 0.18, 0.21 and 0.23 m^{-1}, respectively. By comparing measurement and prediction for different values of fitting absorption coefficient, the octave-band best-fit values were found to be 0.20, 0.15, 0.12, 0.10 and 0.10, respectively. The agreement with experiment was as good using Q(f) and the best-fit, frequency-varying fitting absorption coefficients as that obtained by Hodgson using constant fitting absorption coefficient 0.1 and constant fitting density 0.23 m^{-1}. Since fitting absorption coefficient cannot be measured directly, it is not possible to say which set of prediction parameters best represents reality - only that they give equally good agreement with experiment.

Next let us calculate the fitting absorption coefficient using Eq. (6) - modified for the case of two fitting zones. Hodgson [4] presented "empty-room" and effective "fitted-room" surface-absorption coefficients for the machine shop. The octave-band fitting absorption coefficients for the fitted zone calculated using these values in Eq. (6) are 0.21, 0.14, 0.09, 0.11, and 0.10, respectively. These values are very similar to the best-fit values.

If we use the constant fitting density of 0.23 m^{-1} found by Hodgson [4], the octave-band fitting absorption coefficients become 0.061, 0.059, 0.056, 0.10 and 0.10, respectively. The values at 2000 and 4000 Hz are exactly equal to the results obtained by the best-fit method, but the first three values are significantly smaller. This suggests that fitting densities must vary with frequency.

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EMPIRICAL MODELS FOR PREDICTING NOISE LEVELS AND REVERBERATION TIMES IN INDUSTRIAL WORKROOMS

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INTRODUCTION

There is currently a large gap between performance and complexity in most industrial noise-level prediction models. Hodgson has proposed an empirical model which bridges this gap [1]. However, it does not allow for variable room dimensions, fitting densities and absorptions coefficients. In this paper a new prediction model is proposed which includes adjustable parameters for these factors. In addition, an empirical model for calculating reverberation times in industrial workrooms is presented.

DATA

The models were developed from measurements of sound propagation curves and reverberation times in thirty industrial workrooms using multi-variable regression analysis. An industrial workroom sound-propagation curve (SPC) is the variation with distance from an omnidirectional point source of the sound pressure level minus the source sound power level. Regressions to these data yielded the slope and intercept of the SPC - approximated as a single straight-line segment when plotted on a log-distance scale. Reverberation times used in the empirical model development were octave-band spatially-averaged values. Fitted workspaces displayed additional apparent absorption due to the fittings. This absorption was added to the absorption already present on the hard floor and divided by the floor area to give the effective fitted-floor absorption coefficient, α_eff.

REVERBERATION-TIME PREDICTION

Reverberation times were predicted using the Eyring equation. The effect of fittings was modelled as additional room surface absorption (α_eff). The relationship of α_eff to fitting density Q was calculated by linear regression of the form

$$\alpha\_eff = Slope \bullet Q + Intercept \qquad (1)$$

(results in Table 1) in which Q is defined as

$$Q = \frac{\Sigma SA}{4 \bullet V} \qquad (2)$$

as developed by Kuttruff [2], where ΣSA is the total exposed surface area of the fittings and V is the room volume.

NOISE-LEVEL PREDICTION

Using Tables 2, 3a and 3b the sound pressure level Lp(r) generated by a source is found by

$$Lp(r) = Lw + SPC.Intercept + SPC.Slope \bullet \log 10(r) \qquad (3)$$

where Lw is the sound power level of the source and r is the source/receiver distance. The standard deviations of the SPC prediction error obtained using Eq. (3) are shown in Table 5.

Table 1: Regression of α_eff vs. Q.

	Slope	Intercept	R^2	Standard Error
125 Hz	4.52	0.110	0.60	0.079
250 Hz	5.80	0.017	0.70	0.087
500 Hz	4.32	0.099	0.91	0.035
1 kHz	2.79	0.131	0.78	0.036
2 kHz	2.28	0.140	0.85	0.024
4 kHz	1.94	0.135	0.83	0.022

Table 2: Parameter coefficients of the SPC slope model.

Coef.	S1	S2	S3	S4	S5	S6	S7
125 Hz	-16.09	-12.11	195.9	-0.037	-5.08	14.95	-91.94
250 Hz	-21.94	-14.32	225.4	-0.028	-3.63	18.77	-102.3
500 Hz	-29.92	-12.53	194.1	-0.007	-2.33	17.74	-87.69
1 kHz	-26.89	-12.54	186.8	0.032	-9.79	17.96	-81.11
2 kHz	-24.88	-8.46	127.8	0.131	-11.60	15.94	-60.52
4 kHz	-19.21	-9.16	146.0	0.135	-11.88	13.18	-70.81

Table 3a: Parameter coefficients of the SPC intercept model.

Coef.	I1	I2	I3	I4	I5
125 Hz	-6.32	5.84	-86.7	0	5.03
250 Hz	-2.96	6.58	-98.0	0	5.00
500 Hz	19.4	6.46	-99.8	-121	5.13
1 kHz	-16.5	8.61	-127	48.3	12.4
2 kHz	-18.2	5.59	-85.5	72.2	0
4 kHz	-18.3	9.74	-155	37.1	0

Table 3b: Parameter coefficients of the SPC intercept model.

Coef.	I6	I7	I8	I9	I10
125 Hz	0	-8.33e-5	0	3.10e-3	21.4
250 Hz	0	-6.25e-5	0	2.50e-3	25.5
500 Hz	0	5.64e-5	0	1.14e-3	27.9
1 kHz	-9.04	-1.34e-4	0	1.82e-3	41.1
2 kHz	-10.1	0	-4.87e-4	0	29.0
4 kHz	-21.4	0	-8.40e-4	2.47e-3	65.9

Table 4: Description of SPC model parameters.

Parameter	Description
S1	Factor including air, surface and fitting absorptions
S2, I2	Room height
S3, I3	Log10(room height)
S4	1/Q
S5, I5	Average fitting height/room height
S6, I6	Room surface area / room volume
S7, I10	Constant
I1	Effective absorption of fitted floor (α_eff)
I4	Q
I7	Room volume
I8	Room surface area
I9	$\alpha\_eff \bullet$ floor area

Table 5: Performance of new SPC model.

Room Type	All	Type 1	Type 2	Type 3	Type 4
Fittings	-	No	No	Yes	Yes
Acoustical Treatments	-	No	Yes	No	Yes
Slope	1.05	0.99	0.85	1.09	1.33
Intercept	0.93	1.05	0.99	0.78	1.04

[1] "Preliminary simplified models for predicting sound propagation curves in factories," M.R. Hodgson, Canadian Acoustics 20 (3), 37-38 (1992).

[2] Kuttruff, Room Acoustics (Elsevier Applied Science, New York, 1991).

SURVEILLANCE AUTOMATIQUE DE L'IMPACT DU BRUIT INDUSTRIEL MÉTHODES D'ANALYSE

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ABSTRACT: The automated community noise control systems developed during the last twenty years are mustly dedicated to airport noise monitoring. In these conditions the existing systems measure primarily the peak noise parameters, such as L1%, L10%, Leq, SEL, with sometime and complementary a fine analysis of the noise events under a fixed treshold level. However, this conventional approach is not possible anymore for the impact control of industrial noise, in particular with long propagation distance to residential areas. Where the data are strongly affected by local noise sources (local traffic, lawn movers, etc.). For a continuous industrial noise source, it is easier to consider the background noise parameters, as L95%, L99% or Lmin.; statistical analysis duration is not critical (i.e. use of 30mn or 1h periods, for example) and computer modelisations are possible. This situation corresponds for example to those of the energy production infrastructures, like power generating or transformation stations. It is however possible to have some industries that generate fluctuating noise levels or constant noise levels but for short periods of time. In this case and with a long propagation distance, the control of the equivalent level Leq during fixed time periods (1 hour for example, or 15h day time and 9h night time) is not significant (although those parameters was used by noise legislations). For a good control of the industrial activities, it is necessary to use a shorter statistical analysis duration period, as 5 or 10mn, for example. Afterward, the possible correlations between the different noise monitoring stations must be analysed, in the same time at the source locations and in the concerned residential areas. The use of multiple community noise monitoring stations provides a good solution to eliminate the incertainty pertaining to the local effect of noise sources. Then, correlation analysis must be performed on parameters like L95% or L99%, in order to establish the human perception and the possible impact of the industrial noise.

INTRODUCTION

Les systèmes de surveillance automatique du bruit communautaire développés au cours des vingt dernières années concernent surtout le voisinage des aéroports, pour lesquels les paramètres mesurés s'intéressent essentiellement aux pointes de bruit (L1%, L10%, Leq, SEL, avec parfois une analyse fine des évènements sonores au-dessus d'un seuil de déclanchement). Or pour la surveillance de l'impact des bruits industriels à une certaine distance des sources de bruit, cette approche conventionnelle n'est plus possible, les relevés pouvant être fortement influencés par les bruits générés localement (circulation automobile, tondeuses à gazon, etc.). Lorsque les sources industrielles concernées sont entretenues pour de longue période de temps, il devient plus aisé de considérer des paramètres représentatifs du niveau de bruit de fond, tels que L95%, L99% ou Lmin., la durée de la période d'analyse devient moins significative (des analyses aux 30mn ou à l'heure peuvent très bien convenir) et ces sources peuvent être facilement modélisées.

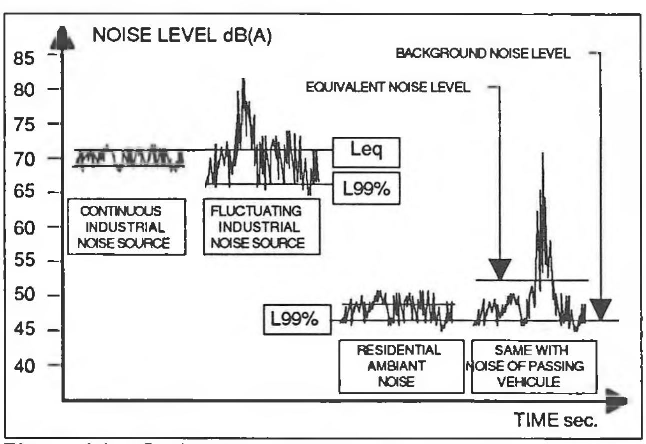

Figure nº 1: Bruits industriels et bruit résultant dans un secteur résidentiel éloigné.

Cette situation correspond notamment à celle des infrastructures de production et de transport énergétique, centrales, postes de transformation, etc. Reste le cas des industries susceptibles de produire des niveaux de bruit fluctuants ou entretenus pendant de courte période de temps. La surveillance du niveau continu équivalent Leq sur des périodes fixes, de 1h par exemple (ou bien même 15h de jour et 9h de nuit) ne signifie plus grand chose, surtout lorsqu'on s'éloigne de la source industrielle (bien que la plupart des textes législatifs se basent sur ce paramètre).

MÉTHODE DES CORRÉLATIONS

Si l'on veut pouvoir intervenir, il faut réduire la période d'acquisition et d'analyse statistique, à 5 ou 10mn par exemple, et ensuite analyser les éventuelles corrélations entre différentes stations de surveillance du bruit, tant à la source que dans le milieu communautaire concerné. La multiplication des stations en milieu résidentiel peut permettre de lever l'incertitude, quant aux sources de bruit locales. Les analyses de corrélation doivent ensuite porter sur des paramètres tels que L95 ou L99%, afin d'établir la perception et l'impact éventuel du bruit industriel. L'expérience montre que cette corrélation n'est possible qu'avec les paramètres de bruit de fond: à la source, ces paramètres sont représentatifs d'une augmentation des activités industrielles bruyantes et, au point d'analyse résidentiel, ils sont suffisamment indépendants des bruits locaux, notamment du trafic automobile. La corrélation apparente entre les stations de mesure indique au moins la possibilité que le bruit industriel devienne audible dans les secteurs résidentiels concernés.

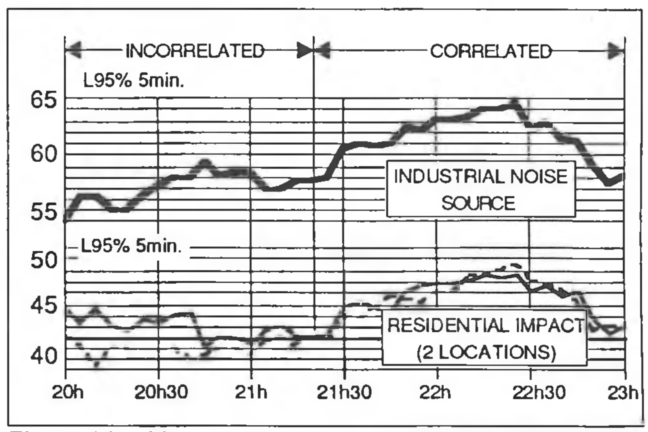

Figure nº 2: Mise en évidence de l'audibilité du bruit industriel par l'analyse des corrélations.

MÉTHODE DES MOYENNES

Lorsqu'on dispose d'une mémorisation suffisante des paramètres mesurés, il est également intéressant de procéder à partir des valeurs moyennes de la semaine précédente ou du mois précédent, pour la même tranche horaire de la journée. Cette méthode permet de calculer tout dépassement par rapport à la moyenne et ainsi d'identifier toute situation d'impact potentiel (confirmée éventuellement par la corrélation entre les stations résidentielles).

RÉFÉRENCE

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EVALUATION OF THE ZWICKER METHOD AS A SUITABLE ENVIRONMENTAL NOISE MEASUREMENT TECHNIQUE

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Background: Environmental noise is a common by-product of industrial energy facilities. Many of these energy facilities have components which radiate a considerable amount of acoustical energy. The acoustical energy may consist of high or low frequencies often containing prominent tonal components. Some of the characteristic sources responsible include compressors, diesel engines, fans, pumps, etc.[1]

The Energy and Utilities Board (EUB) is the primary regulator of the energy industry in Alberta. The EUB maintains a receptor based Noise Control Directive which utilizes an A-weighting Energy Equivalent Sound Level ($L_A$eq) to regulate environmental noise. The $L_A$eq with the unit dB(A), is a metric that emphasizes the middle frequency components similar to the frequency response of the human ear. It simply compresses sound from a broadband of frequencies into a relative response of the middle frequency band (1000-2000 Hz) by eliminating low frequency sounds. This measurement scale is well suited for measuring broadband noise. The Noise Control Directive is receptor based, meaning that it measures the noise of a facility at some distance away. Higher frequency sounds emitted from facilities tend to diminish over distance leaving predominantly low frequency sounds. As a result, it could be said that from the receptor stand point, the energy facilities emit low frequency sounds. Although $L_A$eq is readily accepted as a standard community noise metric it may not fully quantify the true impact of energy industry noise at a distant residence.

Project Scope: Based on this and the EUB experience in dealing with noise complaints, a research program was initiated in 1995 to determine if there was a better method to measure and quantify the dominant low frequency sounds of the energy industry. A method was required that would recognize low frequency sounds and capture the impact of any tonal components. Affected residents often described the tonal component sounds as being the most annoying.

Earlier work by the EUB began research into the Zwicker Method for calculating loudness as described in the ISO 532 Method B as a potential supplemental metric system. The Zwicker Method met the necessary criteria and testing has been ongoing to understand how it might be applied within the framework of the EUB Noise Control Directive.[2] The idea was to develop a supplement to the $L_A$eq measurement if possible.

Research Phase: The techniques employed were carefully chosen and controlled to obtain valid and consistent results. This included the effects of meterological factors, no abnormal activities in the vicinity of research testing, suitability of monitoring location to a noise source, topographical features, etc.[3] With this in mind, the type of sound measurements conducted were series of instantaneous linear, slow response 1/3 octave band measurements. This measurement method produced a frequency spectrum of the sounds. The results were converted to loudness using Zwicker's method and placed into a number of statistical models for analysis and assessment.

Primarily, measurements were conducted at residences near energy industry facilities, however, to obtain a better perspective of loudness results in general, various other ambient measurements were taken. These included natural background sounds, highway traffic, rivers, and other non-energy related industries. Along with the noise surveys, a resident survey was conducted to help determine the quality of the noise and how it affected them. The survey consisted of ten questions designed to understand the way the noise was perceived by the residents and to gauge their level of satisfaction with the industrial operator in case there was some other root concern besides noise.

Evaluation of Results: Analysis of the data concluded that loudness quantifies low frequency noise and tonal components more effectively than the current $L_A$eq method. If used, loudness results should be converted into $L_A$eq using an appropriate technique. However, loudness cannot distinguish between pleasant or annoying sounds. Using loudness measurements for all complaints would hardly be warranted and would effectively result in setting an artificially lower permissible sound level at residences. This is clearly unacceptable as it would unfairly penalize some industry operators. The resident survey questionnaire should be used to determine noise perception and quality as the first step in the assessment for conducting loudness calculations and applying any resulting adjustment factor to compliance survey results.

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EUB INTERNAL NOISE MONITORING POLICY

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Introduction: It is not the role of the Alberta Energy and Utilities Board (EUB) to replace industry's in-house experts or independent acoustical consultants in performing the technical analysis of environmental noise complaints, but rather to expedite problem issues, present a fair and unbiased viewpoint in mediating disputes, and in the administration of the Noise Control Directive ID 94-4 (NCD).[1] The purpose of this policy is to provide guidance in determining when a comprehensive or ambient sound survey can be conducted by the Alberta Energy & Utilities Board (EUB).

Discussion: The EUB operates the Sound Monitoring System (SMS) which consists of a B&K 2231 sound level meter capable of a wide range of acoustical measurements, a B&K 1625 octave band filter set to conduct frequency analysis, and a Mitsubishi Hi-Fidelity VHS player capable of recording actual background noise over a ten-hour period. Accompanying the SMS is software that allows for the calculations and reporting of survey and frequency analysis results. When used in accordance with the criteria below, this equipment can determine if a particular facility is in complince with the NCD. Each area office is also equipped with a much simpler sound level meter which can provide an instataneous Sound Pressure reading in decibels (dBA). This measurement device is used only as an assessment tool to get an idea of the severity of the problem; not for establishing compliance for a facility or as the sole basis for confirming that a complaint is justified warranting remedial action by the operator.

Roles within the EUB: The Noise Monitoring Team, consisting of an acoustical specialist and five Field Surveillance staff, are trained in the use and application of the Sound Measurement System (SMS). Team members are responsible for operating the equipment when noise surveys are warranted in accordance with the strict criteria within the EUB Noise Monitoring Policy[2]. A Team member will conduct the survey and be available to assist in the interpretation and explaination of the NCD. They will also download survey data, analyse the data and generate a report of the survey results.

The acoustical specialist is responsible for the adminstration and interpretation of the NCD and the EUB Noise Monitoring Policy, and serves as an expert technical resource on acoustical engineering matters to the EUB. He is also required to organize and provide opprtunities for training and technical proficiency programs for EUB staff, and advance and promote industry and public understanding of environmental noise control requirements and methodologies.

EUB Noise Monitoring Survey Criteria: The following criteria must be used to assess if it is appropriate and applicable to use the SMS to deal with a complaint or noise control situation. Any exceptions to these criteria must be discussed with the acoustical specialist prior to commiting to a monitoring survey: 1) When the results of an existing survey conducted by the operator or a reputable consultant are in question by the complainant. 2) When there are unreasonable delays (> six weeksfrom the date of EUB request) for a survey to be conducted by the operator or their consultant. 3) For source identification when a multi-source environment exists and field staff are unable to identify the main contributor. 4) When the EUB determines a need based on site-specific circumstances. The acoustical specialist must be consulted in such situations to confirm the need for a survey, and the availability of the SMS. A prime directive within the EUB Noise Monitoring Policy is to not assume the role or responsibility of industry to investigate and respond accordingly to environemtnal noise complaints related to any of their operations. Rather it is an additional tool which serves all parties objectively.

Complaint Resolution: The EUB strives to provide effective and consistent response to environmental noise complaints directed at the energy industry. Various steps are pre-defined and followed in the course of complaint resolution with all significant information and events documneted and communicated to the affected parties in a timely fashion. The EUB may also make recommendations regarding any survey results; communicate with residents or communities regarding the EUB environmental noise control policy; assist in dispute resolutions; and provide enforcement options, where required. The EUB goal in complaint resolution is to bring the matter to a close or to some satisfactory endpoint within 90 days by setting clear expectations for all parties accompanied by well understood and agreed to consequences for failing to meet these.

