Acoustics Australia

- Snodar: an acoustic radar for turbulence
- Immersion index for acoustic spaces
- A simple underwater sound logger
- Noise control using feedback

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PAPERS

Cover: Xuefei Gong with the Snodar at Dome A. *Cover design: Heidi Hereth*

Message from the President

I have just come back from a presentation about how "social media" are impacting our lives. In particular, Twitter is the latest media madness but the question is really how do you communicate with others who have a like interest? I have noticed that a few organisations including some professional ones are moving with the times and have started a Facebook group which one can join. I don't know if that is where we as a Society want to go as of yet but it certainly needs to be on the agenda and we certainly want to keep up with the times. Indeed, it is for that reason that we have made past issues of AA available on line to members and non-members can access past issues by paying a fee. This helps disseminate our collective knowledge. We certainly have some world class papers in our journal. I invite you all to continue contributing papers in your fields of interest.

And talking about papers, I will take this opportunity of reminding you that this year Annual Acoustics Conference will be held in Adelaide on November 23 – 25th 2009. It is promising to be another exciting event in our calendar and the associated exhibition is already sold out! I look forward to seeing you all there soon.

From what I hear, acoustic consultants are generally quite busy right now and have not been badly hit by the Global Financial Crisis. That is certainly good news. However, when it comes to training and education in acoustics, it seems though, that the number of course options is declining as universities face cutbacks in staff etc and the onus is now falling on training in house. One bit of good news is that ADFA is offering a Distance Learning Program for Professional Education in Acoustics. This is a series of short courses and is based on a similar program that has been offered via Universities and the UK Institute of Acoustics. A certificate from the UNSW will be issued for successful completion of each module.

And talking of the GFC, Standards Australia have advised that as a result of the GFC, some Project Manager positions have been made redundant and they do not have the resources available to continue their previous levels of support for Standards Development activities that are not funded or resourced by stakeholders. This means that as of 1 July 2009, all new projects irrespective of the chosen pathway, will require stakeholder funding. The implication is potentially considerable for us as quite a number of Acoustics Standards were either under active consideration or were being provisionally considered for update. These may now be held up for years.

Work is also proceeding in organising the ICA in Sydney for next year (23 – 27 August 2010). This will be held at the Sydney Convention Centre in Darling Harbour and will be followed by the International Symposium on Room Acoustics to be held at the Arts Centre in Melbourne (29 – 31 August, 2010). Mark these in your calendar and start thinking about papers for submission. These are both a great opportunity to showcase Australian acoustics and I look forward to all members contributing in some way. AAS members will get a special rate for attendance. Cheers for now,

Norm Broner

From the Editors

Once again I am astounded by the range of topics covered by acoustics research in Australia. In this issue the papers range from the acoustics of personal to public spaces, and from the local to the most distant terrestrial environments.

One paper deals with a practical feedback controller to reduce low frequency noise in small rooms such as bedrooms and offices. Such a device might become a standard feature in future commercial and even domestic architecture.

Another paper proposes a simple parameter, the acoustic immersion index, designed to describe the extent to which a listener in a hall feels immersed in a sound field, rather than hearing sound from the general direction of the source. A third paper describes how a modified MP3 player can be coupled with a hydrophone to produce a low-cost system to record ambient underwater noise.

The final paper describes the Snodar installation at Dome A, located at the highest point on the Antarctic plateau. These are two high resolution acoustic radars that astronomers are using to measure atmospheric turbulence. Dome A is an extremely remote place with very difficult access; indeed no human has yet witnessed a sunrise or sunset at this location. It is particularly attractive to astronomers because of the excellent 'seeing' that result from its thin, dry, clean and very still air. However, designing instruments that must operate unattended for most of the year in temperatures lower than –80ºC while providing their own heat and power must certainly pose challenges, particularly as the apparatus will be towed by tractors for over one thousand kilometres.

The spectrum of topics is broadened if we add the topics of the previous issue. One paper compared the vocal tract shape determined by acoustic or electromagnetic waves. Another calculated the acoustic radiation of submarines, and another presented an acoustic study of saxophones.

The August issue also had three interesting Technical Notes, on wind farm noise, on audiometric testing, and a case study of the noise and vibration design for an electron microscopy building.

Sadly, we have no Technical Notes for you this issue. We are sure that our readership could provide an equal breadth of interest in technical reports and hope to see some in time for the December issue.