Compliance and Enforcement: When compliance to the NCD has been achieved by the operator, it is the responsibility of the EUB to see that this is appropriately communicated and understood by all parties. When there is non-compliance an appropriate course of enforcement must be chosen having regard for the severity of the problem, the willingness of the complainants to accept the terms, and consistency with internal policy. It is important to communicate this to all parties and to set out reasonable expectations to resolve the matter. Some enforcement options may range from total shut down of the offending facility, to allowing continued operation while mitigation and sound attenuation measures are being engineered and installed. The operator or the complainant may appeal a staff ruling detailing their reasons to the acoutical specialist. The EUB may then consider one or more of the following options: 1) further noise assessments, 2) initiating a dispute resolution process, 3) requesting noise attenuation at the facility, 4) appropriate enforcement procedure, 5)to refer the matter to the Board for a Public Hearing into the matter.

References:

1. "Noise Control Directive ID 94-4" Alberta Energy and Utilities Board (August 94)
2 "EUB Internal Noise Monitoring Policy" Alberta Energy and Utilities Board (July 96)

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Multidimensional scaling of unfamiliar, complex sounds: Age and Context Effects

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In a series of experiments, multidimensional scaling was used to evaluate the encoding of complex sounds by school-aged children (6-11 years) and adults. Listeners rated the similarity of all possible pairs of sounds within a given set. The ratings were used to derive a multidimensional space in which the stimuli were represented as points and the axes represent the perceptual dimensions used in making the judgements. Using a computer procedure listeners heard pairs of sounds and were asked to place two computer images at a distance from one another that reflected their perceived similarity.

Experiment I. Age effects. The stimuli were 17, 430 ms signals including 3 pure tones (250, 1000, and 4000 Hz), 6 harmonic complexes consisting of the low (2-6), high (12-16) or wide (2-16) harmonics of either a 110 or 200 Hz fundamental, 2 AM noises (500 Hz and 2000 Hz noises, 12 dB modulation at 6 Hz), 2 FM tones (500 and 2000 Hz, frequency modulated at 10 Hz), 3 narrow band noises (centred at 500, 1000, and 2000 Hz) and 1 wide band noise (500-2000 Hz). Listeners were 10 children aged 6- to 7-years (mean: 7 yrs, 2 mos) 11 aged 10- to 11-years (mean: 10 yrs,7 mos) and 11 adults (mean: 26 yrs) with normal hearing.

A 3-dimensional space was derived for each age group using INDSCAL (Carrol & Chang, 1970). With increasing age the proportion of the variance accounted for increased. All listeners evaluated the sounds according to both spectral and temporal features but the relative weights assigned to each dimension, the integration of features, and the resolution along the dimensions varied with age. The adults grouped the stimuli into 3 groups representing the tones, noises, and harmonic complexes. Group formation was based on similarities in the first 2 dimensions reflecting temporal structure (number of components per critical band) and spectral shape (number of spectral peaks). Within each group the stimuli were ordered according to frequency which represented the third dimension. The 10-yr-olds formed similar stimulus groups but the groups were more loosely defined and the dimensions did not represent independent acoustic features of the sounds but a combination of both spectral and temporal cues. The 7-year-olds also showed an integration of spectral and temporal features in the individual dimensions and further showed less resolution in the temporal dimension, discriminating stimuli only on the basis of temporal fine structure but not envelope variations.

Experiment II. Stimulus range effects. Increasing the range of a features should increase its salience (e.g. Ashkenasy & Odom, 1982, JECP 34). Given that the adults weighted frequency as the 3rd most salient dimension in Exp. 1, its range was increased to test this hypothesis. Eight adult listeners (7 from Experiment 1) participated. Stimuli were 4 tones (250, 1000, 4000, and 6000 Hz), 6 harmonic complexes (F0=110 or 400 Hz and harmonics 2-6, 12-16 or 2-16), 3 noise bands (centre frequencies of 250, 1500, and 4000 Hz), 2 wide band noises (500-2000 Hz, and 200-4000 Hz), and 2 inharmonic complexes (5 randomly chosen from the 220-660 Hz and 4800-6400 Hz range). The 3-d solution no longer showed a clustering of stimuli as in Experiment 1. Increasing the frequency range increased its salience to dimension 1.

Experiment III. Stimulus distribution effects. It has also been suggested that salience will be determined by the diagnostic value of the feature reflecting the usefulness of the dimension for forming categories (e.g. Tversky, 1977, Psych. Rev. 84). To examine this hypothesis, four groups of adults were each asked to evaluate the similarity of a different stimulus set. Stimuli were complexes that varied either continuously or categorically in frequency and number of components. A low and a high frequency range were used that were adjacent to one another (continuous distribution) or separated by 3 critical bands (categorical distribution). The number of components was varied either continuously (1-9) or categorically (1-2 or 5-9). Results showed that the number of components overall was not a significant factor, but that the number of components per critical band (periodicity) was. When both frequency and periodicity varied continuously, the listeners' solution reflected a trend for categorization of stimuli based upon these two parameters. The clustering was enhanced when the parameters varied categorically. When only one parameter varied categorically, it received the highest salience and was used to classify the stimuli with stimuli evenly distributed along the dimension that was continuously sampled.

DEVELOPMENT OF MEMORY FOR SEQUENCES OF ANIMAL SOUNDS: RELATION TO DIGIT-SPAN AND LANGUAGE ABILITY

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Background

The ability to remember a sequence of sounds is crucial to many auditory tasks, such as speech production and recognition. Previously Cohen and O'Connor (1994), by means of a new computerized test, examined the development of memory for sequences of meaningful nonverbal sounds across four different age groups. Five readily identifiable animal sounds provided the test material. In the first block of test trials, a sequence of two animal sounds was presented. In successive blocks, additional sounds were presented to a maximum of five animal sounds. Listeners responded by pointing to the pictures of the animals in the order in which the sounds had occurred. Performance increased systematically with age and sound duration (250 vs 500 ms) and decreased with increasing number of sounds in the sequence.

Present Study

The present study aimed to replicate and extend these findings. The same test was administered to preschool children, primary school children, adolescents and young adults, with 10 children in each group. The pattern of Cohen and O'Connor's results was replicated with respect to age, sequence length, and duration. In addition, data on digit-span memory was obtained for each subject in this study. Digit span correlated significantly with performance, but left over 60% of the variance unexplained. Thus, the present task measures processes beyond those required for a test of auditory memory of numerical sequences, a test often used to assess auditory sequential memory.

Follow-up Study and Conclusions

In a follow-up study, the potential diagnostic potential of the present task was investigated. Nine children, 8 to 9 years of age, classified by their teachers as having language impairment (primarily reading) were compared on the animal sounds task, with 9 children having normal language abilities. Language impairment was associated with poorer performance in one condition of the test, the most difficult (5-sound) condition. This nevertheless suggests the sensitivity of the animal sound sequencing task to language impairment. It also supports the notion of a link between the ability to remember the order of nonverbal sounds and normal language development. This relation is also implied by Lincoln, Dickstein, Courchesne, Elmasian, R. & Tallal, (1992) who compared children of normal language development and with children having developmental language disorder on a modified Repetition Task (cf. Tallal, Stark, Kallman, & Mellits, 1981) entailing memory for the presentation of patterns of two tones of different frequency.

Whereas the animal sounds task and the Repetition Task may test similar abilities, the animal sounds task has an advantage to children of being intrinsically more interesting. The use of environmental sounds in audiometry for children has also been suggested by Myers, Letowski, Abouchacra, Haas and Kalb (1994) .

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Note

The paper is based on an honours thesis of Patti C. Graham, The development of auditory sequencing ability with a focus on specific language impairment, Department of Psychology, Univ. of Prince Edward Island, Dec. 1995.

Processing of Dynamic Signals Independent of Presbycusic Hearing Loss.

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Processing of dynamic signals as a function of: age, frequency region, background noise; transition direction, and endpoint frequency was examined among individuals free of presbycusic hearing loss.

Participants

Individuals ranging from 20 to 75 years divided into 5 decades : (20-34 years, n = 25; 35-44 years, n = 20; 45-54 years, n = 20; 55-64 years, n= 15; 65-75 years, n = 18) with normal bilateral air-conducted thresholds (no > than 25 dB HL from .5 kHz to 8 kHz); type A tympanograms, and no history of otological disorders participated in this study .

Stimuli and Procedure

Signals (50 ms duration) were synthesized at 2 frequency regions: 1) at a center frequency of 1030 Hz of the maximal transition excursion ; 2) at a center frequency of 2685 Hz of the maximal transition excursion. Two different patterns of: transition trajectory: (upward or downward); and, end frequency conditions: (varying onset frequencies common offset frequency; common onset frequency varying offset frequencies) created 4 series at each frequency region: converging up (CU); converging down (CD); diverging up (DU) and diverging down (DD). Thresholds were determined in 1) quiet and 2) continuous speech spectrum noise for : converging upward signals (CU-N); and, diverging downward signals (DD-N) in a 2AFC paradigm with individually randomized trials.

Results and Conclusions

There were discontinuities in the effect of age; i.e., advancing age did not produce a concomitant worsening of performance. For converging signals the eldest group performed significantly worse than the youngest listeners but the pattern of responses for the 45-54 year olds was not as predictable. For CD signals 45-54 year olds showed lower thresholds than the 55- 75 year olds, but for CU signals the 45-54 year olds were not significantly better than the 65-75 year olds but the 55-64 year olds were better than both of these groups. For diverging signals 45-54 year olds performed better than the 65-75 year olds for DU signals but ***not*** for DD signals. Transition direction was not a significant factor

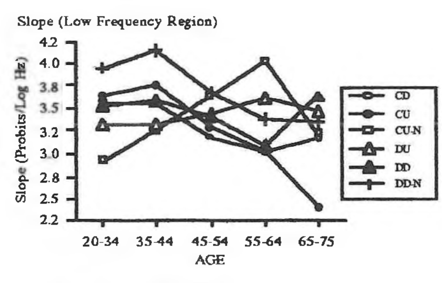
At the higher frequency region again the performance of the mid-age range group of listeners was the most enlightening. 65-75 year olds performed significantly worse than all other age groups for signals with a common offset frequency, but for signals which diverged to varying offset frequencies not only the 65-75 year olds but also the 45-54 year olds demonstrated higher thresholds than all other age groups i.e., the 55-64 year olds performed better than the 45-54 year olds for these series. Though downward transitions were easier to discriminate than were upward transitions this effect was significant only for the 45-75 year olds. Moreover, diverging signals were significantly easier than converging signals to discriminate only for 55-75 year olds.

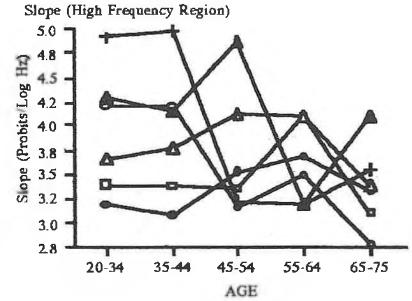
In the low frequency region, among 45-54 year olds, the presence of noise negatively impacted CU signals but *decreased* the average threshold for DD signals. Eldest listeners did not show adverse effects for noise for diverging signals but did demonstrate increased thresholds for converging signals. At the higher frequency region, the effect of noise was most noted for the impact on endpoint frequency. Converging signals were more difficult to discriminate in noise than were diverging signals, relative to the same signals presented in quiet, principally for listeners 55 years and older.

Certainty of response as assessed by the slopes of the psychometric functions was not a linear function of age (Fig. 1) with neither a requisite shift to the right nor a flattening of the slopes with age. At the low frequency region, in quiet there was no effect on the slopes for direction of the transition. There was an effect for endpoint frequency with 65-75 year olds showing shallower slopes for converging signals while 20-34 year olds showed steeper slopes. In noise, there was no difference in the slopes for diverging signals however for converging signals, 55-75 year olds showed steeper slopes; 20-34 year olds shallower slopes. At the higher frequency region, in quiet all age groups except for the 55-64 year olds showed shallower slopes for converging signals than for diverging ones. Only listeners aged 20-44 years showed steeper slopes for downward transitions versus upward transitions. In noise, the only effect was for diverging signals where 45-54 year olds and 65-75 year olds showed shallower slopes; 20-44 year olds showed steeper slopes, while 55-64 year olds showed no difference for quiet versus noise conditions. Slopes obtained in quiet for diverging signals had greater ***intragroup*** variability relative to those obtained in the presence of background noise for both high and low frequency regions. In contrast, slopes obtained in noise for converging signals showed ***less*** variability within groups than slopes obtained in quiet at both frequency regions.

Evidence indicates that: 1) age effects are not a monotonic function of signal difficulty; 2) deterioration in processing occurs early in mid-life with a potential recovery due to 'compensatory mechanisms'; and, 3) noise may enhance the processing of certain signals. These effects comprise an interactive consequence on perception involving combinative aspects of: stationary (external) and nonstationary (internal) noise; temporal-spectral resolution; and compensatory processes.

Figure 1. Age effects of Psychometric Slopes





FREQUENCY RESOLUTION IN NOISE EXPOSED MUSICIANS

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Studies suggest that the measurement of frequency resolution may be used as a sensitive early indicator of noise-related auditory damage (West and Evans, 1990; Bergman et al.,1992). Unfortunately, the current research literature falls short in providing data on the musician population.

This study was designed to investigate the possibility of using a measure of frequency resolution as an early indicator of auditory damage in musicians resulting from exposure to loud music.

Researchers investigating frequency resolution in musicians have centered their attention on the possibility that the auditory filters in musicians are narrower than non-musicians as proposed by Soderquist (1970), perhaps due to musical training and experience. This proposed "narrower" filter could have an effect on the reliability of using a measurement of frequency resolution as an early detector of noise related auditory damage in musicians, especially in terms of producing normative frequency resolution data.

Five groups of ten adult subjects twenty to forty years of age were studied. The subject groups included; a control group of non-musician subjects with normal hearing (limited exposure), untrained musicians (low exposure), untrained musicians (exposed), trained musicians (low exposure), and untrained musicians (exposed). Subjects were placed into the appropriate groups on the basis of responses to a questionnaire. The questionnaire was designed to document the individual's otologic history, noise exposure history, musical training, and performance experience. Individuals with significant otological histories or exposure to other potentially hazardous noise sources were excluded from the study. Pure tone audiometry was used to exclude all subjects with a high frequency average (1, 2 & 4 kHz) greater than 20 dB HL.

Frequency resolution was measured using the notched noise paradigm suggested by Patterson (1976). A computer program developed by Glasberg and Moore (1990) was used to estimate the bandwidth, filter slope and detection efficiency parameters of the underlying auditory filters.

Results from this study revealed a significant widening of the bandwidths of the auditory filters of the exposed musicians consistent with a deterioration of frequency resolving abilities. The equivalent rectangular bandwidth of the filters were 13% wider for exposed musicians than for low exposure musicians. The slope of the lower filter skirt was significantly shallower for exposed musicians as compared with low exposure musicians. No significant effects of musical training or exposure-training interactions were found.

These results are encouraging and suggest that there is an effect of noise exposure on the bandwidth and lower slope of the auditory filters of musicians. Research into the clinical applications of such measures is underway.

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DEFICITS IN TEMPORAL PROCESSING ABILITY WITH APHASIA

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1.0 INTRODUCTION

Aphasia is a communication disorder, typically due to a stroke affecting the left hemisphere of the cortex. In a schema proposed by McNeil and Kimelman,[1] it was suggested that observed language deficits in aphasia are secondary to primary deficits in the processing of the intensity, duration and frequency of sound. Research studies support this view. Individuals with temporal lobe pathology and speech perception deficits also have reduced acuity for a change in frequency and duration.[2,3] Comprehension deficits specific to aphasia are related to difficulty with the processing of temporal order.[4] The present study was conducted to determine the relevance of both site of pathology and the severity of language processing difficulties to the processing of nonlinguistic auditory temporal stimuli in aphasic individuals.

2.0 METHODS AND MATERIALS

2.1 Subjects

One group of normal subjects (N=20) and two groups with confirmed pathology of the left posterior (N=16) and left anterior (N=6) hemisphere from stroke, participated. Months post onset of symptoms ranged from 2-37 months. Subjects in all three groups had screened normal hearing and were right-handed, native English speakers, and 30-70 yrs of age.

2.2 Procedure

Standardized tests of speech-language performance were administered to each subject in a quiet room. The tests included the Porch Index of Communicative Abilities (PICA),[5] the Boston Diagnostic Aphasia Examination (BDAE),[6] and the Revised Token Test (RTT).[7] Each test included subscales which required subjects to listen, speak, read, write, gesture and manipulate objects.

The psychoacoustic tests were carried out in a double-walled sound proof booth. The apparatus has been previously described.[8] Duration difference limens (DLs) were measured in each ear for standard durations of 50 ms (R/D=10 ms) and 300 ms (R/D=50 ms), using a two-interval forced-choice procedure.[9] The short standard represented the shortest duration encountered in speech (e.g. fricative) and the long standard was the average duration of a syllable. The stimulus was a one-third octave noise band centred at 2 kHz, presented at a comfortable listening level. Subjects responded by means of a response box placed on the side contralateral to the lesion.

3.0 RESULTS AND DISCUSSION

The mean DLs observed for the short standard were 14 ms, 35 ms and 45 ms for the normal, left posterior and left anterior groups respectively. The DLs for the long standard were 45 ms, 79 ms and 82 ms. There was no difference between ears within group. Outcomes for the normal group were comparable to published values.[8] The differences in the DLs for the normal and left posterior groups were statistically significantly ($p<0.001$). Individual results for the six subjects with left anterior lesions were within the range observed for the left posterior group.

The left posterior group also showed significantly lower scores on each of the three speech tests, relative to normal. Again, the range of values for the left anterior subjects overlapped those of the posterior group. For individuals with left posterior lesions, the DLs for the short standard measured in either ear and the DL for the the long standard measured in the right ear were negatively correlated with the results of the PICA ($p<0.01$).

The results confirm that individuals with left hemisphere lesions have reduced duration discrimination ability. Site of lesion, posterior versus anterior, does not appear to be a signicant factor. Severity of language processing deficit as measured by the PICA is a significant correlate of outcome.

Acknowledgments

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Localization in Real and Virtual Rooms

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Introduction

Elderly listeners experience more difficulty understanding spoken language in non-ideal listening conditions than do younger adults. Specifically, they have more trouble in many everyday listening situations where background noise or reverberation conditions are unfavourable. Despite their difficulty understanding what is said in non-ideal conditions, elderly listeners often perform like younger adults in ideal listening conditions, such as when they are talking to one familiar person in a small, quiet room. Furthermore, many do not have clinically significant elevations in pure-tone thresholds, and existing clinical tests conducted in the artificial conditions of soundbooths are not useful in predicting an individual's performance in real-world communication situations. Therefore, it is important to devise new methods to allow researchers and clinicians to better evaluate how listeners perform in non-ideal, real-world conditions.

Testing listeners in actual acoustic conditions would be the most ecologically valid approach; however, precise control of the test stimuli would be jeopardized. Auralization[1], or the simulation of acoustic environments, is another approach that permits more realistic conditions to be created while maintaining precise control of test stimuli. As a first step in adopting the latter approach, we tested and compared the abilities of listeners to localize speech signals in a real room and in simulations of the same room.

Method

The Real Room: The real room was a 12.92' wide x 17.42' long x 8.83' high room, with one door and no windows, located in a modern building with research offices and labs. Eight loudspeakers were arranged in a circle, 45° apart, such that each loudspeaker was 5' from the listener seated in the center of the room, and at the same height as the head of the listener. The reflective characteristics of the surfaces (walls, floor and ceiling) and the reverberation times per octave band for the room were measured [2].

The Virtual Rooms: Four simulations of the real room were constructed using Tucker-Davis Technologies (TDT) hardware and modified software [2]. The first simulation included only the direct wavefront. The second included the direct wavefront and the first reflections from each of the six surfaces, where the surfaces were modelled using the frequency-specific reflective characteristics measured in the real room. The third included the direct wavefront and a reverberant tail, which was modelled using the average of the frequency-specific reverberation times that were measured in the real room. The fourth included the direct wavefront, the first reflections, and the reverberant tail. The four simulations were convolved with each of three head-related transfer functions (HRTFs) provided with the TDT Soundstage software. Thus, a total of 12 conditions were constructed (4 room simulations x 3 HRTFs).

Stimuli: A set of 20 soundfiles were used. In each soundfile was a 4-second segment of 8-talker babble. The segments were shaped with a rise-fall time of 100 msec. The RMS of the soundfiles ranged from 2.013 to 2.014 Volts. In the real room, soundfiles were played out of the loudspeakers at a sound pressure level of 70 dB SPL. In the virtual rooms, the software was used to assign one receiver position corresponding to the position of the listener in the real room, and 8 source positions corresponding to the 8 positions of the loudspeakers in the real room.

Subjects: Twenty-four normal-hearing, young adult listeners were paid for their participation in the study.

Procedure: Each of the 20 babble files was played out once from each of the 8 loudspeaker positions. The order of presentation of files at each position was random. On each trial, the subject pushed one of 8 buttons labelled in compass directions (N, NE, E, SE, S, SW, W, NW) to indicate where the sound source seemed to be located. In Experiment 1, subjects received no feedback. In Experiment 2, subjects received feedback indicating that the response was correct or, if it was incorrect, what the correct response should have been. Each subject completed both experiments in the real room in a single one-hour session. Similarly, 12 sessions were completed later in the virtual rooms, with one session for each of the 12 conditions, and with the order of conditions varied across subjects.

Results

In both the real and virtual rooms, without feedback, subjects were poorest at localizing sound sources behind them (S), with the most common error being a front-back (N-S) confusion. However, with feedback, performance improved in both the real and virtual conditions.

Discussion

Auralization is feasible for use in the laboratory. It provides useful information about the real-world abilities of listeners to localize sounds. Its usefulness for measuring other aspects of perceptual performance in research and clinical studies remain to be determined.

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THE RELATION BETWEEN PSYCHOACOUSTIC COMBINATION TONE GENERATION AND TWO-TONE SUPPRESSION IN AN AUDITORY MODEL

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INTRODUCTION

This paper reports on the computational modelling of the 2f1-f2 psychoacoustic combination tone (CT) elicited by two closely-spaced primary tones f1 and f2. Previous research indicated that the amplitude of the perceived CT depends on the method of measurement [1,2]. Typically, a cancellation-tone or a loudness-matching experiment is used. The latter method gives consistently lower estimates of the CT amplitude. It was suggested that the difference between methods could be attributed to suppression effects by the primary tone f1 [1,2]. Computer simulations carried out with a model of the auditory periphery support this hypothesis [3,4]. These results are summarized in this paper.

COMBINATION TONES

In the cancellation-tone experiment, a probe tone at the frequency of the CT to be measured, i.e. 2f1-f2, is presented externally and *simultaneously* with the primary tones f1 and f2. The task of the subject is to adjust the amplitude and phase of the probe until the pitch sensation of the CT becomes inaudible. In the model, this experiment is simulated by adjusting the amplitude and phase of the probe until the peak in the basilar membrane excitation pattern at the corresponding CT frequency disappears completely [3].

In the loudness-matching experiment, the probe tone 2f1-f2 is presented externally but *non-simultaneously* with the primary tones f1 and f2, e.g. by alternating successively between probe tone and primary tones. The task of the subject is to adjust the amplitude of the probe until it matches the loudness of the CT during presentation of the two primaries. In the model, this is simulated by adjusting the amplitude of the probe until the peak of its excitation pattern equals that of the CT during presentation of the primaries [3].

The results from both psychoacoustic experiments [1,2] and model simulations [3,4] give consistently lower estimates of the CT when measured with a loudness-matching rather than a cancellation-tone method. The difference between methods increases with primary level for L1=L2. When measured as a function of L2 for L1 = constant, the difference between methods is essentially constant, so that L1 seems to be the controlling factor. The experimental and model data agree well as shown in Table I for typical stimulus conditions.

TWO-TONE SUPPRESSION

Under certain conditions, the loudness of a stimulus tone decreases when presented simultaneously with a second stronger tone. Thus, the probe in the cancellation-tone experiment could be subjected to suppression effects by the primaries, especially by f1.

Table I: Comparison of estimates of the 2f1-f2 CT level from two different methods for three subjects [1] and for the model. Stimulus: f1 = 1400 Hz, f2 = 1680 Hz, L1 = L2 = 50 dB SL (for subjects) or 50 dB SPL (for model).

Subject	Cancellation-tone	Loudness-matching	Diff (dB)
GS	24.2 dB SL	17.3 dB SL	6.9
FW	20.4 dB SL	15.2 dB SL	5.2
TH	28.4 dB SL	16.4 dB SL	12.0
Model	22.4 dB SPL	15.0 dB SPL	7.4

To test this hypothesis, the suppression of a stimulus tone 2f1-f2 by a tone f1 in absence of f2 was studied in the model for the stimulus conditions corresponding to the data presented in Table I. It was found that a stimulus tone 2f1-f2 of 22.4 dB SPL presented simultaneously with a tone f1 of 50 dB SPL would be suppressed to the level of a 15.0 dB SPL tone when presented alone. This and other simulations carried out with the model suggest that the probe is subjected to suppression effects in the cancellation-tone experiment. Moreover, the amount of suppression is exactly equal to the difference between the two main methods of measuring the CT level (Table I). Further simulations also revealed that, as a result of suppression by f1, the phase of the probe in the cancellation-tone experiment becomes exactly out-of-phase with the internally-generated CT. These conditions guarantee cancellation.

In summary, this modelling study supports the hypothesis that the difference between methods of measuring the CT amplitude is due to suppression effects, and that the generation of auditory distortion products and the effects of two-tone suppression have a common origin in the nonlinearity of the cochlea.

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EVALUATION AND IMPROVEMENT OF THE ACOUSTICAL PERFORMANCE OF VENTED EARPLUGS

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Introduction

Ear plugs are a common type of hearing protector. Those studied here were of the custom-moulded type, and were made of silicon rubber. They were either solid or were 'vented' to allow the equalization of air pressure and to reduce the attenuation of speech frequencies to allow speech to be understood. The vent consists of a small-diameter 'core' tube into which is inserted a 'filter' which further reduces the vent diameter in one or more steps. The noise-attenuation performance (insertion loss - the difference in the sound-pressure levels measured at the eardrum without and with the earplug) of earplugs is usually determined using 'subjective' REAT tests, involving human subjects, according to ANSI S3.19-1974. This paper discusses research conducted to:

- evaluate objective methods for measuring earplug performance using a dummy head;

- to model vented earplugs and improve their performance by redesigning the filter. More specifically, it was hoped to increase the insertion loss at 1000 Hz by about 2 dB.

Objective Test Method

'Objective' measurements of earplug insertion loss were made using a KUNOV dummy head with compliant ears in three environments and the results compared to existing REAT data. The environments were created by headphone presentation in an audiometric booth, and by loudspeaker presentation in a semi-diffuse sound field (conforming to ANSI S12.6-1984) and in an anechoic chamber for various angle of incidence. Fig. 1 shows some results for a vented plug. Loudspeaker presentation in the semi-diffuse field gave the best agreement with REAT results.

Vented Earplug Model

A theoretical model for predicting the insertion loss of vented earplugs inserted in an ear was developed based on theory proposed by Iberall [1] and Egolf [2, 3]. The model assumes that sound energy reached the eardrum due to transmission through the vent only. The vent, filter and earcanal were modelled as a series of rigid, cylindrical tubes terminated by an average human eardrum impedance.

Fig. 2 shows a comparison between the predicted 125-8000 Hz insertion losses of an earplug vent, and those of a vented plug as measured subjectively and objectively. At all but the lowest and highest frequencies the predicted vent insertion losses are 10 dB higher that those measured. This suggests strongly that the sound energy transmitted through the vent is significantly lower than that transmitted through the solid plug material. The obvious conclusion is that redesigning the filter in the vent will have negligible effect on earplug performance at these middle frequencies.

Earplug Redesign

The insertion losses of solid plugs with different densities of solid material were measured. Density did not substantially and consistently affect sound transmission through the solid material.

Despite coming to the conclusion that the performance of the vented earplugs studied here could not be improved by filter redesign, the prediction model was used to investigate how changes in filter dimensions would affect earplug insertion loss in the absence of significant sound transmission through the solid plug material. The prediction results demonstrated that changing the filter dimensions affects performance over a wide range of frequencies; dimensional changes do not allow performance to be tuned at individual frequencies.

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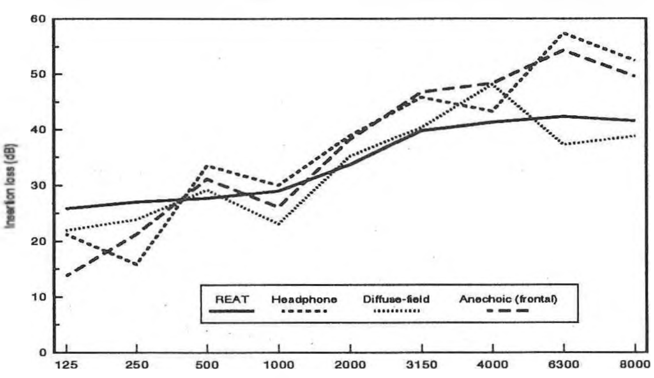

Fig. 1. Insertion loss of a vented earplug as measured 'subjectively' by REAT, and 'objectively' using a dummy head and either headphone, ANSI diffuse-field or anechoic, frontal-incidence signal presentation.

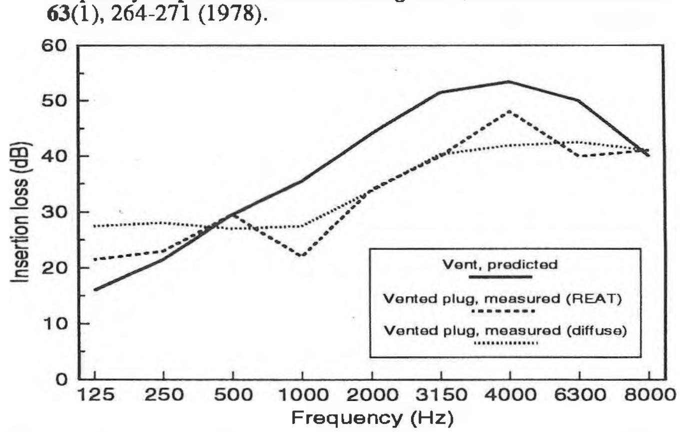

Fig. 2. Predicted insertion loss of an earplug vent, and that of a solid earplug as measured objectively.

AN EXAMINATION OF COCHLEAR FILTER RESPONSE PROPERTIES USING F_1- AND F_2-SWEEP DPOAE PHASE DELAY ESTIMATES IN HUMAN ADULTS

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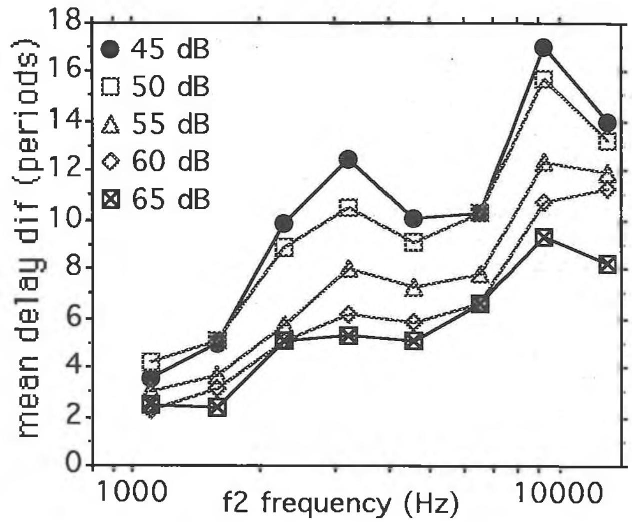
Distortion product otoacoustic emissions (DPOAEs) are recorded in the ear canal when two continuous pure tones (or primaries) of different frequency are presented simultaneously, and energy is emitted at a third frequency. Distortion is generated by the nonlinear interaction of the primaries, where the traveling-waves of the lower frequency (f_1) and higher primary (f_2) overlap along the basilar membrane. The distortion provides an indication of the degree to which the vibration of a particular region of the cochlea can be influenced by tones at other frequencies. Thus, DPOAEs reflect the mechanical tuning of the cochlea.

The slope of the unwrapped DPOAE phase response is used to determine DPOAE latency. The f_1-sweep DPOAE phase delay is composed of an anterograde and retrograde traveling-wave delay plus an intensity dependent filter build-up time at the site of OAE generation (Kimberley et al., 1993). DPOAE delay estimates in a f_2-sweep paradigm are longer than delay estimates in a f_1-sweep stimulation paradigm at the same frequency and intensity (O'Mahoney & Kemp, 1995; Bowman et al., 1996). The f_2-sweep and f_1-sweep phase delay difference is intensity and frequency dependent. On the basis of these observations it has been suggested that the f_2-sweep DPOAE phase delay is composed of a greater proportion of the filter build-up time at the site of DPOAE generation than the f_1-sweep delay. Bowman et al. (1996) have posited that a proportion of the DPOAE filter response can be isolated by subtracting f_1-sweep DPOAE delays from f_2-sweep delays at similar f_2 frequencies in normal adult ears.

A number of investigators have shown that the cochlear filter can be described by the impulse response of the basilar membrane when the impulse response is defined as the product of an nth order gamma function and a cosine (Goldstein et al., 1971; Eggermont, 1979). The shape of the cochlear filter is determined by the order of the gamma function under the assumption of minimum phase delay. The order of the gamma function can be calculated from the number of periods of delay to the peak of the impulse response.

This study examined DPOAE filter response properties obtained from f_2- and f_1-sweep DPOAE phase delay estimates at eight different f_2 frequencies (1.1-13.0 kHz) and five intensities in 60 normal hearing adult human ears. f_2- and f_1-sweep phase delay differences were calculated by subtracting the f_1-sweep delays from f_2-sweep delays for each subject. The basilar membrane impulse response was calculated from the mean DPOAE phase delay difference at each f_2 frequency and intensity.

In the present study, the mean phase delay difference ranges from 2.0 periods at 1.1 kHz (65 dB) to 17 periods at 9.2 kHz (45 dB). Delay increases as f_2 frequency increases and intensity decreases.



The long delays observed at low primary intensity levels and high frequencies are consistent with the findings Ruggero (1992a) which indicated that the build-up time of the impulse response of the basilar membrane reflects filter tuning properties. Sharply tuned responses at low intensities had longer build-up times than broadly tuned responses at high intensities. Eggermont (1979) has similarly shown that sharp cochlear filters have long impulse responses with several cycles of delay to the response maximum when filters are derived from narrow band AP latency in humans.. The CF normalized tuning curve bandwidth of single auditory nerve fibers decreases as CF increases (Ruggero 1992b). The long filter build-up times observed in this study at high f_2 frequencies may therefore reflect frequency dependent differences in the tuning of higher frequencies responses.

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THE EFFECTS OF NOISE MASKING ON THE CORTICAL AUDITORY EVENT-RELATED POTENTIALS TO SPEECH SOUNDS /BA/ AND /DA/.*

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Introduction. Cortical event-related potentials (ERPs) provide measures of the timing, sequence, and anatomical location of brain processing reflecting the detection and discrimination of sounds. Cortical ERP N1 occurs about 80-150 ms following stimulus onset, and reflects an obligatory response of the brain to stimulus onset or offset. Its major generators lie in the primary auditory cortex bilaterally. The mismatch negativity (MMN) begins during N1 and can lost longer than N1. This response, generated in the primary auditory cortex bilaterally, is a preconscious response elicited by the mismatch or physical difference between stimuli presented in an oddball paradigm. In the oddball paradigm (e.g., /ba, ba, ba, ba, ba, **da**, ba/, etc.), infrequently occurring "deviant" stimuli (e.g., /da/) are embedded in a series of frequently occurring "standard" stimuli (e.g., /ba/). MMN increases in amplitude and decreases in latency the greater the physical difference between the standard and deviant stimuli. ERP wave N2 (or N2b) occurs approximately 200 ms following the beginning of stimulus deviance, is usually present only in an attend condition, requires unpredictable stimulus presentation, and is seen for auditory and visual modalities (all in contrast to the MMN). Active attend and passive ignore conditions are employed to separate MMN from N2. ERP wave P3 (or P3b) occurs approximately 300-500 ms following the beginning of stimulus deviance. Like the N2, it requires unpredictable stimuli which are attended. N2 and P3 intracranial generators are multiple and not fully determined, but both reflect later processing of stimuli. N2 occurs when a stimulus mismatch has been consciously noted. P3 latency has been suggested to reflect stimulus evaluation time, or post-decision evaluation of the deviant stimuli. Thus, these ERP measures provide, in humans, measures of the timing (reflected by latencies), strength (reflected by response amplitude), and location (reflected by generators of the waves) of brain processing at different stages in the auditory system. The recordings are even more informative when recorded together with, and constrained by, behavioural measures, such as reaction time (RT) and discriminability (e.g., d' from %correct and false alarm data).

Many behavioural studies have investigated the effects of hearing loss, real or simulated by stimulus filtering or masking, on perception of speech. Little is known of changes in human brain processing of auditory stimuli occurring with hearing loss. In the present studies, we investigated the effects of decreased audibility, caused by either broadband (BBN) or by high-pass (HPN) noise masking, on cortical ERP and behavioural measures of detection and discrimination of the consonant-vowel syllables /ba/ and /da/. The /ba/ - /da/ stimuli were presented at 65 (Studies 1,2 & 3) and 80 (Studies 1 & 2) dB peak-to-peak equivalent (ppe) SPL.

Results/Discussion. In *study 1*, we investigated the effects of BBN maskers on cortical ERPs recorded in an active attend condition (N1, N2 and P3), as well as behavioural measures. BBN masking noise was presented at 50, 60 and 70 dB SPL for 65 dB stimuli, and at 60, 70 and 80 dB SPL for 80 dB stimuli. These maskers produced "flat" behavioural threshold elevations of 18, 25, 35 and 48 dB, respectively. The BBN maskers produced significant decreases (relative to a QUIET condition) in ERP amplitudes and behavioural discriminability. These decreases did not occur, however, until the noise masker intensity (in dB SPL) was equal to or greater than the speech stimulus intensity (in dB ppe SPL), that is, until speech-to-noise ratios were $\leq$ 0 dB. N1 remained present even after N2, P3, and behavioural discriminability were absent. In contrast to amplitudes, ERP and behavioural latencies showed significant decreases at higher (better) speech-to-noise ratios. Significant latency increases occurred when the noise maskers were within 10-20 dB of the stimuli. At these levels, *amplitudes* showed no change. Latency increases occurred with less masking for N1 than for P3 or behavioural reaction time.

Study 2 investigated the effects of HPN masking on the cortical ERPs recorded in an active attend condition (N1, N2 and P3), as well as behavioural measures. The HPN maskers resulted in sloping high-frequency losses, with pure-tone behavioural threshold elevations of 38 dB at the high-pass cutoff, and 50 dB one octave above the cutoff frequency. ERP results show that as the HPN cutoff frequency is lowered, ERP latencies increased and amplitudes decreased. The cutoff frequency where the changes first occurred and the rate of change differed for N1 compared to N2, P3 and the behavioural measures. N1 showed small and systematic changes. N2, P3 and RT/d' showed marked changes only when the HPN cutoff was below 2000 Hz.

Study 3 investigated the effects of HPN masking on the cortical ERPs recorded in a passive ignore condition (N1, MMN). Results for N1 are similar to those seen in Study 2. MMN and behavioural results showed a similar pattern to those seen for N2 and P3 in Study 2: marked changes (increased latency and decreased amplitude) only when the HPN cutoff was below 2000 Hz (i.e., at 1000 Hz). The marked changes in latency and amplitude of ERP waves MMN, N2, and P3 (and behavioural RT and d') occur when the HPN masker cutoff was lowered from 2000 to 1000 Hz. This is the region of the second formant frequency transition differentiating /ba/ from /da/. N1 changes with HPN masking were systematic, and lacked this marked change at a specific cutoff.