John Smith

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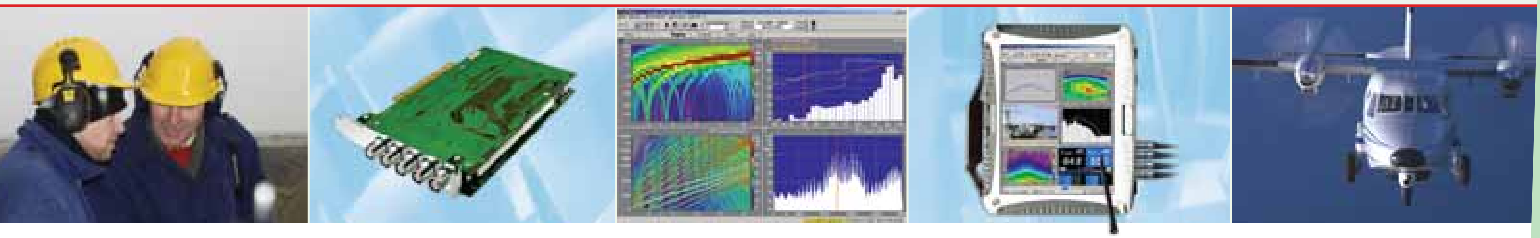

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SNODAR: AN ACOUSTIC RADAR FOR ATMOSPHERIC TURBULENCE PROFILING WITH 1m RESOLUTION

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ABSTRACT: Snodar is a 5 kHz monostatic acoustic radar designed to measure the atmospheric turbulence within the first 200 m of the atmosphere with a vertical resolution of 1 m. An in situ calibration target is used to give absolute intensity calibration. The primary motivation for building Snodar is to quantify the site conditions for a future astronomical observatory in Antarctica. Two Snodar instruments are operating at Dome A, Antarctica, during 2009 as part of the completely robotic "PLATO" facility. The instruments are separated by 20 m and sample from 8 m to 200 m with a resolution of 1 m allowing the spatial and temporal characteristics of the atmospheric boundary layer to be investigated. We present here the acoustic design of Snodar and example data demonstrating the performance of the instrument.

1. INTRODUCTION

Optical observations of astronomical objects made from ground-based observatories are hampered by the Earth's atmosphere. This is primarily due to turbulent mixing within the atmosphere causing temperature fluctuations, and hence variations in the refractive index, resulting in "optical turbulence". This causes stars to no longer appear as point sources, a phenomenon that astronomers refer to as *seeing*, and limits the resolving power of large optical telescopes. The characteristics of the optical turbulence at a given site is of major importance in deciding whether to build an observatory there.

Optical turbulence is generally confined to the lowest 20 km of the Earth's atmosphere, with a dramatic increase within the lowest 1 km due to the direct interaction of the atmosphere with the Earth's surface. The lower layer is called the atmospheric boundary layer, and its height is dependent on the local topography, the surface roughness, and the surface energy budget, which in turn depends on the position of the sun, cloud coverage, surface type, and other factors. The portion of the atmosphere above the atmospheric boundary layer is called the free atmosphere.

Harper [1] and Gillingham [2] made early predictions that Antarctica could be a favorable location from which to make astronomical observations. Dome C, at a height of 3233 m on the Antarctic plateau as shown in Figure 1, has been shown to have exceptional free-air seeing [3]. Subsequent experiments at Dome C have shown that $\sim$90% of the total optical turbulence is confined within a shallow atmospheric boundary layer with a typical height of 30 m [4]. This result is very promising for astronomers as it is technically feasible to place a 2 m class optical telescope on a 30 m tower [5][6]. Such a telescope could have a comparable resolving power to the Hubble Space Telescope for a useful percentage of the time. The plateau observatory, PLATO, has been designed to assess the potential of Dome A, the highest location on the Antarctic plateau, as an astronomical observatory [7]. Designed at

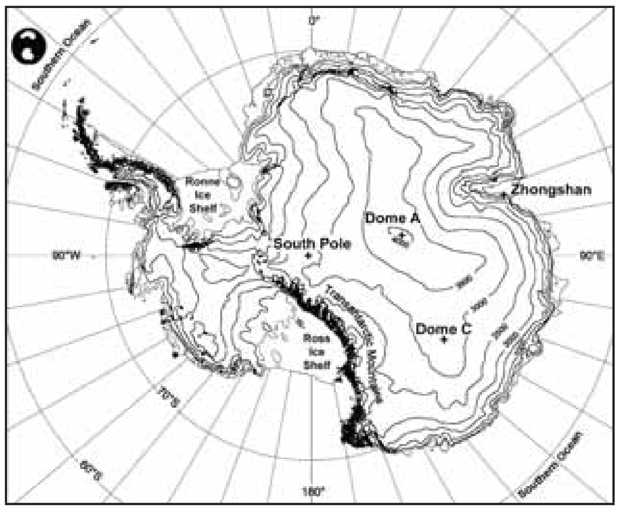

Figure 1. Map of Antarctica showing the South Pole, Dome C and Dome A. Basic map courtesy of the Australian Antarctic Data Center.

the University of New South Wales, PLATO was deployed to Dome A in January 2008 by the National Astronomical Observatories of the Chinese Academy of Sciences and the Polar Research Institute of China. One of the key science goals for PLATO is to characterize the height and variability of the atmospheric boundary layer.