Conclusions. Several general conclusions are reached from these studies: (i) ERP and behavioural latencies are more sensitive to the effects of masking than are amplitude (or %correct) measures; (ii) Latencies increase and amplitudes (or %correct) decrease with masking. To the extent that the audibility changes with masking are similar to those of hearing loss, then hearing loss might be expected to produce similar changes. (We are currently assessing this in subjects with SNHL.) (iii) The effects of masking are different for ERP wave N1 compared to ERP waves MMN, N2 and P3, as well as for behavioural measures, reflecting the different functional significance of N1 compared to these other measures. N1 is an "obligatory" response, indicating detection by the brain of audible stimulus energy; it does not reflect discrimination of stimuli. MMN, N2, P3, and behavioural measures reflect detection *and* discrimination; (iv) changes in ERP waves MMN, N2, and P3 latency/amplitude with masking are strongly correlated with changes in behavioural measures.

These studies have implications both for basic research into brain processing of auditory stimuli and changes with masking and hearing loss, and for possible applications to clinical populations.

* *The studies presented in this Summary form part of Brett A. Martin's Ph.D. dissertation.*

TEMPORAL PROCESSING IN THE YOUNG AND OLD AUDITORY CORTEX

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A common complaint among the elderly is a difficulty in understanding speech. One central factor that may contribute to this difficulty, is a deterioration in the ability to process the dynamic aspects of speech such as the formant transitions. Formant transitions, which are characterized by a change in frequency over time, enable us to discriminate one consonant sound from another. A number of investigators have suggested that processing speed deteriorates with age. For the aging auditory system, this deterioration may be manifest as a deficit in processing time-varying sounds that contain rapidly changing sounds, such as the formant transitions. Thus, if temporal processing deteriorates with age, then our ability to recognize speech could be seriously affected. Unfortunately, the nature of temporal processing in the aged auditory system, has not been explored very extensively, particularly in relation to speech comprehension. Thus, the primary goal of this study was to explore one aspect of the neural mechanisms underlying the effects of aging on temporal processing. A stimulus which lends itself well to studying this type of processing is the frequency modulated (FM) sweep which in many respects, FM sweeps resemble formant transitions found in a variety of communication signals.

METHODS. Experiments were conducted on 12 young (3-4 months) and 3 old (24-30 months) male Long Evans hooded rats. Rats were anaesthetized and maintained at a surgical level of anaesthesia throughout the experiment with Equithesin (3 mg/kg i.p.). Animals were placed in a modified head holder and a craniotomy performed. Earphones connected to speculae were placed within 3 mm of the tympanic membranes. All extracellular single unit recordings were conducted in a sound attenuating chamber.

Rats were initially stimulated monaurally through the contralateral ear with tone bursts (100 ms duration with a 10 ms rise/decay time, 700 msec interstimulus interval) to determine characteristic frequency (CF) and threshold, followed by linear FM sweeps ranging from 150 Hz to 45.0 kHz (upward-directed) and 45.0 kHz to 150 Hz (downward-directed) at speeds of 0.8, 0.3, and 0.05 kHz/msec. All stimuli were generated and data collected by a Macintosh computer using the MALab system.

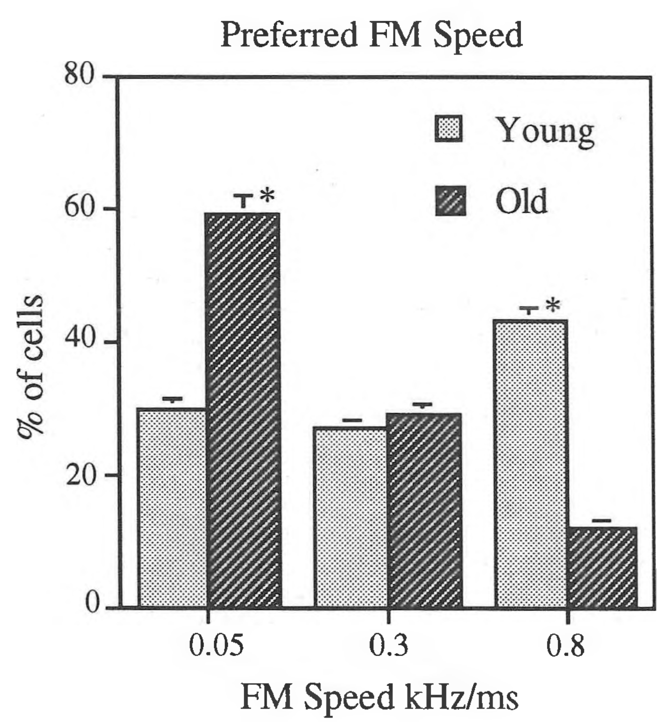
RESULTS. A total of 60 units were studied of which 40 were recorded from young rats and 20 from old rats. The average CF was 14.2 kHz for cells recorded from young rats and 11.0 kHz for those tested in old rats.

The results show that the majority of units in both age groups (Y: 37/40, O: 17/20) were selective for the direction and/or speed of FM sweep. There were 3 units in each age group that did not appear to be sensitive to FM sweeps and thus, were excluded from further analysis.

Preferred Speed. All of the FM-sensitive units examined were sensitive to speed of frequency modulation. Perhaps the most striking result observed was a significant difference in the distribution of preferred speed between young and old animals. In the young animals, the largest number of cells (43%; n=16) responded most vigorously to the fast speed while the preferred speed of the remaining units was more evenly distributed between the medium (27%; n=10) and slow speed (30%; n=11). In contrast, the majority of units (59%; n=10) recorded from the old animals responded best to the slow speed while relatively few units preferred the fast speed (12%; n=2).

Direction Selectivity. Using a criterion where the cell \ responded at least twice as vigorously in one direction as in the opposite direction, almost half (49%) of the units recorded from young rats showed no preference for the direction of FM sweep. Of the cells that exhibited direction selectivity, all of them preferred upward-directed sweeps. For the old rats, the majority of units (71%) were direction-selective with 58% of these units preferring upward-directed sweeps.

DISCUSSION. The results of the present study provide one of the first demonstrations of a difference between young and old animals in cortical temporal processing speed of dynamic stimuli. The difference in FM speed preferences observed between the neurons recorded from young and old rats may reflect a deterioration of temporal processing speed in the aging auditory cortex.



TRANSIENT AND PERIODICITY CODING IN TWO FIELDS IN AUDITORY CORTEX OF THE CAT

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Periodically modulated sounds are coded temporally in the peripheral auditory system, but this changes gradually into a rate coding for more central nuclei. In the auditory cortex, however, the average firing rate does change very little with modulation frequency or click repetition rate whereas the synchronized firing rate does. Several dramatic differences in the coding of periodic sounds in cortex are observed: first of all the modulation frequencies or repetition rates that can be followed are much lower than in subcortical areas and generally less than 50 Hz, secondly different cortical areas appear to have different abilities to code for modulation frequency. A complication arises because of prominent EEG rhythms in the 8-12 Hz range which tend to interact with stimulus modulation frequencies in that range and limit the capacity to follow higher rates. This suggests that network properties determine to a large extent the preferred modulation frequency.

The procedures have been described in detail in previous publications (e.g., Eggermont, 1994) and we will only repeat those aspects that are important for the interpretation of the results. Cats were under ketamine anesthesia. Stimuli were presented from a speaker placed 55 cm in front of the cat's head. The sound-treated room was anechoic for frequencies above 625 Hz. The recording procedures were similar to those described previously (Eggermont, 1994). Three tungsten micro electrodes were independently advanced into the primary auditory cortex (AI) and anterior auditory field (AAF) using remotely controlled motorized hydraulic microdrives.

Periodicity coding was investigated for single units and local field potentials (LFP) in the middle layers of cat AI and AAF using click trains and amplitude modulated (AM) noise or AM CF-tones (Eggermont, 1994). Modulation frequencies were between 1 and 64 Hz. The click-following and AM-following capacity of the neurons was evaluated from the synchronization modulation transfer functions obtained by Fourier transformation of the period histograms. The best modulating frequency (BMF) was defined as that click rate or AM frequency for which the synchronized firing rate was maximal.

Results are from 62 simultaneous recordings in AI and AAF resulting in 186 multi-unit (MU) records and 186 LFP recordings. Seventy single units with good locking to the periodic click trains were isolated from the MU records. Simultaneous recordings from AI and AAF were used to ensure similar anesthesia conditions. We demonstrate that the best modulation frequency (BMF) for periodic click train stimulation is strongly correlated with the EEG spindle frequency. In contrast to earlier reports (Schreiner and Urbas, 1988), which found that the average BMF for sinusoidally modulated CF-tones was larger in AAF than in AI, no dramatic differences between AI and AAF were found with respect to best click rate. No differences were found in limiting rate, the magnitude of the temporal modulation transfer function is 50% of maximum, either. If anything the BMFs were on average 1-2 Hz higher in AI, but this was not statistically significant.

The previous findings that AAF may be better tuned to sinusoidal CF-tone modulation was further explored. LFP data for simultaneous recording in AI and AAF for three periodic stimuli were compared. We used periodic click trains, exponential-sine AM noise and exponential-sine AM CF-tones. The LFPs in AAF were slightly lower tuned for clicks, responded somewhat less to high AM rates for modulated noise, but were far better responsive (up to modulation frequencies of 32 Hz) to CF carrier tones than the simultaneously recorded LFPs in AI. The same was found for multi and single-unit activity.

Psychophysically, temporal modulation transfer functions obtained for sinusoidally AM noise show an decrease in sensitivity above 30 Hz by about 3-4 dB/oct (Viemeister, 1979) whereas sinusoidal AM of pure tone carriers show a great dependence on the carrier frequency (Fassel and Kohlrausch, 1995). For instance at 1 kHz, the tMTF is low pass with a cut-off frequency (-6 dB) of about 150 Hz, but for a 3 kHz carrier there is an enhanced sensitivity around 200-300 Hz and a cut-off point at about 400 Hz. In these studies the high-frequency slope of the tMTF was about 8-12 dB/oct. This contrasts with the small effect found in electrophysiological studies where the most frequently found BMFs are below 20 Hz, regardless of the carrier used: the highest BMFs are still at least a factor 10 below those found in psychophysical studies. The presently available electrophysiological data thus suggests that the physiological substrate for the psychophysical performance is likely found in subcortical structures.

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EFFECTS OF DEAFNESS AND COCHLEAR IMPLANT USE ON DEVELOPMENT OF HUMAN AUDITORY FUNCTION

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Profound deafness in children disrupts and delays the development of language and communication skills. The extent to which these skills are affected depends on the age of onset and the duration of deafness (Busby et al., 1989). For profoundly deaf children, meaningful auditory sensation may be restored by electrical stimulation of auditory nerve fibers through a cochlear implant. By examining children fitted with a cochlear implant, it may be possible to determine the effects that deafness and the subsequent reintroduction of stimulation have on the maturation of those areas of the brain that are deprived of their normal sensory input.

A direct and objective measure of auditory cortical maturation can be obtained by recording electrophysiological activity such as the cortical auditory evoked response. Early components of the cortical auditory evoked response reflect stimulus-onset detection in sensory cortex (Courchesne, 1978). Maturational changes in cortical activity are reflected in age-related decreases in the latency of these components.

To assess the effects of deafness and cochlear implant use on cortical maturation, age-related changes in the latency of the positive peak, P_1, were examined in implanted and normal-hearing children and adults. For normal-hearing subjects, cortical activity was evoked by applying 100 µs voltage pulses to a headphone. For implant users, special computer hardware that bypassed the speech processor was used to deliver 200 µs/phase biphasic current pulses directly to the implant. Each stimulus consisted of a brief train of ten acoustic clicks or electric pulses. Click stimuli were presented monaurally to the left ear of normal-hearing subjects at approximately 65 dB above threshold. Stimulation levels were set individually for each implant user at a loud but comfortable level. Evoked responses were recorded at 30 electrode locations on the scalp. Evoked response latencies are reported for the vertex electrode Cz, as this is the most commonly used electrode location in both research and clinical settings.

Large wave-shape differences are apparent between the younger children (< age 10) and adults for both normal-hearing and implanted subjects. For the six and seven year olds, the AER is dominated by a large positive peak at about 100 ms. The general pattern of age-related latency decrease for this peak suggests that it is probably equivalent to the adult P_1. Age-related latency changes for P_1 are non-linear, with the difference between children and adults decreasing exponentially for both normal-hearing and implanted children. Curve fit analyses of the P_1 data showed that the rate of latency decrease was the same for normal-hearing and implanted children. However for implanted children, the age at which P_1 latency becomes adult-like is delayed. To further examine the nature of this delay, the implanted children were divided into short (≤ 2 years), medium 4-6 years), and long-term (8-9 years) auditory deprivation groups based on the duration of deafness (see Figure 1). A near perfect correspondence exists between the duration of deprivation and P_1 maturational delay if it is assumed that some time elapses between the onset of deafness and its detection.

These findings suggest that in absence of cortical activation during the period of deafness, maturation of activity in auditory cortex does not progress. However, once stimulation of the auditory pathway is restored, some, if not all, maturational processes resume a normal course.

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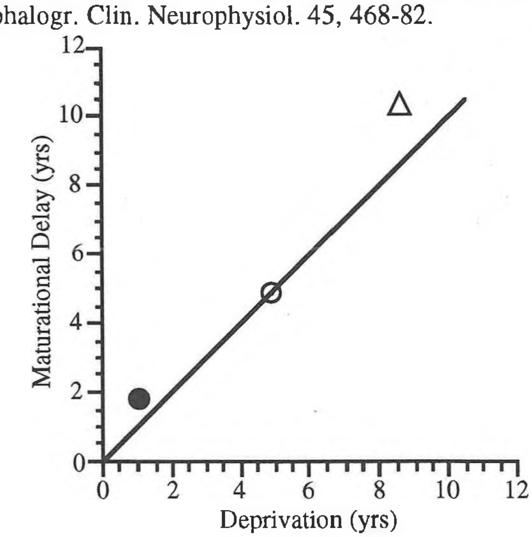

Figure 1: Maturational delay for P_1 latency as a function of duration of auditory deprivation (time between detection of deafness and time of implantation).

THE EFFECT OF AGE AND STIMULUS PARAMETERS ON THE OPTIMUM f_2/f_1 RATIO FOR DPOAEs.

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Distortion Product Otoacoustic Emissions (DPOAE) are commonly used to study the pre-neural activity of the cochlea in adults and infants. The parameters used to generate the optimum DPOAE have been studied in adults (see Probst et al., 1991 for review). One of these parameters is the optimum f_2/f_1 ratio, this ratio between the primary tones can influence the maximum amplitude of the DPOAE. An early report indicated that the optimum f_2/f_1 ratio was 1.25 (Kemp & Brown, 1983). There has been agreement across studies that in adults the optimum f_2/f_1 ratio is approximately 1.2. Harris et al. (1989) found the optimum ratio to be 1.22, but they also described an inverse relationship between optimum f_2/f_1 ratio and the frequency of the DPOAE with larger ratios required at 1 kHz than at 4 kHz. In addition, they reported that the optimal ratio changed as a function of the intensity of the primaries in such as way as the intensity of the primaries increased, the ratio used to elicit a maximal DPE amplitude also increased. Although this ratio was from studies in the adult, it was used as the f_2/f_1 ratio when testing infants. Brown et al. (1994) investigated the optimum f_2/f_1 ratio at 2 and 4 kHz in term-born infants and found the ratio to be 1.2 which was in agreement with the optimum ratio for adults.

This study measured DPOAEs in 3 infant groups (30-33 wks.; 34-37 wks.; 38-42 wks.) and an adult group. Each group had 40 subjects with the exception of the youngest preterm group which only had 25 infants. DPOAEs were measured from one ear of each of the 145 subjects tested. Each subject was tested at six f_2 frequencies from 10 to 1.7 kHz in half octave steps. The f_2 tone is fixed at a given frequency and the f1 tone is swept around it from a ratio of 1.3 to 1.1. The optimum f_2/f_1 ratio is determined as the ratio between the two primary tones (f_2 and f_1) which creates the largest amplitude $2f_1$-f_2 (DPE) emission for a given f_2 frequency. The level of the primaries for all groups were 45 and 60 dB SPL for f_2 and f_1 respectively. In addition, the adult group also had DPOAEs measured at primary levels between 20 and 45 dB SPL (in 5 dB steps) for the f_2 frequency while keeping f_1-f_2 =15 dB.

We found that the dominant factor in determining the optimum f_2/f_1 ratio was the f_2 frequency and that there were no significant effects of age. The frequency effect showed that the optimum f_1 was relatively lower for lower f_2 frequencies (i.e. a higher f_2/f_1 ratio) and relatively higher for higher f_2 frequencies (i.e. a lower f_2/f_1 ratio).

The ratio becomes larger as the f_2 frequency decreases. There appears to be a natural break in the optimum ratio between 3.5 and 4.9 kHz. It is at this point (approximately 4 kHz) where the ratio changes, above and below this point the ratios are significantly different from each other. The optimum ratio above f_2 >4 kHz is smaller (1.18) than the ratio below f_2 <4 kHz (1.22). This is different than previous studies who reported a constant ratio across frequencies but consistent with Harris et al. (1989) who also noted an inverse relationship. This may be explained by the change in the shape of the excitation profiles along the BM. As indicated by tuning curves, the low frequencies are more broad tuned than high frequencies and change the slope of the profile at the apical end. The overlap on the BM associated with a particular ratio in the high frequencies will be different than in the low frequencies. These results indicate that the ratio is stable and that there is no effect of maturation. This suggests that cochlear excitation profiles, which are assumed to be responsible for that ratio are maturing early.

An adult group was used to study intensity effects on the optimum f_2/f_1 ratio. The results suggest a trend for lower optimum ratios for lower f_2 intensities. The effect of intensity on the optimum f_2/f_1 ratio revealed that as intensity increases the optimum ratio also increases, this occurred at all but two frequencies (f_2 = 3.4 and 7.0 kHz). As the intensity of the primary tones increases resulting in a broadening of the BM tuning, maximum overlap occurs at a lower f_1 frequency or a frequency further away from the f_2 frequency place. This occurs so that the maximum overlap will be maintained. There must be some balance for the two primary tones between overlap and suppression. Overlap of the two primary tones causes the largest non-linear interaction whereas suppression of one tone over another tends to reduce the importance of f_2 and also the emission.

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NEW CONSIDERATIONS AND TECHNIQUES IN AIRCRAFT NOISE ABATEMENT

During the last six years, planning for the abatement and mitigation of aircraft noise has changed significantly. A large portion of this change has been brought about by the failure of well-intentioned early plans to meet the general noise reduction goals anticipated by both planners and neighboring communities.

Early federally-funded attempts to establish noise compatibility programs at airports within the United States (as well as in many other countries) consisted, in many cases, of a menu of broadly defined measures affecting large groups of airport neighbors. Local disputes over project priorities, inadequacy of project funding to maintain hoped-for schedules, and changes of public perceptions as to what constitutes an "objectionable" noise level have frequently combined to result in less than expected levels of success.

The benefits of time and experience have led many airports to reevaluate their current programs. Sponsors have often broadened, eliminated or refined ongoing mitigation measures to better respond to the realities of implementation. The boundaries of areas eligible for mitigation programs such as acquisition or sound insulation have often been based on average noise levels grounded in imprecise estimates of future aircraft operational characteristics. Use of new and refined data acquisition and dissemination techniques such as Airport Noise and Operations Monitoring Systems, GIS land use tracking, and internet home pages are being coupled with better assessments of aircraft flight characteristics and more definitive applications of local surface conditions to result in more definitive mitigation than previously available.

While the DNL metric has remained the measure of choice in programs funded by the U.S. Federal Aviation Administration, community advocacy groups are making strong arguments for greater consideration of other measures. Single event L_{max} and SELs, as well as durations above predefined dBA levels are being introduced into many noise compatibility plan updates as tools to evaluate the effectiveness of alternatives or suitability for various types of mitigation.

The feasibility to implement desirable changes to the air traffic patterns at airports has not been thoroughly examined in most of the first round of planning studies. Although flight track changes are often recommended in abatement plans, their relocation is frequently impeded by broader issues of how those tracks feed into the regional airspace framework. The combination of noise modeling and airspace simulation modeling is in its infancy as a technique to evaluate regional noise issues. Work accomplished on the East Coast plan and underway in the Chicago metroplex is providing direction to these analyses.

This paper provides an overview of many of the changes which are taking place in the management of aircraft noise and its impacts on airport neighbors. Studies conducted by the author and/or his firm will serve as the focus for illustration, but will be supplemented by leading innovative techniques of aircraft noise assessment now being undertaken by others.