Several techniques have been developed to remotely measure optical turbulence, e.g. [8][9]. Of these, acoustic radars (or SODARs) have the advantage of relatively high spatial and temporal resolution. However, acoustic radars are non-trivial to calibrate and are normally used to profile 3-dimensional wind speed within the first 2 km of the atmosphere. Typical commercial SODARs also only have a vertical resolution of approximately 10 m.

No commercial instruments are available to profile the first 100 m of the atmosphere to a resolution of 1 m or better, which is what is needed on the Antarctic plateau. For this reason we developed Snodar. Snodar is a robust, autonomous, high-resolution turbulence profiling acoustic radar that can operate down to temperatures of −80°C. We describe here the theoretical background, acoustic design and performance of Snodar. The electrical design of Snodar has been published by Bonner *et al.* [10][11] while initial results from PLATO have been published by Yang *et al.* [12], and Ashley *et al.* [13].

2. THEORETICAL BACKGROUND

In 1941 Andrey Kolmogorov developed theories [14] that allowed the chaotic phenomenon of turbulence to be described statistically for very high Reynolds numbers. Simply put, the Kolmogorov theory of turbulence states that energy enters a turbulent field at some outer length scale L_0 and is redistributed into smaller and smaller scales by eddy action until some inner scale l_0 is reached. Energy enters the turbulent field at L_0 due to variations in the average wind velocity and leaves the field at l_0 as heat due to molecular viscosity. Eddies with length scales of L where $l_o \ll L \ll L_o$ are said to be in the inertial range. Turbulence within the inertial range is ideally homogenous and isotropic. Tatarskii [15] shows that structure functions can be used to describe turbulent fields. The structure function for turbulence within the inertial range of fully developed turbulence obeys the $\frac{2}{3}$ law from Kolmogorov and Obukhov [15] and is defined as

$$D(\mathbf{r}) = C^2 \varepsilon^{\frac{2}{3}} \mathbf{r}^{\frac{2}{3}} \tag{1}$$

where ε is the energy dissipation rate and C is the structure function constant for the parameter of interest. A structure function constant can be physically interpreted as the mean squared difference of some parameter between points throughout the turbulent field and describes the intensity of the turbulence. The refractive index structure function constant C_N^2 is of primary interest to astronomers as astronomical seeing is proportional to the three-fifths power of the integral of C_N^2 along the line of sight [15]. Tatarskii also shows that the temperature structure function constant C_T^2 can be related to the optical refractive index structure function constant C_N^2.

Temperature and velocity inhomogeneities caused by turbulence within a medium on the scale of $\lambda/2$ are responsible for scattering acoustic waves with a wavelength λ. The acoustic scattering cross-section σ of a turbulent volume is a function of the temperature and velocity structure function constants, C_T^2 and C_V^2 respectively. The scattering cross-section is the scattered power per unit area per incident power per unit volume having dimensions $m^2 m^{-3}$ and is given by [15]

$$\sigma = 0.03 \left(\frac{\omega}{c(h)}\right)^{\frac{1}{3}} cos^2\theta \left[\frac{C_V^2}{c(h)^2} cos^2\frac{\theta}{2} + 0.13\frac{C_T^2}{T(h)^2}\right] \left(sin\frac{\theta}{2}\right)^{\frac{-11}{3}} \tag{2}$$

where ω is the frequency of the acoustic wave in radians per second, θ is the scattering angle relative to the original wave vector, T is the average temperature of the scattering volume and $c(h)$ is the speed of sound at height h. It can been seen by examining Equation 2 that if $\theta = \pi$ then σ is only dependent on C_T^2 i.e. if a monostatic system—one with a collocated transmitter and receiver—is used, then the back scatter is solely a function of the temperature structure function constant.

Little [16] first proposed using acoustic radars, or SODARs, to probe the structure of the Earth's lower atmosphere in 1969. Little demonstrated that the acoustic power received at an antenna is related to the scattering cross-section by the SODAR equation

$$P_R = P_T \eta A \sigma \frac{c(h)\tau}{2} \frac{e^{-2\alpha h}}{h^2} \tag{3}$$

where h is the height of the turbulent volume with a scattering cross-section of σ, τ is the pulse duration, η is the system gain, α is the atmospheric attenuation constant and P_T and P_R are the power transmitted and the power received by the antenna's effective area A. Therefore the contribution to astronomical seeing can be calculated from the scattered acoustic energy received by a monostatic acoustic radar after correcting for atmospheric attenuation and the system gain. The process of determining the system gain, or calibrating the instrument, is non-trivial and will be discussed in Section 4. It is worth noting that P_R/P_T is typically on the order of 10^{-15}.

Snodar is an acoustic radar and works by sending an intense acoustic pulse into the atmosphere and recording the faint backscatter off the atmospheric turbulence. The vertical resolution $\triangle h$ of an acoustic radar is

$$\triangle h = \frac{c(h)\tau}{2} \tag{4}$$

The height h of the scattering volume is a function of the time of flight t and the speed of sound along the acoustic path and is given by solving

$$h(t) = \frac{1}{2}\int_0^t c(h)dt \tag{5}$$

The minimum sampling height of an acoustic radar is limited by transducer ringing, antenna reverberation or echoes from fixed objects, also called ground clutter. The minimum sampling height for commercial units is typically 20∼40 m. Such commercial units are not useful when investigating a shallow atmospheric boundary layer ≤30 m in height.

3. ACOUSTIC DESIGN

Snodar was designed to investigate the atmospheric boundary layer on the high Antarctic plateau throughout the polar year. To accomplish this, Snodar must 1) have a vertical resolution of 1 m or better, 2) have carefully controlled transducer ringing and antenna reverberation to allow sampling within 10 m of the ground, 3) operate fully autonomously, and 4) have a well-determined absolute intensity calibration throughout the polar year. The requirement of maintaining intensity calibration posed a considerable design challenge due to wide ambient temperature range of $-80°C$ to $-30°C$ and the requirements of completely robotic operation. Calibration of the instrument will be discussed in Section 4. The acoustic design of the instrument will be discussed in this Section.

The nominal operating frequency of an acoustic radar is the prime design decision as it determines the scale of the antenna and possible transducers. The optimal operating frequency can be found by looking for the maximum in Equation

3 with respect to ω after substituting for σ from Equation 2. Interestingly, acoustic energy propagates exceedingly well in Antarctica due to the extreme cold. For example, the attenuation at 5 kHz at normal room temperature with 50% relative humidity is about 40 dB/km, whereas at $-60°$C the attenuation is only 4 db/km [17]. The attenuation increases to about 16 dB/km at Dome A due to the reduced atmospheric pressure at this elevation (4091 m). The optimal operating frequency for Snodar was found to be 5 kHz. A transmitted pulse duration of 33 cycles at 5 kHz provides a vertical resolution of 1 m.

Snodar uses a single transducer mounted near the focal point of a parabolic reflector to transmit an intense acoustic pulse and receive the faint backscatter. This makes Snodar a true monostatic acoustic radar. A $0.9m\ f/0.6$ commercial parabolic satellite dish is used to collimate the acoustic energy generated by the transducer. The transducer we selected is a JBL2402H horn-loaded compression driver manufactured by JBL. The JBL2402H is rated for a continuous power of 40 W over a frequency range from 3 kHz to 15 kHz. The beam pattern generated by the antenna at 5 kHz is essentially diffraction limited with a beam width of $\approx 4°$. The reflector and transducer are housed in a 1.6 m tall 12-sided sound cone to reduce side lobes and acoustic noise. A cross-sectional view of the sound cone, reflector and transducer is shown in Figure 2. The sound cone consists of 12 separate panels and is flat-packable for easy transportation. The walls of the sound cone are tapered away from the acoustic beam at an angle of $8°$. The top surface of the parabolic reflector was aluminized to reduce its thermal emissivity by approximately an order of magnitude. This allowed higher temperatures to be reached when heating the dish from electrical heat pads glued to its back surface. Heating was used to remove any accumulated snow. Focusing of sunlight by the aluminized reflector is not an issue while operating Snodar in polar regions as the sound cone ensures that the dish is always in shadow.

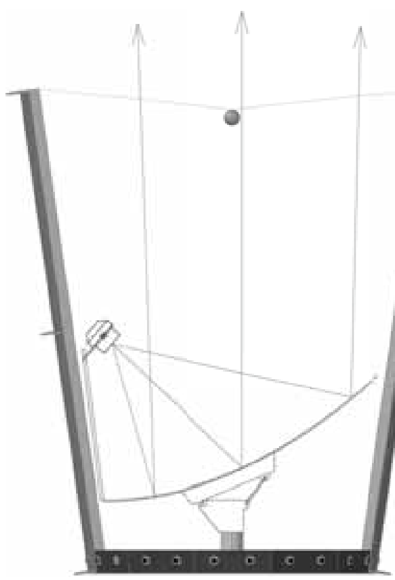

Figure 2. Cross-section of Snodar's 1.6 m tall antenna. The offset parabolic reflector and transducer are supported by a post at the bottom of the sound cone. The inside of the sound cone is lined with 50 mm of glasswool sound absorber. The calibration sphere, which is discussed in Section 4, is shown suspended across the antenna's aperture.