ACOUSTICAL SCALE MODELLING OF HIGHWAY NOISE BARRIERS

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Introduction

The noise attenuation of scale-model roadside barriers was studied using a scale-model road configuration and an ultrasonic air-jet to recreate a line source of traffic. Scale-model materials were found that adequately represented outdoor ground surfaces. Scale-model noise barriers of three types were tested: vertical walls; earth berms; and earth berms crested by a wall. A-weighted Insertion Losses (ILAs) were obtained by applying an A-weighted traffic-noise spectrum before integrating over the 80–2500Hz full-scale third-octave bands.

Scale-Model Materials

Scale-model materials were selected by measuring the material's Excess Attenuation at scale-factors of 1:20, 25, 31.5, 40, and 50. A material is a good scale-model surface when the normal impedance of the scale-model surface at ultrasonic frequencies is equivalent to the normal impedance of the ground surface at full-scale frequencies. Theoretical Excess Attenuations were calculated using a sound-propagation model that describes the normal impedance using the Delany & Bazley [1] flow-resistivity model. An array of scale-factor versus effective flow-resistivity values was generated for each test material, and the residuals between the measured and theoretical curves were calculated for each cell of the array. An optimal scale-factor of 1:31.5 was selected, in conjunction with the selection of 3 model materials to simulate earth berms and soft ground(150–300 c.g.s.-rayl), and vertical walls or roadways(20000 c.g.s.-rayl). Roadways were simulated by varnished particle board, walls by dense polystyrene, and both earth berms and soft ground were simulated using extruded polystyrene. The impedance of berm surfaces was also altered using dense polystyrene and felt.

Insertion-Loss Results

The most significant factors affecting barrier IL were: earth-berm slope; relative earth-berm and berm-crest wall height; and earth berm surface impedance. Wall thickness did not significantly affect ILA, as long as the wall is thick enough to make sound transmission negligible. Walls were found to outperform earth berms by about 2dBA. Variations in the top profile of the earth berm, from a wedge to a flat-top of 1m or 2m width or a round top of 1m or 2m radius, produced at most an 0.8 dBA improvement in berm performance. Examining Fig.1, it is apparent that as earth-berm slope becomes shallower, the Insertion Loss is *reduced* in the vicinity of the 250Hz third-octave band. The application of the A-weighted traffic-noise spectrum suppressed many of the differences observed in the third-octave Insertion Losses, since its peak power was in the 1000Hz third-octave band. For all three earth-berm surface impedances, a shallower slope resulted in a decreased ILA, with the vertical wall being a limiting case for the grass-berm tests. In Fig.2, for earth berms with a crest wall, shallower slopes *improved* Insertion Losses so that they were as high as for a pure wall, or even up to 1dBA more effective, in the case of an earth berm of 3m height topped by a 1m crest wall. When a berm's height represented a smaller proportion of the total 4m barrier height, the affect on attenuation was reduced. The softening of an earth berm's surface produced substantial improvements in the ILA. However, when a wall was present the benefits of softer berm slopes were not as substantial; the beneficial effects of adding a wall or softening the earth berm's surfaces were not independent of one another.

[1] M.E. Delany and E.N. Bazley. "Acoustical Properties of Fibrous Absorbent Materials". *Applied Acoustics*, 3:105–116, 1970.

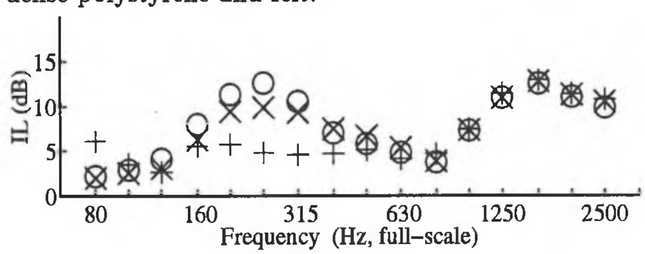

Figure 1: Measured third-octave Insertion Losses for wedge-shaped grass berms of 4m height with slopes of 1.5,2, and 3 to 1: (o) 1.5:1, IL=9.0, ILA=7.0; (x) 2:1, IL=8.5, ILA=7.2; (+) 3:1, IL=7.9, ILA=7.0

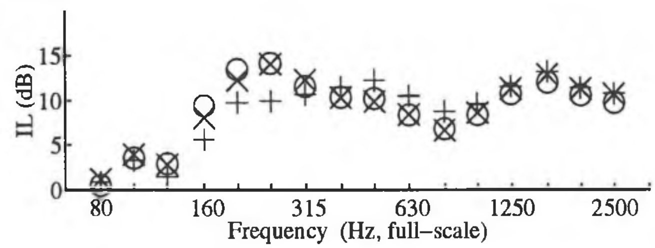

Figure 2: Measured third-octave Insertion Losses for flat-topped grass berms of 3m height and of 2m top width, and with slopes of 1.5:1, 2:1, and 3:1, topped by a 1m high crest wall that preserves a total barrier height of 4m: (o) 1.5:1,IL=10.1, ILA=8.9; (x) 2:1, IL=10.2, ILA=9.1; (+) 3:1, IL=9.9, ILA=10.2

DEVELOPMENT OF COMMUNITY NOISE STANDARDS FOR A NEW TOWN PLANNED FOR SOUTHERN VANCOUVER ISLAND

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1. BACKGROUND

A new community of approximately 12,000 residents is planned for the eastern slopes of the Malahat Mountain near Victoria B.C. The 650 Hectare site includes the former Bamberton Cement Plant property, part of which is currently utilized by Tilbury Cement as a barge-unloading, bulk storage and truck distribution centre for southern Vancouver Island. Since the Tilbury Cement operation is intended to be a long-term "neighbour", it was essential to assure its compatibility with the new community that will grow up around it over the next 15 to 25 years. The developers, South Island Development Corporation, asked the author to review the noise standards proposed by Tilbury Cement to determine if compliance with them would permit future residents to fully enjoy their unique living situation and would prevent any negative impact on the marketability of the homesites.

2. AMBIENT NOISE ENVIRONMENT

The location of the site on wooded slopes overlooking a narrow arm of the Saanich Inlet would, in the absence of the Tilbury operation, result in ambient noise levels being controlled by a combination of natural sources and the activities of the future residents. Such future background noise levels are not then expected to exceed 40 to 45 dBA during the day and 30 to 35 dBA during the night.

3. CEMENT OPERATION NOISE

The Tilbury Cement operation produces two principal types of noise: 1. steady, tonal noise from compressors and dry bulk blowers which transfer product from barges to dock silos and from the dock silos to the truck load-out silos (42 dBA at 200 m), and intermittent, impulsive noises created by the sudden release of compressed air within the truck-loading silo dust collection system and from the valve which controls the flow of cement into trucks (70 to 85 dBAI at 10 m from base of silo). Both types of noise would then, by their natures, tend to be more easily perceptible and therefore more annoying than noises which were neither tonal nor impulsive.

4. TILBURY'S NOISE STANDARDS

The noise standards proposed by Tilbury were based on noise by-laws long in use in Vancouver and other GVRD municipalities. Limits were specified for continuous and non-continuous sounds from stationary equipment. During Normal Hours of Operation (06:00 to 20:00 hours except for cement barge off-loading which may be a 24-hour operation), these limits were 55 dBA and 60 dBA for continuous sounds at residential and commercial points of reception respectively, and 75 dBA for non-continuous sounds at all points of reception.

5. REVISIONS TO NOISE STANDARDS

It was felt that the noise standards proposed by the operator, while perhaps valid as general guidelines for more urban areas, did not reflect the sensitivity of the Bamberton site nor the specific nature of the industrial noises being produced. In particular, since the proposed standards stated that cement trucks would comply with Federal and Provincial noise regulations, there was felt to be no need to include a non-continuous sound limit. Instead, in recognition of anticipated low background noise levels, limits for most continuous sounds during normal operating hours were reduced to 45 dBA at residential locations. Barge off-loading noise however, due to its potential 24-hour duration, was assigned a 35 dBA limit. Further, any continuous noise containing a perceptible tone would be assigned a 5 dBA penalty.

Finally, a 45 dBAI limit was assigned to impulsive noises from stationary sources. This was obtained by applying a 5 dBAI "low background noise" penalty to the 50 dBAI limit recommended in the Health and Welfare Canada's "National Guidelines for environmental Noise Control".

6. COMPLIANCE

While the recommended revised noise limits are much lower, given the current noise output from stationary equipment plus the planned 60 m minimum residential setback distance, the operator should not experience any undue hardship in achieving compliance.

THE UTILITY OF AVIATION NOISE IMPACT ASSESSMENT STUDIES IN MANAGING AVIATION NOISE

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1. INTRODUCTION

This paper summarizes Aviation Noise Impact Assessment Study (ANIAS) procedures developed and used by the author and describes in general terms the value of these studies. The paper illustrates the utility and success of these studies in managing a wide-variety of aviation noise impacts. These studies can be performed by airport authorities and acoustical consultants, among others, for a wide variety of aviation noise impact applications and can complement other aviation noise management methods such as Noise Exposure Forecast analyses and noise control studies. ANIAS produce a comprehensive and factual account of actual aviation noise impacts, resulting in improved noise management to the benefit of all stakeholders.

2. METHODOLOGY

ANIAS are an interrogative noise management tool that address small-scale issues with sufficient application flexibility to address actual and specific conditions. Such a method provides necessary technical information to support decision-making about specific aviation noise impacts based on local conditions, and do so at a reasonable cost.

ANIAS can be used to describe both quantitatively and qualitatively the actual noise climate in the vicinity of the study area. This is accomplished through consultation and co-operation with the stakeholders, measurement, observation and research. Several types of analysis are performed to render the most comprehensive and accurate impact assessment possible. Analyses are tailored to produce information that will best address the objectives of each particular study. The following types of analyses can be performed:

Comparison of aviation noise levels (single events and dosages) with those of other common environmental and ambient noise sources in the study area; comparison of aviation noise levels with single event activity interference criteria; comparison of aviation noise levels with noise dosage criteria; investigation and explanation of the evoked or potential community response (to help understand and respond to human response) and, suggestions for mitigation measures.

Quantitatively, ANIAS describe the actual noise climate in terms of single event (SEL and L_{AMax}) impacts and noise dosage (L_{eq}) impacts. Representative measurements and calculations are made for all significant and typical noise sources. For both types of measures the impact of aviation noise is compared to that produced by the other noise sources. Furthermore, average single-event values are compared with activity interference criteria such as speech interference and sleep disturbance for both aviation and non-aviation noise sources. Average dosage values can be compared with provincial indoor and outdoor noise limits. Frequency analyses are also beneficial. Of particular use, is the real-time frequency analysis capability provided by some easily-portable measuring instruments.

Qualitatively, ANIAS can include an explanation of the frequency characteristics of the noise sources, their temporal variations and any pertinent contextual information. This information can be associated with the quantitative results and sociological findings to explain observed or potential human response and hence improve the impact assessment.

3. CONCLUSION

The ANIAS has proven itself to be a cost-effective, valuable and efficient analytical tool for TC planners, airport planners, managers and decision-makers (including local and regional authorities), in managing aviation noise. The methodology employed in these studies is flexible enough to be tailored to address the objectives of any specific noise impact assessment study and, is designed to yield results which can be directly used in noise management decision-making. A host of varied and self-explanatory reports have been written by TC which demonstrate the utility of Aviation Noise Assessment Studies. Copies of these reports are available from the author. A more detailed version of this paper is also available from the author.

NEW TECHNOLOGIES IN AIRPORT NOISE & FLIGHT TRACK MONITORING

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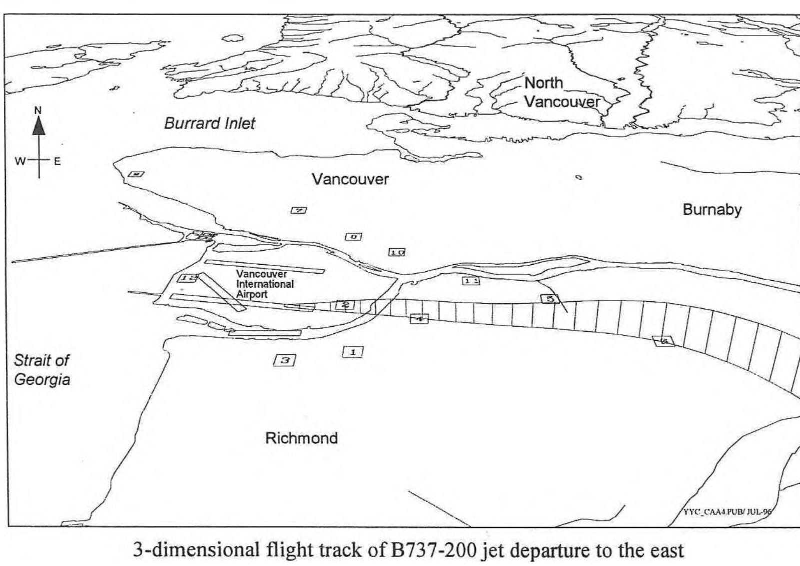
The art and science of airport noise monitoring has evolved to more advanced levels in the last decade. This has allowed those airports with new noise monitoring and radar flight tracking technologies to gain a better understanding of the true contribution of aircraft noise in the surrounding communities. In the past, permanent outdoor noise monitoring stations were strategically placed with the intent of measuring the noise of aircraft flyovers. However, without detailed analysis and sometimes human observation, it was not possible to distinguish a noise event between an aircraft or other noise source.

The new system acquired by the Vancouver International Airport Authority measures noise in real-time at 12 permanent outdoor noise monitoring terminals (NMT). The system also has a real-time interface to the air traffic control radar, that gives information about all aircraft operations including the type, airline, position, altitude and speed. The system automatically correlates noise events at the NMTs with the appropriate aircraft. In addition, the new system integrates all the analysis tools which were once separate, including noise monitoring, aircraft flight track investigations, statistical analysis, weather analysis, compliance monitoring, complaint investigation, mapping and reporting.

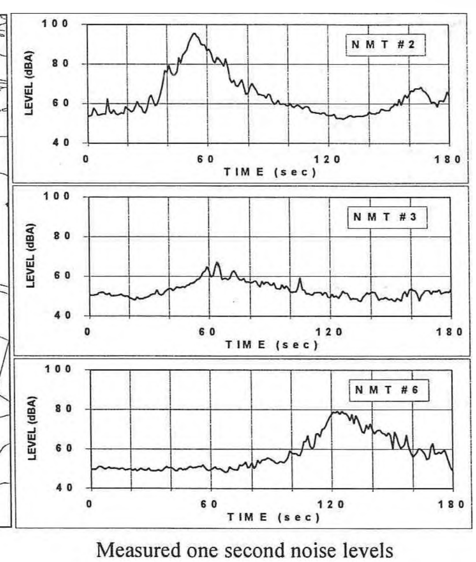
The figure below is an example 3-dimensional flight track of a B737-200 jet aircraft, which depicts the departure profile from the main runway of Vancouver International Airport (YVR). The measured one second time history at three of the NMTs is presented in the accompanying charts. The charts illustrate that the noise level for close overhead flyover at NMT #2 is about 20 dBA greater than for the higher altitude flyover at NMT #6, and 30 dBA greater than for the sideline noise experienced at NMT #3. The system automatically calculates the SEL (sound exposure level) by integrating the energy above a prescribed threshold for each time history curve.

Installation of this system was a commitment made to the recommendations of the Environmental Assessment Review Process (EARP) Panel for the new parallel runway at YVR. The commitment included stringent operational restrictions on the new runway, which will be monitored using the system. About one year of baseline noise and flight track data will have been collected by the system before the new runway opens in November 1996. This data will be be complemented by a baseline noise social survey, which was undertaken in August 1995. The intent of this survey was to ascertain the psycho-acoustic reaction of communities to airport noise, and compare the findings with other airports.

The system in conjunction with the overall noise management program is an important benefit, as it has enhanced the airport's ability to answer technical queries from airlines and air traffic control, as well as to address community concerns.



3-dimensional flight track of B737-200 jet departure to the east



Measured one second noise levels

THE INFLUENCE OF EXHAUST EMISSION CONTROLS ON THE COMBUSTION NOISE LEVEL OF AN I.D.I DIESEL ENGINE

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University of Calgary
Calgary, Alberta, T2N 1N4

The pronounced "knock" of a diesel engine is generally considered to be the result of differences in the form of the cylinder pressure development as opposed to variations in engine structure. The more abrupt the change of slope from the compression curve upon combustion, the greater the combustion noise level. Typically, diesel engine combustion noise is the result of the high rate of pressure rise in the initial stages of combustion, while the subsequent rise of pressure up to the peak pressure, which is at a much lower rate, has a negligible effect [1]. In general, it is the premixed volume of the fuel and air that is the fundamental factor that directly influences the level of combustion noise. This volume can be linked to the length of the ignition delay; a long delay is generally assumed to be an indicator of high combustion noise levels due to the larger amount of premixed fuel and air at the time of ignition.

The combustion noise research to date has primarily focused upon identifying the various engine design and operational parameters that affect the level of combustion noise produced. Unfortunately, with the introduction of more stringent exhaust emission standards, there is a lack of information regarding the influence of the ensuing pollution control techniques on the levels of combustion noise generated. One of the more promising exhaust emission control methods for diesel engines is the application of exhaust gas recirculation (EGR), which involves mixing part of the exhaust gas with the fresh intake air charge to act as a diluent in the combustion process. However, the influence of the exhaust gases in the intake mixture on the combustion noise level is unknown. Consequently, an experimental investigation to determine the effects of these non-conventional atmospheres on the level of combustion noise was carried out using a specially designed test rig at the University of Calgary.

In order to delineate the effects of different intake working fluid compositions on the level of combustion noise the experimental investigation was performed in four segments: (i) the effects of atmospheric air, (ii) the effects of various intake O_2/N_2 ratios, (iii) the effects of CO_2 addition ($CO_2/O_2/N_2$ mixtures), and (iv) the effects of synthetic EGR.

(i) Naturally Aspirated Condition: The results obtained when operating with a conventional, naturally aspirated atmosphere were similar to those published in the literature [1]. The combustion noise level increased with speed at a rate of 31.8 dBA/decade and was generally independent of load. Additionally, for the conditions investigated the combustion noise level was found to be a function of the maximum rate of pressure rise (dP/dt_{max}).

(ii) Variable O_2/N_2 Ratios: The experimental results obtained when operating with various O_2/N_2 ratios indicated that the combustion noise level exhibited a slight decreasing trend as the intake O_2 concentration was increased. However, unlike the naturally aspirated condition, these noise levels were influenced by the engine load. The decrease in the combustion noise level was believed to be primarily due to the decreased ignition delay as the intake O_2 concentration was increased. However, the decrease in the combustion noise level was not at significant as expected. This was attributed to the mechanism by which the ignition delay was altered. Variations in the intake mixture O_2/N_2 ratios modified both the chemical and physical elements of the ignition delay. However, the most influenential factor that affected the ignition delay under these conditions was the variation of the O_2 concentration which altered both the diffusion/mixing rates between the fuel and O_2 and the chemical oxidation reactions that occur to form a flammable mixture.

(iii) CO_2 Addition: The results obtained when operating with various $CO_2/O_2/N_2$ mixtures indicated that the combustion noise level exhibited a generally increasing trend whose magnitude was also dependent upon the engine load. Similar to the O_2/N_2 mixtures, the change in the combustion noise level was attributed to a change in the ignition delay, however, the mechanism by which the ignition delay period was altered was different. The presence of CO_2 in the intake, at a constant intake O_2 concentration of 23.3% by mass, primarily influences the chemical portion of the ignition delay by altering the specific heat capacity of the mixture. Therefore, the increase in the combustion noise level was much more significant when compared to the O_2/N_2 data because during the extended ignition delay period a larger volume of premixed fuel and air was formed. Hence upon ignition rapid burning rates and high rates of pressure rise result.