The inside of the sound cone is lined with 50 mm thick unfaced resin-bonded glasswool sound absorber. The sound absorber used is Quietel$^{\mathrm{TM}}$ manufactured by CSR Bradford Insulation. A glasswool sound absorber was selected as fridge tests indicated that these materials maintain their acoustic properties down to $-80°C$ unlike many polymer-based sound absorbers which lose their acoustic properties and mechanical strength at sub-zero temperatures. Glasswool sound absorbers are also more suitable for long-term installation in direct sunlight.

The electronics that control Snodar are located inside the PLATO instrument module, which is kept above $-5°C$. A low-noise solid state switch and impedance-matching network are mounted close to the transducer in the sound cone to reduce the system's susceptibility to electromagnetic interference. The impedance-matching network is an important part of the system as it dramatically reduces transducer ringing. A WiFi based webcam has also been installed within the sound cone to visually monitor snow and ice accumulation.

4. INTENSITY CALIBRATION

The problem of calibrating the absolute intensity of SODARs is nontrivial and has been given considerable attention in recent decades [18]. For Snodar, we have the additional problems of coping with the wide operating temperature range of between $-30°C$ and $-80°C$, and the possible accumulation of snow and ice within the antenna. The variability of the speed of sound in air due to changes in the ambient air temperature can be corrected for during post-processing with data from automatic weather stations.

To maintain an absolute intensity calibration Snodar uses a calibration sphere made from solid phenolic resin (a billiard ball) suspended in the middle of the sound cone aperture, as shown in Figure 2. The calibration sphere provides a fixed echo with a known scattering cross-section at a known height. The positioning of the calibration sphere allows the gain of the system to be monitored and corrected for. A small amount of frost may form on the sphere, however the size and shape of the sphere is not expected to substantially change. The gain of the system however is expected to vary with transducer life, ambient temperature and accumulation of snow and ice on the parabolic reflector.

The scattering cross-section of the calibration sphere can be determined analytically, numerically or experimentally. Dragonette *et al.* [19] gives the acoustic scattering cross-section of spheres in the far field. Similar techniques could be applied to find an analytical solution for the scattering cross-section of Snodar's calibration sphere; however this is much more complicated as the sphere is in the near field of the antenna. Instead, Snodar is initially calibrated against tower-mounted microthermal sensors which measure C_T^2 directly; the scattering cross-section of the sphere is then found by comparing the echo from the sphere with the echo from turbulence with known C_T^2. The sphere remains with Snodar allowing the instrument to be recalibrated without tower mounted instrumentation.

As the acoustic scattering from the calibration sphere is dependent on the acoustic wavelength which is dependent on ambient temperature, it is essential to calibrate Snodar at a range of wavelengths. A look-up-table of calibration constants as a function of wavelengths is generated during calibration allowing Snodar to be recalibrated at various ambient temperatures. Snodar currently recalibrates itself every 30 minutes at Dome A.

5. PERFORMANCE

Snodar was deployed to Dome A in 2008 as part of the PLATO facility [7]. During this first year we only obtained one week of data due to failure of the transducer diaphragm. However, these early results were promising since they clearly showed the expected diurnal cycle of the atmospheric boundary layer with an excellent signal-to-noise-ratio of up to 60dB. The 2008 instrument was repaired in 2009 and a second instrument was deployed. The two instruments are separated by 20 m and sample from 8 m to 200 m with a 1 m vertical resolution allowing the temporal and spatial characteristics of the atmospheric boundary layer to be investigated. Data are transmitted to Sydney Australia through the PLATO facility via the Iridium satellite network every 6 hours and stored in a MySQL database. The height of the atmospheric boundary layer is automatically extracted from the data and stored for statistical analysis.

The performance of the instrument can be seen with an echo from the 2008 data set. Figure 3 shows the received power as a function of scattering height for a single 5 kHz echo with a vertical resolution of 1 m; the height of the scattering volume is given by Equation 5. The raw signal was filtered to a bandwidth of 500 Hz centered at 5 kHz before the power was calculated with a moving average of the signal squared. The in-band transducer ringing/antenna reverberation is indicated by the straight line at approximately 5 m. The ringing/reverberation reduces to the noise floor at approximately 8 m. The signal power clearly deviates from $1/r^2$ at 40 m. Making the reasonable assumption that T is approximately constant around 40 m and that there is not a sudden increase in atmospheric attenuation above 40 m, Equation 3 indicates that there is a sharp drop in σ and therefore C_T^2, at a height of 40 m. This sudden drop in C_T^2 indicates the height of the atmospheric boundary layer. The power spectrum as a function of time (expressed as height via Equation 5) for the echo shown in Figure 3 is shown in Figure 4. The dark band within the first 5 m is due to spectral leakage from the intense initial pulse; the input signal was also being clipped within ± 1.4V for this period. Second and third order harmonics are also visible. The transducer's ringing is visible at its natural frequencies of 2.0 kHz, 2.5 kHz and 7.8 kHz. The natural frequencies of the transducer vary slightly with temperature but always remain outside the passband of the 4750 Hz to 5250 Hz digital filter. A Doppler shift is imparted on scattered waves if the velocity of the scattering surface or volume has a non-zero

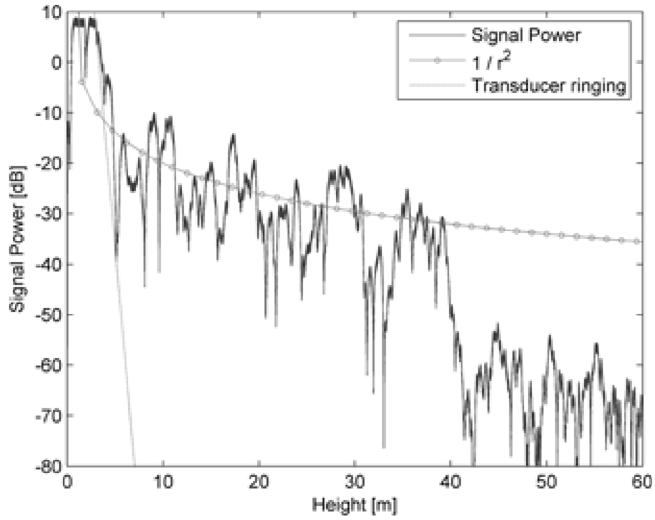

Figure 3. The first 60 m of a single 5 kHz echo from Snodar while operating at Dome A, Antarctica with a vertical resolution of 1 m. The raw signal was filtered to a bandwidth of 500 Hz centered at 5 kHz. The in-band transducer ringing/antenna reverberation is indicated by the straight line at approximately 5 m. $1/r^2$ is shown for comparison with Equation 3.

velocity component along the line of sight. This can sometimes be used to distinguish echoes from fixed objects and scattering from atmospheric turbulence, as the atmosphere is seldom stationary. This method, however, does not always allow antenna reverberation and scattering from turbulence to be separated for vertically pointed monostatic acoustic radars on the ground. This is because there is not always a vertical wind velocity several meters above the ground or within the antenna itself. Instead, Snodar uses linear frequency chirps and impulses of the form given in Equations 6 and 7 respectively where A_0 is the signal amplitude, ω is the chirp start frequency, α is the frequency sweep rate, δ is the Dirac-delta function. Linear frequency chirps and impulses are not efficiently scattered by turbulence which allows fixed echoes and antenna reverberation to be identified. This is only possible as Snodar has a wide operating range from 3 kHz to 15 kHz.

$$f(t) = A_0 sin((\omega + \alpha t)t) \qquad f(t) = A_0 \delta(t) \qquad (6,7)$$

6. CONCLUSION

We have presented here the acoustic design of a new instrument called Snodar designed specifically to profile atmospheric turbulence within the lowest 200 m of the Earth's atmosphere to a resolution of 1 m on the Antarctic plateau. The performance of the instrument has been demonstrated with actual data from Dome A, Antarctica obtained during 2008.

It is expected that the system gain will reduce with the accumulation of snow and ice within the antenna structure. The full impact of this will be assessed by comparing the system gain as determined by the in situ calibration sphere and the webcam images from within the sound cone throughout the polar year.

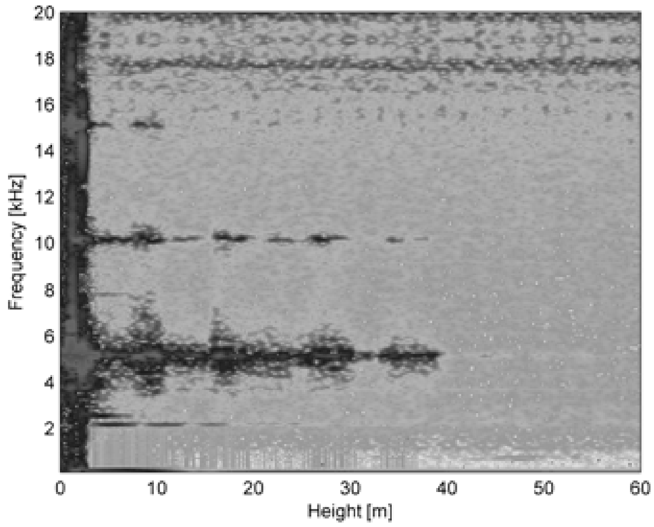

Figure 4. The power spectrum as a function of time (expressed as height via Equation 5) for the echo shown in Figure 3.