(iv) Synthetic EGR: The first three stages of this experimental program have been designed to illustrated how various diluents affect the combustion noise in order to provide some insight into the effect of EGR on the combustion noise level. The introduction of EGR involves a reduction of the intake oxygen concentration and an increase in the carbon dioxide concentration. Therefore, based on the above experimental results, it is expected that EGR would increase the combustion noise level. However, the preliminary results indicated that the combustion noise level of the engine remained constant and independent of load when operating under EGR conditions. Although, at excessive levels of EGR (above 80% recirc) the combustion noise level increased which was attributed to the poor combustion characteristics within the cylinder.

Similar to the previous stages, the combustion noise characteristics were determined by the ignition delay period. An increase in the amount of EGR resulted in an increase in the delay period because of the lack of O_2 which hindered the diffusion/mixing rates between the fuel and O_2 as well as the chemical oxidation reactions that are necessary to form a flammable mixture. However, as the quantity of EGR increased the intake CO_2 concentration also increased thereby modifying the polytropic exponent of the engine compression characteristics which lowered the pressure at the start of combustion. This lower pressure counteracts the effects of the longer ignition delay and hence the combustion noise level remains constant.

The results of this experimental investigation have shown that when operating on both naturally aspirated and non-conventional atmospheres, the main engine parameter that influenced the level of combustion noise was the ignition delay period, and hence the volume of premixed fuel and air at the time of ignition. However, as illustrated by the EGR results, it would also appear that the pressure at the start of combustion, as determined by the compression characteristics of the cycle, is also an important factor. Nevertheless, contradictory to some non-conventional atmospheres, the application of EGR to the intake mixture of a diesel engine for exhaust emission control does not affect the level of combustion noise generated, thereby enhancing it's potential for future use.

REFERENCES

[1] T. Priede, et. al., 1968, "Combustion Induced Noise in Diesel Engines", Diesel Engineers and User Association, Publication 317.

PREDICTION MODEL "IMPACT" TO DETERMINE THE ACOUSTIC EFFECT OF NOISE BARRIERS ALONG HIGH SPEED ROADS

WEIXIONG WU

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This study is performed to evaluate the model "IMPACT" of prediction of outdoor and traffic noise for barrier attenuation, used by *'Centre de recherche en aménagement et développement (C.R.A.D.), Université Laval'*. This prediction model was compared to the detailed measurements and the calculation model *"Road and Rail Noise: Effects on Housing"* (Canada Mortgage and Housing Corporation).

An experimental campaign was organized in Duberger, Québec City, nearby the freeway *Boulevard de la Capitale*: free flow, grooved asphalt, one side noise barrier, 60912 vehicles/day for Capitale-Est, 63528 vehicles/day for Capitale-Quest, 110km/h vehicle speed. Sound levels were measured at twenty positions at a height of 1.5 meters and at various distances in front of and behind the barrier. In all measuring sites, noise levels were measured in terms of L_{eq} dB(A). Measurement periods lasted a minimum 10 min/run. During the L_{eq} measurements, traffic volume was counted lane-by-lane and classified as automobiles and heavy trucks. Traffic speeds were obtained by a car traveling with traffic.

In order to calculate traffic sound levels, "IMPACT" required: the roadway-barrier-receiver geometry, traffic flow conditions, traffic sound attenuation rates, and adjustments to account for additional attenuation. The highway traffic sound levels reported by "IMPACT" were A-weighted and they were presented in terms of the description L_{eq}. The geometry used in "IMPACT" was 3-dimensional. For the digital process of the basic mapping and for the treatment of the modeling outputs, "IMPACT" was equipped with a module for data exchange with the software *Microstation*. In order to provide the program with the coordinates of the receivers, roadways and barriers, a plan of the side under investigation was obtained. The plan showed all existing structures, (houses, roads, freeway, etc.) as well as the elevations and features of the surrounding terrain. This model produces an output file for use by the noise level design program.

Using the model *Road and Rail Noise: Effects on Housing* (CMHC) to calculate L_{eq} for the roadway traffic noise prediction for the verification of the "IMPACT" results was also studied. The reference level was calculated at a distance of 30 m from the center of the road. Step-by-step procedure: traffic count, base noise level, correction for road gradient, determination of equivalent source height, correction for distance, and correction for barriers. All the corrections were then included to take into account the usual propagation variables.

CONCLUSION

The differences between experimental measurements and the results of the "IMPACT" model are appreciably low for the noise level. The agreement is satisfactory between the CMHC model and "IMPACT" mode. The results are shown in Figure.1.

There are some differences between "IMPACT" and measured noise levels at certain positions behind the barrier, such as No.2, No.7, No.12, and No.17. It seems the effects of sound transmission varies through the barrier. It is recommended that the research be supported and actively followed for an upcoming project that will develop an "IMPACT" model in sound transmission of the barrier.

Even in these cases the differences between "IMPACT" and measured noise levels are within the range of measured ones. This lets one say that the determination of the sound field with provisional models is satisfactory enough to avoid expensive and useless experimental campaigns of measurements.

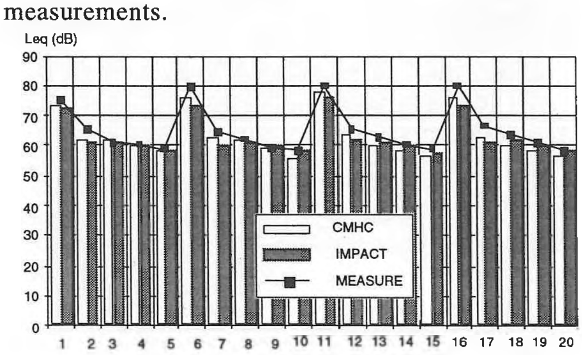
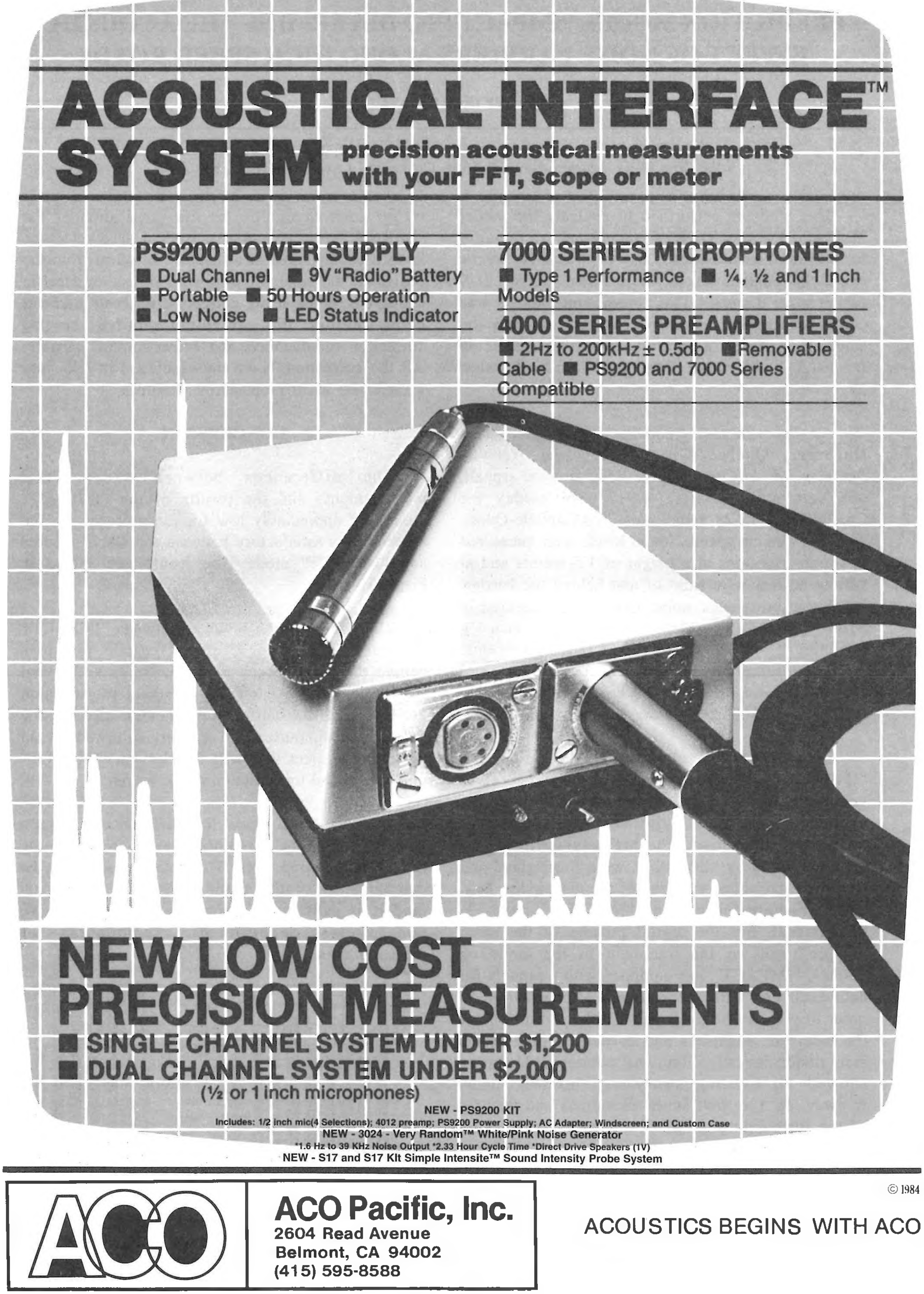


Figure.1 Comparison of the *Leq* data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ACOUSTICS BEGINS WITH ACO

A Bernoulli-Euler Stiffness Matrix Approach for Vibrational Analysis of Linearly Tapered Beams

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[1] Dept. of Mech. Eng., Laval University, Québec, G1K 7P4 [2] l'INSA de Rouen, Rouen, France

1. Introduction. A particular area of interest is the vibrational behaviour of blade-type structures in which tapered beams have been, widely, studied. In finite element formulations[1], polynomial shape functions are often used. A deviation from this practice will pay dividends if improved accuracies can be obtained by using other shape functions . This is the case of the Dynamic Stiffness Matrix (DSM) appoach, in which, the frequency dependent shape functions are used. In this paper an appoach leading to the DSM of Bernoulli-Euler linearly tapered beams is presented.

2.Method. The differential equation for the lateral free vibrations of a Bernoulli-Euler tapered beam is:

$$(H_{fy}(x)w_{,xx})_{,xx} + m(x)w_{,tt} = 0.(1)$$

where, $H_{fy}(x) = EI_y(x)$, $m(x) = \rho A(x)$; w and I_y represent displacement and the second moment of area. Considering harmonic vibrations, eq.(1) becomes;

$$H_{fy}(w_{,xxxx} - \alpha^4 w) - (\underbrace{H_{fyDEV}}_{(H_{fy}-H_{fy}(x))} w_{,xx})_{,xx} + \underbrace{m_{DEV}}_{(m-m(x))} \omega^2 w = 0. \quad (2)$$

where; $\alpha^4 = m\omega^2/H_{fy}; H_{fy} = H_{fy}(x)_{average}; m = m(x)_{average}$.

The non-dimensionalized weak form for the element k, associated to eq.(2) can be written as:

$$W^k{}_{ND} = \int_0^1 (\frac{\gamma_k}{\bar{l}_k^3}\bar{w}''\,\bar{\delta w}'' - \mu^2 \bar{m}_k \bar{l}_k \bar{w}\bar{\delta w})d\eta_k + DEV. \; ; \quad (3)$$

$$DEV. = -(\frac{1}{\bar{l}_k^3})\int_0^1 (\gamma_{DEV}\bar{w}''\,\bar{\delta w}'')d\eta_k + (\mu^2 \bar{l}_k)\int_0^1 (\bar{m}_{DEV}\bar{w}\bar{\delta w})d\eta_k$$

where; $\gamma_k = EI_k/EI_r$; $\bar{l}_k = l_k/L$; $\mu^2 = m_r\omega^2 L^4/EI_r$;

$\bar{m}_{DEV} = m_{DEV}/m_r$; $(r \equiv reference\ value)$.

Eq.(3) is written in the equivalent form:

$$W^k{}_{ND} = \frac{\gamma_k}{\bar{l}_k^3}\int_0^1 \underbrace{(\bar{\delta w}'''' - \bar{\alpha}^4\bar{\delta w})}_{(*)}\bar{w}d\eta_k + \frac{\gamma_k}{\bar{l}_k^3}[\bar{\delta w}''\,\bar{w}' - \bar{\delta w}'''\,\bar{w}]_0^1 + DEV. \quad (4)$$

Then, $\bar{\delta w}$ and $\bar{w}$ are approximated so that (*) vanishes:

$$\bar{\delta w} = < P(\eta) > \{\bar{\delta a}\}; \bar{w} = < P(\eta) > \{\bar{a}\} \quad (5)$$

Considering the four nodal variables: $\bar{w}_1; \bar{w}_1'; \bar{w}_2; \bar{w}_2'$ as $< \bar{\delta a} >$, we obtain $\{\bar{\delta w}_n\} = [P_n] * \{\bar{\delta a}\}$ and hence, the approximation(5) in nodal variables is written as:

$$\bar{w}(\eta) = < P(\eta) > [P_n]^{-1}\{\bar{w}_n\} = < N(\eta) > \{\bar{w}_n\} \quad (6)$$

and by using eq.(4) the element DSM is obtained as:

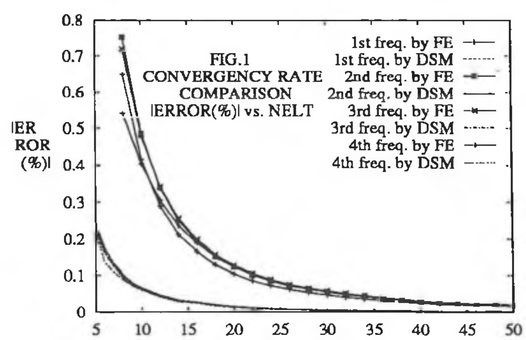
$$W^k{}_{ND} = < \bar{\delta w}_n > \underbrace{([K_{RD}]_{uni} + [K_{RD}]_{DEV})}_{[K_{RD}]}\{\bar{w}_n\}; \quad (7)$$

where; $[K_{RD}]_{uni} = \frac{\gamma_k}{\bar{l}_k^3}[\{N'''\}_0\{-N''\}_0\{-N'''\}_1\{N''\}_1]$;

and $[K_{RDij}]_{DEV} = -(\frac{1}{\bar{l}_k^3})\int_0^1 \gamma_{DEV} N_i'' N_j'' d\eta_k + (\mu^2\bar{l}_k)\int_0^1 \bar{m}_{DEV} N_i N_j d\eta_k.$

Elementary matrices are assembled, and the bisection method[2] is used to find natural frequencies of beams.

3.Results. The DSM is used to find natural frequencies of a cantilever tapered beam so $A_L = \frac{A_0}{2}$(FIG.1), and the results have been compared to those found by F.E. Much better convergency rate is found by DSM.



References: [1]Dhatt G.and TouzotG.- Finit Element Method Displayed. [2]Wittrick W.H.and Williams F.W. Int.J.mech.Sci. 1970. Vol.12, pp781-791.

VIBRATION ISOLATION OF POWERED OVERHEAD DOORS

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Powered overhead doors are becoming more and more prevalent in commercial and high density residential buildings. Unfortunately, a large majority of the installations result in vibration induced noise in adjacent areas, in particular the space directly above the installation. The continual annoyance of this mechanical noise results in tenants complaining to the Building Owners and Developers, holding payments and, in some cases, litigation.

The sources of the noise are strictly due to the vibration of the drive assembly and the rollers running in the track, transmitting energy directly into the structure. Of the two sources, the former is generally the more significant.

The obvious solution to both problems is simply to break the energy path between the vibrating component and the structure. However, during site inspections, vibration isolation materials were often found to be installed on drive assemblies, yet there were still noise complaints. Closer inspections revealed that the 'isolation' materials were simply inadequate or poorly installed. Typical problems found were as follows:

- Electrical grommets that were never intended as vibration isolators were used.
- Resilient materials were placed above and/or below the vibrating frame of the drive assembly, but either the shank or the nut of the attachment bolt was contacting the vibrating attachment point.
- Where a proper isolation grommet or two rubber washers and a cylinder were used, the cylinder section had been torn away or seriously deformed by the threads of the anchor bolt due to the torque created by the drive/chain assembly during operation.
- Contractor had over-torqued the anchor bolts and extruded the isolation component.
- Armoured electrical conduit carrying the power cables to the drive were flexed to a point that it provided a vibration path to the structure.

The solution was to have appropriate vibration isolation mounts, such as Mason series BRA, installed as they eliminate Contractor installation errors, have bridge bearing elastomeric elements for longevity, and are 'fail safe' and seismically rated. Although these mounts may be significantly under-loaded in some applications, they are still effective due to the impedance mis-match through the thick elastomeric elements. A typical mounting of the drive assembly is shown in Figure 1 below, with a cut-away of the typical mount recommended in these applications shown in Figure 2.

The secondary problem of the rollers running in the tracks has been successfully addressed by either isolating the track from structure using grommets (or pad/cylinder isolation components) or by replacing the standard steel rollers with nylon rollers. The latter is considerably less expensive and usually effective. In the rare case that nylon rollers do not solve the problem, isolating the track as well, has effected the solution, without exception.

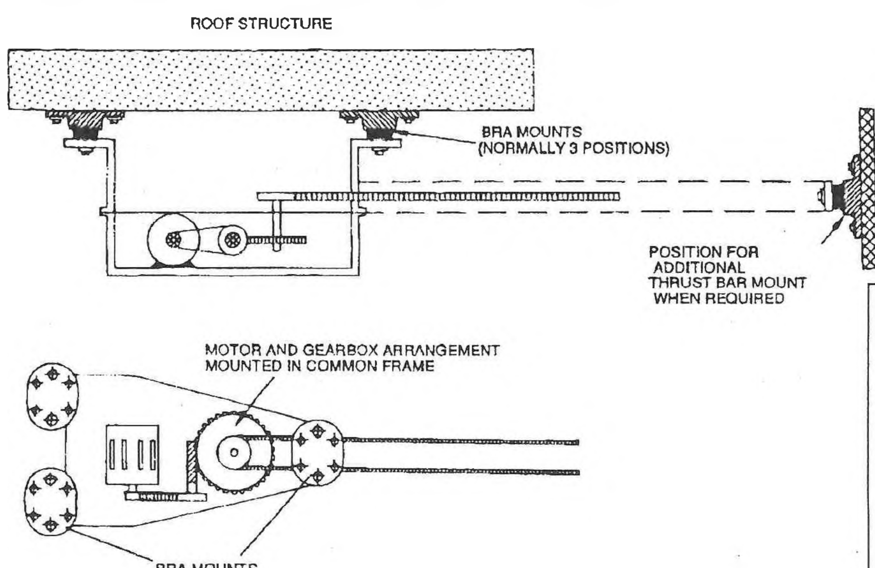

FIGURE 1: Elevation and Plan View of Typical Installation

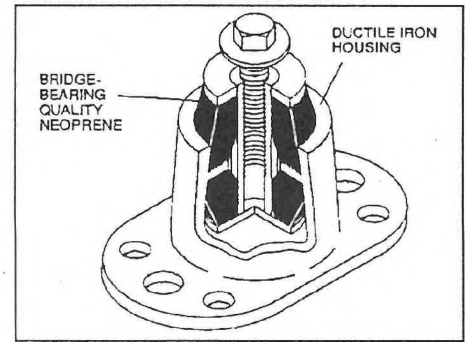

FIGURE 2: Typical BRA-type Isolation Mount

A VIBRATION PROBLEM

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3.0 Advantages of Magnetic Bearings
4.0 A Vibration Problem
5.0 Another Vibration Problem
6.0 Conclusions

1.0 Introduction

Nova Gas Transmission has an installed fleet of 31 pipeline compressors running on magnetic bearings. The total experience in excess of 1 million running hours was accumulated over the past 10 years of operation. The intent of this paper is to present a very brief introduction to magnetic bearings, and to show two test cases in which the bearings were used to determine the cause of machine vibrations and prevent impending failures.

2.0 Brief Review of Magnetic Bearing Technology

The development of magnetic bearings involved several areas of knowledge including: Mechanical engineering design, magnetics, electronics, controls, and rotordynamics. It wasn't until these subject areas were well developed that magnetic bearings could come to maturity.

The magnetic bearing system consists of the bearing actuators (magnets), feedback sensors, and the control system.