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A SIMPLE ACOUSTIC IMMERSION INDEX FOR MUSIC PERFORMANCE SPACES

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A simple acoustic immersion index is proposed that compares the reverberant sound pressure level with the prompt sound pressure level, including both direct and once-reflected contributions, for organ music in a hall defined only by its geometrical dimensions and reverberation time. Two versions of the index are considered. In the first version a very simple calculation is proposed that gives a constant index value S_1 throughout the hall. The second version of the index separates direct and once reflected sound, with the result that the computed index S_2 varies from the front to the rear of the hall. Values of the index S_1 are given for several well known halls and are all close to 0 dB. The exact values appear to correlate well with subjective observations on the acoustical character of the halls.

INTRODUCTION

During the course of a 1998 ASA conference session and subsequent discussion [1] it appeared to me that, while organ builders and architects alike understand and appreciate the requirement for a balance between the clarity of an organ or an orchestra and the fullness contributed by reverberation, they do not have readily to hand any simple index by which this balance might be expressed, and this situation does not seem to have changed much in the ensuing years. For existing halls, of course, they are able to present curves showing direct sound, prompt reflected sound, and reverberation, and there certainly exist means by which such curves can also be calculated at the detailed design stage, or measured in a model. Such studies are valuable, and indeed essential, but they require much detailed design work and subsequent analysis.

It seems that it would also be useful to have a simple index that gives a moderately reliable measure of the ratio of reverberant to "prompt" sound (that which arrives within about 60 ms) for a given hall and that can be worked out with very little labour or detailed design information. The 60 dB reverberation time T_{60} is indeed one such index that is commonly used, but it tells only part of the story. A brief search of standard works dealing with architectural acoustics [2-6] shows that the subject is well understood, as might have been expected, but no index parameter of simplicity comparable to the reverberation time appears to have been suggested. It is the purpose of this paper to propose such an index, which I will call an immersion index. In essence it measures the degree to which the listener feels immersed in the sound field, rather than perceiving it as coming from the general direction of the instrument.

Qualitatively, this immersion index is the inverse of various types of "clarity index" that have been proposed, which generally measure the ratio of directly propagated sound to reverberant sound at various positions in the hall [4]. This concept of clarity will be discussed again in a later section. For the present we simply note that the first order immersion index discussed below has the great advantage of being a single number that is very simply calculable.

Because the sessions that provoked this response dealt with organ sound, and because this is a case that is relatively simple to analyse, it will taken as the basis for discussion. It is hoped, however, that a modified version of the index, or even the same index, might prove useful in preliminary assessment of performance spaces when used for other instruments or for choirs.

Organ sound is particularly suited to this discussion for several reasons. The first is that it is sustained, rather than percussive; the second is that the sound source, considered globally, is spatially distributed; and the third is that organ sound can be readily modified by adding or subtracting ranks of higher pitch. The perceptual attribute of immersion in organ music is also easier to define than in many other cases because of the nature of the music itself: sharp percussive transients are absent, and we deal instead with contrapuntal passages or with massed chords. The acoustical requirements placed upon the performance space are different in each case, but the proposed immersion index, along with the reverberation time, may perhaps serve to provide an adequate semi quantitative initial descriptor.

A FIRST ORDER INDEX

It is reasonable to take the reverberation time T_{60} of a performance space, supplemented by a knowledge of the enclosed volume V, as a zeroth order measure of sound quality. Well known curves [2] give ranges of reverberation time appropriate for particular types of music in halls of specified volume. What we now seek is a somewhat more refined index that measures, in some approximate sense, the relationship between early sound – that which is received within about 20 ms of the direct sound and is perceived as part of it – and the background of reverberant sound. We can think of this situation quite clearly in the case of contrapuntal organ music, because the input of sound energy to the space is approximately constant over a time much longer than the reverberation time, so that a steady reverberant state is achieved.

Consider the case of a simple rectangular "shoe box" performance space of width W, height H and volume V, with the organ distributed over one end. In this first-order approximation we neglect details of direct propagation and reflection and simply assume that all the sound power P of the organ is initially spread uniformly over the whole cross section of the hall after at most a single reflection, so that the prompt intensity is

$$I_P = P/WH \tag{1}$$

It is assumed that reflection losses at surfaces near the front of the hall can be neglected.

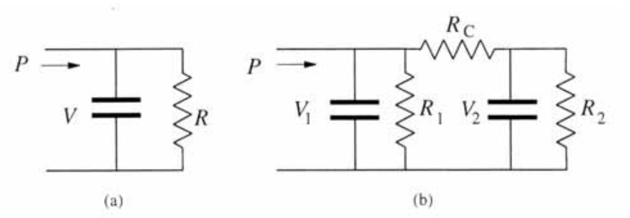

Fig. 1. (a) Energy flow network analog for a simple reverberant space. (b) Energy flow network analog for two coupled reverberant spaces.