The actuators can be either electromagnets (active) or permanent magnets in combination with electromagnets. All of the bearings used in the Nova system are of the active kind. The bearings can provide stiffness and damping tailored to the requirements of the mechanical design. Most systems will have the actuators located at 45 degrees with the vertical axis, sharing the weight for horizontal rotors. Since they can be imbedded in the process fluid, the possible arrangements of the bearings is limited only by the imagination.

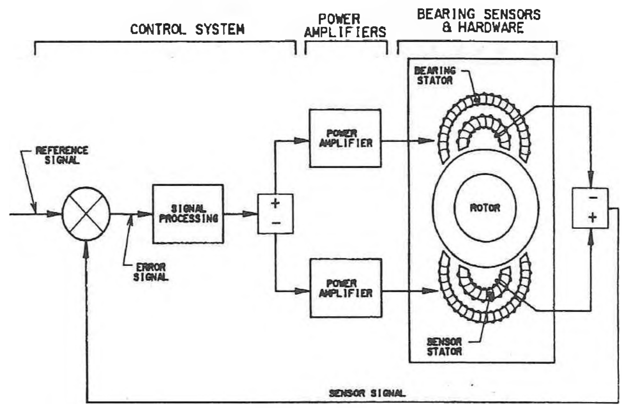
Auxiliary bearings or bushings are necessary to support the rotor when at rest or in case of extreme loads.

The sensors are used to provide position feedback to the control system (and sometimes velocity and force). Sensorless bearings are also being studied. There are generally 5 axis of control, 4 radial and 1 axial. These can be independent or not according to the control scheme.

The controller has to provide stability to the dynamic system which consists of the rotor and bearings over a wide frequency range. This range is typically dictated by the rotating speed of the machine and its rotordynamic characteristics. Digital controllers are used in most installations nowadays.

The rotor center running position can be prescribed and precisely controlled.

The vibrations transmitted to the surroundings can be minimized. Alternately vibrations transmitted to the rotor can be reduced.



3.0 Advantages of magnetic bearings

Immersion in the fluid.
Minimal consumption of energy.
Tailoring of the rotordynamics.
High speed operation. High DN numbers.
High temperature operation.
Vibration suppression. Shock resistance.
Reduced friction losses. 1-2% efficiency increase.
Ability to do field balance at speed. Modal balance.
Diagnosis of process conditions. Can determine the forces from impeller and fluid interaction.
Low maintenance requirements.
Reduce forces transmitted to structures with the use of ABS or optimized feed forward controls.(Autobalance mode of control)
Prediction of running shaft distortion.

4.0 A Vibration Problem

One of our large compressors was exhibiting a high vibration which started at a certain rpm and went up very quickly to the point of shut down. This happened after a new wheel had been installed in this machine.

We suspected something changing in the dynamic system, and decided to implement the auto balance mode of control to see if the change was happening in the rotor. Our suspicions were confirmed, the vibration persisted in spite of the shaft being suspended in zero stiffness bearings (at running speed). We suggested that the wheel was coming loose on the shaft when the centrifugal force due to rotation exceeded the preload given by the interference fit. This was confirmed upon disassembly.

5.0 Another Vibration Problem

One other compressor showed step increases in vibration at running speed. After the second time this happened, we decided to open the unit and inspect. The compressor wheel had two pieces of vanes broken off. Other five vanes were found cracked. The diagnostic capability of the bearing prevented a much more costly incident.

6.0 Conclusions

Magnetic bearings are a mature technology which can greatly enhance the capabilities of rotating equipment in general. For the past year and a half, the reliability of the units in the Nova pipeline system has been around 99.98% which compares favorably with oil bearing technology.

SOME OBSERVATIONS CONCERNING THE VALIDITY OF IIC MEASUREMENTS FOR A FLOATING FLOOR IN A JUDO PRACTISE HALL

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In a judo practise hall, or *dojo*, the practise area of the floor is covered by mats called *tatami*. This traditional kind of Japanese floor covering prevents injury when the players are forcibly thrown to the floor in the course of the practice of this martial art. However, in modern buildings the original floor is typically a concrete slab rather than the traditional wooden structure, and so a false floor is often installed between the *tatami* and the concrete floor.

The false-floor structure was based on an earlier design used for a high degree of noise isolation for marine tugboats. The original floor used SCE-41 closed cell neoprene foam strips 2" (50 mm) wide and 1" (25 mm) thick, laid out in a grid 2 foot by 2 foot (0.6 m X 0.6 m), overlain with two layers of 3/8" plywood, glued and screwed. The resilience of the foam strips are rated as 2 to 5 psi, and are referred to as 3 psi.

There was very little data available, so the final design was optimized for the impact loading typical in this sport, through field experimentation. The calculated resilience (spring constant) for the final design was approx. 1.6×10^6 N/m, calculated using the nominal resilience of the foam. The layout of the final design of the sub-floor is shown in Figure 1.

After the floor had been in place for over a year, IIC measurements were taken of 1) the original floor, 2) the false floor, 3) the finished *tatami* surface, and 4) the *tatami* surface with a wooden plate (a table turned upside down) on top of it. A standard ASTM tapping machine was used for all of these tests, but the tests used only one source location and so are not strictly in compliance with the ASTM standard (E1007-90).

Table 1 shows the field IIC (FIIC) ratings.

IIC Ratings of Layers of *Dojo* Floor	
Layer Tested	FIIC Rating
1. Bare Concrete Floor	41
2. On Sub-floor	45
3. On *Tatami*	57
4. On Table on *tatami*	45

Although condition 4 spreads the impact loading out over a greater area, the noise levels in the dojo increased substantially. The test may be more indicative of the airborne noise isolation than of the impact actually transmitted to and through the floor.

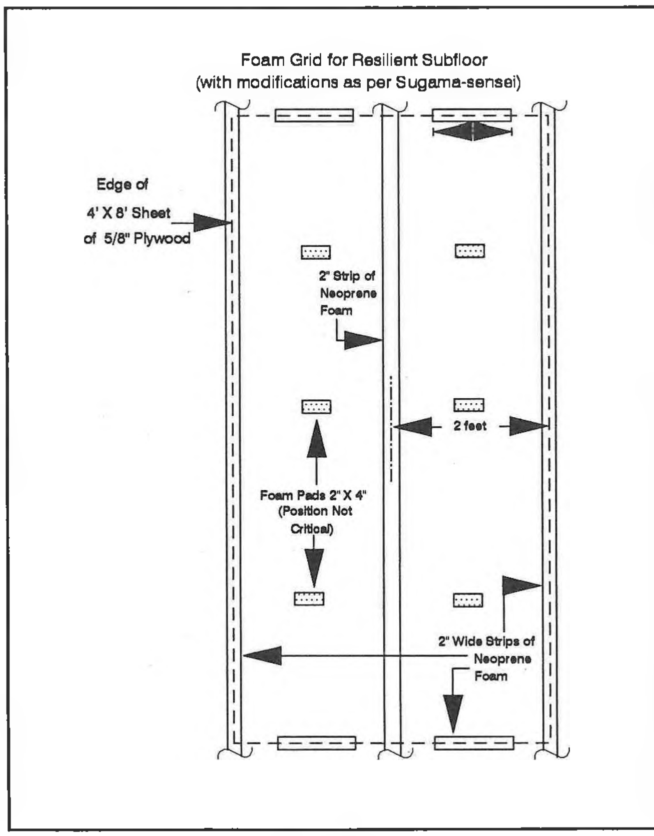


Figure 1

NEWS / INFORMATION

CONFERENCES

The following list of conferences was mainly provided by the Acoustical Society of America.

1996

4-6 September: British Society of Audiology Annual Conference, Winchester, UK. Contact: A.R.D. Thornton, MRC Inst. of Hearing Research, Royal South Hants Hospital, Southampton, Hants SO14 0YG, UK; Fax: +44 1703 825611; E-mail: (JANET) ardt@soton.ac.uk

6-7 September: Seventh National Symposium on Ultrasonics, Mepco, Nagar, Tamil Nadu, India. Contact: S. Jain, Ultrasonics Society of India, National Physical Laboratory, Dr. K. S. Krishnan Road, New Delhi 110 012, India; FAX: 91 11 575 2678.

16-21 September: XLIII Seminar on Acoustics, Ustron-Beskidy Mts., Poland. Contact: Institute of Acoustics, Silesian Technical University, Krzywoustego 2, 44-100 Gliwice, Poland.

18 September: Intensity Measurements, Utrecht, The Netherlands. Contact: NAG, Postbus 162, 2600 AD Delft, The Netherlands.

18-20 September: International Conference on Noise and Vibration Engineering, Leuven, Belgium. Contact: L. Notré, K. U. Leuven PMA, Celestijnenlaan 300B, 3001 Leuven, Belgium; Fax: +32 16 32 29 87; E-mail: lieve.notre@mech.kuleuven.ac.be; WWW: http://www.mech.kuleuven.ac.be/pma/events/isma.html

23-25 September: FASE Symposium "Transport Noise", St. Petersburg, Russia. Contact: J. Thoen, FASE Secretariat, K. U. Leuven~ATF, Celestiknenlaan 200D, 3001 Leuven, Belgium; Fax: +32 16 32 79 84; E-mail: jan.thoen@fys.kuleuven.ac.be

25-26 September: Autumn Meeting, Acoustical Society of Japan, Okayama, Japan. Contact: ASJ Ikeda Building, 2-7-7 Yoyogi, Shibuya-ku, Tokyo, 151 Japan; FAX: +81 3 3379 1456.

25-27 September: 33rd Conference on Acoustics "Building and Architectural Acoustics", Prague, Czech Republic. Contact: CsAS Technicka 2, 166 27 Praha 6, Czech Republic; Fax: +42 2 311 1786.

26-28 September: 5th Session Russian Acoustical Society "Problems of Geoacoustics: Methods and Instruments", Moscow, Russia. Contact: Acoustics Institute, RAS, 4 Shevrnik, Moscow 117036, Russia; Fax: +7 095 126 8411; E-mail: bvp@asu.acoins.msk.su

29 September - 2 October: Noise-Con 96, Bellevue, WA. Contact: Noise-Con 96 Conference Secretariat, Engineering Professional Programs, 3201 Fremont Avenue North, XD-51, Seattle, WA 98103, Tel: 206-543-5539; FAX: 206-543-2352; E-mail: uw-ept@engr.washington.edu

29 September - 3 October: Centennial meeting of the American Academy of Otolaryngology--Head and Neck Surgery, Washington, DC. Contact: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314. Tel.: 703-836-4444; FAX: 703-683-5100.

CONFÉRENCES

La liste de conférences ci-jointe a été offerte en majeure partie par l'Acoustical Society of America.

1996

4-6 septembre: Conférence annuelle de la Société britannique d'audiologie, Winchester, Royaume Uni. Renseignements: A.R.D. Thornton, MRC Inst. of Hearing Research, Royal South Hants Hospital, Southampton, Hants SO14 0YG, UK; Fax: +44 1703 825611; E-mail: (JANET) ardt@soton.ac.uk

6-7 septembre: Septième symposium national sur les ultrasons, Mepco, Nagar, Tamil Nadu, Inde. Renseignements: S. Jain, Ultrasonics Society of India, National Physical Laboratory, Dr. K. S. Krishnan Road, New Delhi 110 012, India; FAX: 91 11 575 2678.

16-21 septembre: 43e séminaire d'acoustique, Ustron-Beskidy Mts., Pologne. Renseignements: Institute of Acoustics, Silesian Technical University, Krzywoustego 2, 44-100 Gliwice, Poland.

18 septembre: Mesures d'intensité, Utrecht, Pays-Bas. Renseignements: NAG, Postbus 162, 2600 AD Delft, The Netherlands.

18-20 septembre: Conférence internationale de l'ingénierie du bruit et des vibrations, Leuven, Belgique. Renseignements: L. Notré, K. U. Leuven PMA, Celestijnenlaan 300B, 3001 Leuven, Belgium; Fax: +32 16 32 29 87; E-mail: lieve.notre@mech.kuleuven.ac.be. WWW: http://www.mech.kuleuven.ac.be/pma/events/isma.html

23-25 septembre: Symposium du FASE sur le bruit des transports, St. Petersburg, Russie. Renseignements: J. Thoen, FASE Secretariat, K. U. Leuven~ATF, Celestiknenlaan 200D, 3001 Leuven, Belgium; Fax: +32 16 32 79 84; E-mail: jan.thoen@fys.kuleuven.ac.be

25-26 septembre: Rencontre d'automne, Société d'acoustique du Japon, Okayama, Japon. Renseignements: ASJ Ikeda Building, 2-7-7 Yoyogi, Shibuya-ku, Tokyo, 151 Japan; FAX: +81 3 3379 1456.

25-27 septembre: 33e conférence d'acoustique "Acoustique architecturale et de bâtiment", Prague, République Tchèque. Renseignements: CsAS Technicka 2, 166 27 Praha 6, Czech Republic; Fax: +42 2 311 1786.

26-28 septembre: 5e session de la Société russe d'acoustique "Problèmes de géoacoustique: méthodes et instrumentation", Moscou, Russie. Renseignements: Acoustics Institute, RAS, 4 Shevrnik, Moscow 117036, Russia; Fax: +7 095 126 8411; E-mail: bvp@asu.acoins.msk.su

29 septembre - 2 octobre: Noise-Con 96, Bellevue, WA. Renseignements: Noise-Con 96 Conference Secretariat, Engineering Professional Programs, 3201 Fremont Avenue North, XD-51, Seattle, WA 98103, Tel.: 206-543-5539; FAX: 206-543-2352; E-mail: uw-ept@engr.washington.edu

29 septembre - 3 octobre: Rencontre centenaire de l'académie américaine d'otolaryngologie -- chirurgie de la tête et du cou, Washington, DC. Renseignements: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314; Tel.: 703-836-4444; FAX: 703-683-5100.

3-6 October: Fourth International Conference on Spoken Language Processing, Philadelphia, PA. Contact: ICSLP 96, Applied Science & Engineering Laboratories, A.I. duPont Institute, P. O. Box 269, Wilmington, DE 19899, Tel.: 302-651-6830; TDD: 302-651-6834; FAX: 302-651-6895; E-mail: ISCLP96@asel.udel.edu; WWW: http://www.asel.udel.edu /speech/icslp/html

7-11 October: 1996 Acoustics Week in Canada, Calgary, Canada. Contact: Dr. E. Slawinski, Department of Psychology, University of Calgary, 2500 University Drive NW, Calgary, AB, T2N 1N4. Tel.: 403-220-5205; FAX: 403-282-8249; E-mail: eslawins@acs.ucalgary.ca

24-27 October: Reproduced Sound 12, Windmere, U.K. Contact: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

3-6 November: 1996 IEEE International Ultrasonics Symposium, San Antonio, Texas, USA. Contact: J.S. Schoenwald, Rockwell International Science Center, Mail Code A9, 1049 Camono dos Rios, Thousand Oaks, CA 91358, USA. FAX: +31.10.414.7988.

13-15 November: Australian Acoustical Society Annual Conference, Brisbane, Australia. Contact: R. Palmer, P.O. Box 150, Mount Ommaney, Queensland 4074, Australia.

17-22 November: ASME International Mechanical Engineering Congress & Exposition, Atlanta, GA. Contact: Gary H. Koopmann, Ctr. for Acoustics and Vibration, 157 Hammond Bldg., Penn State Univ., University Park, PA 16802, Tel.: 814-865-2761; FAX: 814-863-7222; E-mail: ghk@kirkof.psu.edu

21-24 November: Autumn Conference--Speech and Hearing, Windmere, U.K. Contact: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

2-6 December: Third Joint Meeting of the Acoustical Society of America and the Acoustical Society of Japan, Honolulu, HI. Contact: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org, WWW: http://asa.aip.org

8-13 December: 14th World Conference on Non-Destructive Testing, New Delhi. Contact: B. Jaj, Metallurgy and Materials Group, Indira Gandhi Centre for Atomic Research, Kalpakkam 603102, India; E-mail: dmg@igcar.iitm.emet.in

1997

27-28 February: Penn State Ultrasonic Transducer Engineering Workshop, Newport Beach, CA. Contact: Donna Rode, SPIE, P.O. BOX 10, Bellingham, WA 98227-0010, Tel.: 360-676-3290; E-mail: donnar@mom.spie.org or K. Kirk Shung, 231 Hallowell Bldg., Penn State Univ., University Park, PA 16802, Tel.: 814-865-1407; E-mail: kksbio@engr.psu.edu

17-19 March: Spring Meeting ASJ, Kyoto, Japan. Contact: ASJ Ikeda Building, 2-7-7 Yoyogi, Shibuya-ku, Tokyo, 151 Japan; FAX: +81 3 3379 1456.

2-4 April: International Symposium on Simulation, Visualization, and Auzalization for Acoustic Research and Education, Tokyo, Japan. Contact: M. Morimoto, Faculty of Engineering, Kobe University, Rokko, Nada, Kobe 657, Japan; Fax: +81 78 881 2508.

13-16 April: 23rd International Symposium on Acoustical Imaging, Boston, MA. Contact: Sidney Lees, Bioengineering Dept., Forsyth Dental Ctr., 140 Fenway, Boston, MA 02115; FAX: 617-262-4021; E-mail: slees@forsyth.org

3-6 octobre: 4e conférence internationale sur le traitement de la langue parlée, Philadelphie, PA. Renseignements: ICSLP 96, Applied Science & Engineering Laboratories, A.I. duPont Institute, P. O. Box 269, Wilmington, DE 19899, Tel.: 302-651-6830; TDD: 302-651-6834; FAX: 302-651-6895; E-mail: ISCLP96@asel.udel.edu; WWW: http://www.asel .udel.edu/speech/icslp/html

7-11 octobre: Semaine canadienne d'acoustique 1996, Calgary, Alberta, Canada. Renseignements: Dr. E. Slawinski, Department of Psychology, University of Calgary, 2500 University Drive NW, Calgary, AB, T2N 1N4. Tel.: 403-220-5205; FAX: 403-282-8249; E-mail: eslawins @acs.ucalgary.ca

24-27 octobre: Reproductions sonores 12, Windmere, Royaume Uni. Renseignements: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

3-6 novembre: Symposium international de 1996 de l'IEEE sur les ultrasons, San Antonio, Texas. Renseignements: J.S. Schoenwald, Rockwell International Science Center, Mail Code A9, 1049 Camono dos Rios, Thousand Oaks, CA 91358, USA. FAX: +31.10.414.7988.

13-15 novembre: Conférence annuelle de la Société australienne d'acoustique, Brisbane, Australie. Renseignements: R. Palmer, P.O. Box 150, Mount Ommaney, Queensland 4074, Australia.

17-22 novembre: Congrès et exposition d'ingénierie mécanique de l'ASME, Atlanta, GA. Renseignements: Gary H. Koopmann, Ctr. for Acoustics and Vibration, 157 Hammond Bldg., Penn State Univ., University Park, PA 16802, Tel.: 814-865-2761; FAX: 814-863-7222; E-mail: ghk@kirkof.psu.edu

21-24 novembre: Conférence d'automne - parole et audition, Windmere, Royaume-Uni. Renseignements: Institute of Acoustics, P.O. Box 320, St. Albans, AL1 1PZ, U.K.

2-6 décembre: 3e rencontre conjointe de l'Acoustical Society of America et de l'Acoustical Society of Japan, Honolulu, HI. Renseignements: ASA, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; FAX: 516-576-2377; E-mail: asa@aip.org; WWW: http://asa.aip.org

8-13 décembre: 14e conférence mondiale sur les tests non-destructifs, New Delhi, Inde. Renseignements: B. Jaj, Metallurgy and Materials Group, Indira Gandhi Centre for Atomic Research, Kalpakkam 603102, India; E-mail: dmg@igcar.iitm.emet.in

1997

27-28 février: Séminaire d'ingénierie de Penn State sur les transducteurs ultrasoniques, Newport Beach, CA. Renseignements: Donna Rode, SPIE, P.O. BOx 10, Bellingham, WA 98227-0010, Tel.: 360-676-3290; E-mail: donnar@mom.spie.org or K. Kirk Shung, 231 Hallowell Bldg., Penn State Univ., University Park, PA 16802, Tel.: 814-865-1407; E-mail: kksbio@engr.psu.edu

17-19 mars: Rencontre du printemps ASJ, Kyoto, Japon. Renseignements: ASJ Ikeda Building, 2-7-7 Yoyogi, Shibuya-ku, Tokyo, 151 Japan; FAX: +81 3 3379 1456.

2-4 avril: Symposium international sur la simulation, visualisation et l'auzalisation pour la recherche et l'éducation en acoustique, Tokyo, Japon. Renseignements: M. Morimoto, Faculty of Engineering, Kobe University, Rokko, Nada, Kobe 657, Japan; Fax: +81 78 881 2508.

13-16 avril: 23e symposium international sur l'imagerie, Boston, MA. Renseignements: Sidney Lees, Bioengineering Dept., Forsyth Dental Ctr., 140 Fenway, Boston, MA 02115; FAX: 617-262-4021; E-mail: slees@forsyth.org

14-18 April: Fourth French Congress on Acoustics, Marseille, France. Contact: Secretariat CFA4, 31 chemin J. Aiguier, 13402, Marseille, cedex 20, France; Fax: +33 91228248; E-mail: cfa-4@lma.cnrs-mrs.fr

21-24 April: International Conference on Acoustics, Speech, and Signal Processing ICASSP 97, Munich, Germany. Contact: H. Fastl, Lehrstuhl fur Mensch-Maschine-Kommunikation, Technische Universitat Munchen, 80290 Munchen, Germany; Fax: +49 89 2105 8535; E-mail: fas@mmk.e-tchnik.tu.muenchen.de

12-16 May: FASE Symposium on Hydroacoustics, Jurata/Gdansk, Poland. Contact: Institute of Experimental Physics, Gdansk University, Wita Stwosza 57, 80-952 Gdansk, Poland: Fax: +489 58 413175; E-mail: fizas@halina.univ.gda.pl

5-7 June: Conference on ICP and Inner Ear Pressure, Bath, UK. Contact: British Society of Audiology, 80 Brighton Rd., Reading RG6 1PS, UK; Fax: +44 1734 351915.

15-20 June: Eighth International Symposium on Nondestructive Characterization of Materials, Boulder, CO. Contact: Debbie Harris, The Johns Hopkins University, Ctr. for Nondestructive Evaluation, 102 Maryland Hall, 3400 N. Charles St., Baltimore, MD 21218, Tel.: 410-516-5397; FAX: 410-516-7249, E-mail: cnde@jhuvms.hcf.jhu.edu

15-17 June: NOISE-CON 97, State College, PA. Contact: Institute of Noise Control Engineering, P.O. Box 320, Arlington Branch, Poughkeepsie, NY 12603, Tel.: 914-891-1407; FAX: 914-463-0201.

16-20 June: 133rd Meeting of the Acoustical Society of America, State College, PA. Contact: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org

25-27 June: 12th Echocardiology Symposium and 9th Meeting of the International Cardiac Doppler Society, Rotterdam, The Netherlands. Contact: LMC Congress Service, P.O. Box 593, 3700 AN Zeist, The Netherlands, FAX: +31 343 533 357.

2-4 July: Ultrasonics International '97, Delft, The Netherlands. Contact: W. Sachse, Dept. of Theoretical and Applied Mechanics, Cornell Univ., Ithaca, NY 14853; Fax: 607 255 9179; E-mail: sachs@msc.cornell.edu

9-13 July: International Clarinet Association, Texas Tech Univ., Lubbock, TX. Contact: Keith Koons, Music Department, Univ. of Central Florida, P.O. Box 161354, Orlando, FL 23816-1354, Tel; 407-823-5116; E-mail: kkoons@pegasus.cc.ucf.edu

14-17 August: 1997 World Congress on Ultrasonics, Yokohama, Japan. Contact: S. Ueha, Precision and Intelligence Lab., Tokyo Inst. of Technology 4259 Nagatsuta, Midori-ku, Yokohama 226, Japan; Fax: +81 45 921 0898; E-mail: ucu97@pi.titech.ac.jp

20-23 August: New Zealand Acoustical Society Biennial Conference, Christchurch, New Zealand. Contact: NZ Acoustical Society, P.O. Box 1181, Auckland, New Zealand.

21-23 August: ACTIVE 97 Inter-Noise Satellite Symposium, Budapest, Hungary. Contact: ACTIVE 97 Secretariat, POAKFI, Fou 68, 1028 Budapest, Hungary; FAX: +36 1 202 0452.

14-18 avril: 4e congrès français sur l'acoustique, Marseille, France. Renseignements: Secrétariat CFA4, 31 Chemin J. Aiguier, 13402, Marseille, cedex 20, France; Fax: +33 91228248; E-mail: cfa-4@lma.cnrs-mrs.fr

21-24 avril: Conférence internationale sur l'acoustique, la parole et le traitement de signal ICASSP 97, Munich, Allemagne. Renseignements: H. Fastl, Lehrstuhl fur Mensch-Maschine-Kommunikation, Technische Universitat Munchen, 80290 Mnchen, Germany; Fax: +49 89 2105 8535; E-mail: fas@mmk.e-tchnik.tu.muenchen.de

12-16 mai: Symposium FASE sur l'hydroacoustique, Jurata/Gdansk, Pologne. Renseignements: Institute of Experimental Physics, Gdansk University, Wita Stwosza 57, 80-952 Gdansk, Poland: Fax: +489 58 413175; E-mail: fizas@halina.univ.gda.pl

5-7 juin: Conférence sur l'ICP et la pression de l'oreille interne, Bath, Royaume Uni. Renseignements: British Society of Audiology, 80 Brighton Rd., Reading RG6 1PS, UK; Fax: +44 1734 351915.

15-20 juin: Huitième symposium international sur la caractérisation non-destructive des matériaux, Boulder, CO. Renseignements: Debbie Harris, The Johns Hopkins University, Ctr. for Nondestructive Evaluation, 102 Maryland Hall, 3400 N. Charles St., Baltimore, MD 21218, Tel.: 410-516-5397; FAX: 410-516-7249, E-mail: cnde@jhuvms.hcf .jhu.edu

15-17 juin: NOISE-CON 97, State College, PA. Renseignements: Institute of Noise Control Engineering, P.O. Box 320, Arlington Branch, Poughkeepsie, NY 12603, Tel.: 914-891-1407; FAX: 914-463-0201.

16-20 juin: 133e rencontre de l'Acoustical Society of America, State College, Pennsylvanie. Renseignements: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org

25-27 juin: 12e symposium d'échocardiologie et 9e rencontre de la Société internationale du doppler cardiaque, Rotterdam, Pays Bas. Renseignements: LMC Congress Service, P.O. Box 593, 3700 AN Zeist, The Netherlands, FAX: +31 343 533 357.

2-4 juillet: Ultrasonics International '97, Delft, Pays-Bas. Renseignements: W. Sachse, Dept. of Theoretical and Applied Mechanics, Cornell Univ., Ithaca, NY 14853; Fax: 607 255 9179; E-mail: sachs@msc.cornell.edu

9-13 juillet: Association internationale de la clarinette, Texas Tech Univ., Lubbock, TX. Renseignements: Keith Koons, Music Department, Univ. of Central Florida, P.O. Box 161354, Orlando, FL 23816-1354, Tel: 407-823-5116; E-mail: kkoons@pegasus.cc.ucf.edu

14-17 août: 1997 congrès mondial sur les ultrasons, Yokohama, Japan. Renseignements: S. Ueha, Precision and Intelligence Lab., Tokyo Inst. of Technology 4259 Nagatsuta, Midori-ku, Yokohama 226, Japan; Fax: +81 45 921 0898; E-mail: ucu97@pi.titech.ac.jp

20-23 août: Conférence biennale de la Société d'acoustique de la Nouvelle-Zélande, Christchurch, Nouvelle-Zélande. Renseignements: NZ Acoustical Society, P.O. Box 1181, Auckland, New Zealand.

21-23 août: ACTIVE 97 Symposium satellite d'Inter-Noise, Budapest, Hongrie. Renseignements: ACTIVE 97 Secretariat, POAKFI, Fou 68, 1028 Budapest, Hungary; FAX: +36 1 202 0452.

25-27 August: Internoise 97, Budapest, Hungary. Contact: OPAKFI, Fo. u. 68, 1027 Budapest, Hungary; Fax: +36 1 202 0452.

7-11 September: American Academy of Otolaryngology--Head and Neck Surgery, San Francisco, CA. Contact: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314. Tel.: 703-836-4444; FAX: 703-683-5100.

22-24 September: Second Biennial Hearing Aid Research and Development Conference, Bethesda, MD. Contact: National Institute of Deafness and Other Communication Disorders, 301-970-3844; FAX: 301-907-9666; E-mail: hearingaid@tascon.com

22-25 September: 5th European Conference on Speech Communication and Technology, Patras, Greece. Contact: G. Kokkinakis, Department of Electrical and Computer Engineering, University of Patras, 26110 Rion-Patras, Greece; Fax: +30 61 991 855, E-mail: gkokkin @wcl.ee.upatras.gr

7-10 October: 1997 IEEE Ultrasonics Symposium, Toronto, Canada. Contact: S. Foster, Department of Medical Biophysics, Sunnybrook Health Science Ctr., 2075 Bayview Avenue, Toronto, Ontario M4N 3M5, Canada; E-mail: stuart@owl.sunnybrook.utoronto.ca

8-10 October: 1997 Acoustics Week in Canada, Windsor, Canada. Contact: Dr. R. Ramakrishnan, Vibron Ltd, 1720 Meyerside Drive, Mississauga, Ontario, L5T 1A3. Tel.: (905) 670-4922; FAX: (905) 670-1698.

1-5 December: 134th Meeting of the Acoustical Society of America, San Diego, CA. Contact: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org

1998

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

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

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

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

25-27 août: Internoise 97, Budapest, Hongrie. Renseignements: OPAKFI, Fo. u. 68, 1027 Budapest, Hungary; Fax: +36 1 202 0452.

7-11 septembre: Académie américaine d'otolaryngologie - Chirurgie de la tête et du cou, San Francisco, CA. Renseignements: American Academy of Otolaryngology--Head and Neck Surgery, One Prince St., Alexandria, VA 22314; Tel.: 703-836-4444; FAX: 703-683-5100.

22-24 septembre: 2e conférence biennale sur la recherche et le développement des protèses auditives, Bethesda, MD. Renseignements: National Institute of Deafness and Other Communication Disorders, 301-970-3844; FAX: 301-907-9666; E-mail: hearingaid@tascon.com

22-25 septembre: 5e conférence européenne de la communication et la technologie de la parole, Patras, Grèce. Renseignements: G. Kokkinakis, Department of Electrical and Computer Engineering, University of Patras, 26110 Rion-Patras, Greece; Fax: +30 61 991 855, E-mail: gkokkin @wcl.ee.upatras.gr

7-10 octobre: Symposium de 1997 de l'IEEE sur les ultrasons, Toronto, Canada Renseignements: S. Foster, Department of Medical Biophysics, Sunnybrook Health Science Ctr., 2075 Bayview Avenue, Toronto, Ontario M4N 3M5, Canada; E-mail: stuart@owl.sunnybrook.utoronto.ca

8-10 octobre: Semaine canadienne d'acoustique 1997, Windsor, Canada. Renseignements: Dr. R. Ramakrishnan, Vibron Ltd, 1720 Meyerside Drive, Mississauga, Ontario, L5T 1A3. Tel.: (905) 670-4922; Fax: (905) 670-1698.

1-5 décembre: 134e rencontre de l'Acoustical Society of America, San Diego, Californie. Renseignements: Acoustical Society of America, 500 Sunnyside Blvd., Woodbury, NY 11797, Tel.: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org; WWW: http//asa.aip.org

1998

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

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

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

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

MORE NEWS...

The Asian acoustical community under the leadership of the Hong Kong Institute of Acoustics and the Hong Kong Polytechnic University will be hosting WESTPRAC VI in November 1997. WESTPRAC VI will be a 3-day technical acoustics conference involving all of the Asian acoustical societies: Hong Kong, China, Korea, Japan, Singapore, Australia etc. This is the sixth meeting of these societies. About 200 delegates are expected, and about 100 technical papers will be presented in areas such as transportation noise, environmental noise, noise emission and physical acoustics. An exhibition for acoustical software and hardware manufactures is also planned. Contact: Dr S K Tang, Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hom, Kowloon, Hong Kong. Tel.: (852) 2766 5855; Fax: (852) 2774 6146; E-mail: besktang@polyu.edu.hk.

CAA on the Web! The Canadian Acoustical Association now has a home page on the World Wide Web. This can be located at: "http://www.uwo.ca/hhcru/caa/". Members of CAA are invited to visit the web page and to provide comments and suggestions for the development of our page. Particularly welcome are suggestions for links to the web sites of Canadian laboratories involved in acoustics research and to key sources of acoustics information throughout the world.

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

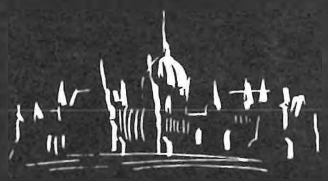
AUTRES NOUVELLES...

La communauté acoustique asiatique, sous la direction de l'Institut d'acoustique de Hong Kong et l'université Polytechnique de Hong Kong, sera l'hôte de WESTPRAC VI en novembre 1997. WESTPRAC VI sera une conférence acoustique de 3 jours à laquelle participera toutes les sociétés acoustiques asiatiques: : Hong Kong, Chine, Corée, Japon, Singapour, Australie etc. Ce sera la sixième réunion de ces sociétés. Environ 200 délégués sont attendus, and environ 100 présentations techniques seront données sur des sujets tels: bruit des transports, bruit environnemental, émission de bruit et acoustique physique. Une exhibition des manufacturiers de logiciels et de matériel informatique est aussi prévue. Renseignements: Dr S K Tang, Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hom, Kowloon, Hong Kong. Tel.: (852) 2766 5855; Fax: (852) 2774 6146; E-mail: besktang@polyu.edu.hk.

L'ACA sur le Web! L'Association canadienne d'acoustique a maintenant une page sur le World Wide Web. Elle est située au "http://www.uwo.ca/hhcru/caa/". Les membres de l'ACA sont invités à visiter notre page et à nous envoyer leurs commentaires et suggestions. Nous apprécierions particulièrement des suggestions de liens avec d'autres centres canadiens impliqués dans la recherche acoustique et avec des sources d'information sur l'acoustique au travers du monde.

Si vous avez des nouvelles à nous communiquer, n'hésitez pas à nous les envoyer par courrier ou fax (coordonnées incluses à l'envers de la page couverture), ou par courrier électronique à desharnais@drea.dnd.ca

HELP QUIET THE WORLD
FOR A HIGHER QUALITY LIFE



inter·noise 97

THE 1997 INTERNATIONAL CONFERENCE ON NOISE CONTROL ENGINEERING

BUDAPEST-HUNGARY

1997 AUGUST 25-27

FIRST ANNOUNCEMENT

INTER-NOISE 97, the 1997 International Congress on Noise Control Engineering, will be held at the Technical University of Budapest, in the capital of Hungary from 1997 August 25 to 27. The Congress is sponsored by the International Institute of Noise Control Engineering, and is being organized by the Acoustical Commission of the Hungarian Academy of Sciences and the Hungarian Scientific Society for Optics, Acoustics, Motion Pictures and Theatre Technology.

INTER-NOISE 97 will be the twenty-sixth in a series of international congresses on noise control engineering that have been held all over the world since 1972. The theme of INTER-NOISE 97 is: HELP QUIET THE WORLD FOR A HIGHER QUALITY LIFE.
Technical papers in all areas of noise control engineering will be considered for presentation at the congress and for publication in the Congress Proceedings.

An Announcement and Call for Papers will be issued shortly; copies will be available from the Conference Secretariat at the address given below.

A major acoustical equipment, materials and instrument exhibition will be held in conjunction with INTER-NOISE 97. The exhibition will include materials and devices for noise control as well as instruments such as sound level meters, acoustical signal processing systems, and equipment for active noise control.

Programs for "accompanying persons" and social activities for all delegates will be organized.

Further information on the Congress and the Exhibition may be obtained from the INTER-NOISE Conference Secretariat

RETURN COUPON

Please return this coupon if you are interested in being added to the mailing list for INTER-NOISE 97.

☐ I am interested in attending INTER-NOISE 97

☐ I am interested in presenting a technical paper

☐ My company may be interested in participating in equipment exhibition

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THE SCIENTIFIC SOCIETY FOR OPTICS, ACOUSTICS, MOTION PICTURES AND THEATRE TECHNOLOGY (OPAKFI)

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Hungary
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The Canadian Acoustical Association
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ANNONCE DE PRIX

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

Prix Post-Doctoral Edgar et Millicent Shaw en Acoustique

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

1990	*Li Cheng*	*Université de Sherbrooke*
1993	*Roland Woodcock*	*University of British Columbia*
1994	*John Osier*	*Defense Research Establishment Atlantic*
1995	*Jing-Fang Li*	*University of British Columbia*

Prix Étudiant Alexander Graham Bell en Communication Verbale et Acoustique Comportementale

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

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

Prix Étudiant Fessenden en Acoustique Sous-marine

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

1992	*Daniela Dilorio*	*University of Victoria*
1993	*Douglas J. Wilson*	*Memorial University*
1994	*Craig L. McNeil*	*University of Victoria*

Prix Étudiant Eckel en Controle du Bruit

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

1994	*Todd Busch*	*University of British Columbia*
1995	*Raymond Panneton*	*Université de Sherbrooke*

Prix des Directeurs

Trois prix sont décernés, à tous les ans, aux auteurs des trois meilleurs articles publiés dans l'*Acoustique Canadienne*. Tout manuscrit rapportant des résultats originaux ou faisant le point sur l'état des connaissances dans un domaine particulier sont éligibles; les notes techniques ne le sont pas. Le premier prix, de $500, est décerné à un(e) étudiant(e) gradué(e). Le deuxième et le troisième prix, de $250 chacun, sont décernés à des auteurs professionnels âgés de moins de 30 ans et de 30 ans et plus, respectivement. Coordonnateur: Blaise Gosselin, Hydro Québec, 16^{e} étage, 75 boul. René Lévesque ouest, Montréal, QC H2Z 1A4.

Prix de Presentation Étudiant

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

The Canadian Acoustical Association
l'Association Canadienne d'Acoustique

PRIZE ANNOUNCEMENT

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

Edgar and Millicent Shaw Postdoctoral Prize in Acoustics

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

1990	*Li Cheng*	*Université de Sherbrooke*
1993	*Roland Woodcock*	*University of British Columbia*
1994	*John Osler*	*Defense Research Establishment Atlantic*
1995	*Jing-Fang Li*	*University of British Columbia*

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

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

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

Fessenden Student Prize in Underwater Acoustics

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

1992	*Daniela DiIorio*	*University of Victoria*
1993	*Douglas J. Wilson*	*Memorial University*
1994	*Craig L. McNeil*	*University of Victoria*

Eckel Student Prize in Noise Control

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

1994	*Todd Busch*	*University of British Columbia*
1995	*Raymond Panneton*	*Université de Sherbrooke*

Directors' Awards

Three awards are made annually to the authors of the best papers published in *Canadian Acoustics*. All papers reporting new results as well as review and tutorial papers are eligible; technical notes are not. The first award, for $500, is made to a graduate student author. The second and third awards, each for $250, are made to professional authors under 30 years of age and 30 years of age or older, respectively. Coordinator: Blaise Gosselin, Hydro Québec, 16^{e} étage, 75 boul. René Lévesque ouest, Montréal, QC H2Z 1A4.

Student Presentation Awards

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

INSTRUCTIONS TO AUTHORS FOR THE PREPARATION OF MANUSCRIPTS

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

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

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

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

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

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

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

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

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

Photographs: Submit original glossy, black and white photograph.

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

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

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

DIRECTIVES A L'INTENTION DES AUTEURS PREPARATION DES MANUSCRITS

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

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

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

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

Sommaire: En versions anglaise et française. Titre en 12 pt, lettres majuscules, caractères gras, centré. Paragraphe 0.5" en alinéa de la marge, des 2 cotés.

Titres des sections: Tous en caractères gras, 12 pt, Times-Roman. Premiers titres: numéroter 1, 2, 3, ..., en lettres majuscules; sous-titres: numéroter 1.1, 1.2, 1.3, ..., en lettres majuscules et minuscules; sous-sous-titres: ne pas numéroter, en lettres majuscules et minuscules et soulignés.

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

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