The reverberant sound pressure can be calculated in a simple fashion, which we set out in some detail because it will be elaborated later. Fig. 1(a) shows an electric network analog for the acoustical problem, but in rather different terms from the usual analog. In this case the analog of electric current is taken to be the acoustic power P, and the analog of electric capacitance to be the enclosure volume V, so that the potential across the capacitance represents the acoustic energy density E. The electrical resistance R is proportional to the inverse of the acoustic absorption in the space, so that

$$R = \frac{K}{\sum \alpha_i A_i} \tag{2}$$

where A_i is the area and α_i the absorption coefficient of surface i and K is a constant that will be defined later. It is clear that the energy density for a constant acoustic power input P is given by $E_0 = PR$ and that, if this steady input is suddenly interrupted, the energy density will decay as $E_0 \exp(-t/\tau)$ where

$$\tau = RV = \frac{KV}{\sum \alpha_i A_i} \tag{3}$$

The resemblance between this equation and the normal Sabine reverberation-time relation is clear; the difference is that τ is the time for a decay of a factor e, which is equivalent to 4.343 dB, while T_{60} is the time for a decay of 60 dB, so that $T_{60} = 13.8\tau$. In the Sabine equation, τ is replaced by T_{60} and K by 0.163 $m^{-1}s$ when metric units are used, so that comparison yields the value $K = 0.012 m^{-1}s$. The steady energy density in the enclosure for input power P is therefore

$$E_0 = \frac{0.012P}{\sum \alpha_i A_i} = \frac{0.074 T_{60} P}{V} \tag{4}$$

The reverberant energy density given by (4) can be converted to a reverberant sound pressure p_R by the relation $E_0 = p_R^2/\rho c^2$, where ρ is the density of air and c the speed of sound in air, while the corresponding relation for the prompt sound pressure is $I_R = p_P^2/\rho c^2$. Using (1) and the second form of (4), we can then define the first order immersion index to be

$$S_1 \equiv 20\log_{10}(p_R / p_P) = 10\log_{10}(25WHT_{60}/V) \text{ dB} \tag{5}$$

in which the numerical coefficient arises from substituting c=343 ms^{-1}.

Extension of this index to halls that are not rectangular presents some problems that will be discussed later. For simple applications, it may be of value, however, to have a single number for the first order index, even for halls that are not rectangular. This is easily derived by noting that the volume of the hall is $V = \langle WH \rangle L$, where L is the length of the hall and the brackets indicate an average. With this convention

$$S_1 = 10 \log_{10} (25T_{60}/L) \quad \text{dB} \tag{6}$$

This approximation applies to halls of any shape, provided they can be regarded as a single space rather than a set of coupled spaces.

EXAMPLES

It is useful now to calculate this first order immersion index for a few well known halls for which appropriate figures are readily available in the literature [4,5,7]. Table 1 shows relevant physical data for six well known halls, together with numerical values for the index S_1. In each case the reverberation time quoted is an average over the 500–1000 Hz band with a full audience present.

There are several interesting features of the data in Table 1. In the first place, the immersion index is surprisingly close to 0dB, indicating near-equality between the reverberant sound level and the prompt sound level. Transients, however, will

Table 1. First-order immersion index for some well-known halls

Hall	**V (m^3)**	**W (m)**	**H (m)**	**T_{60} (s)**	**S_1 (dB)**
Symphony Hall, Boston	18,700	23	19	1.8	+0.2
Grosser Musikvereinssaal, Vienna	15,000	20	18	2.05	+0.3
Herkulessaal, Munich	13,600	22	16	1.8	+0.7
Royal Festival Hall, London	22,000	33	16	1.47	–0.6
Concertgebouw, Amsterdam	18,700	29	18	2.0	+1.4
Cologne Cathedral	230,000	L =	120 m	13	+4.3

show up much more sharply and will generally not be masked by the existing sound. The index also refers to the frequency range 500–1000 Hz. At the higher frequencies up to say 6 kHz, characteristic of much organ sound, the reverberation time is reduced to about 60% of its 500 Hz magnitude, which decreases the value of S_1 by about 2 dB and gives the listener much more directional information. It is for this reason that octave ranks, mutations, and mixtures are so important in contrapuntal organ music. The rather small range of values of S_1 should be borne in mind when evaluating differences between acoustic environments.

The second notable feature is that the value of S_1 appears to correlate quite well with subjective assessment of the acoustics of the halls concerned. Royal Festival Hall, for example, is said to be "crisp and clear", while the Concertgebouw is "warm and mello" [4]. The high value of S_1 for Cologne Cathedral similarly accords well with the listener's subjective feeling of immersion in the music. It should be noted, however, that the acoustics of such a large and reverberant cathedral, while excellent for general atmosphere, are perhaps not ideal for music except that specifically written for such buildings.

It is interesting, as an aside, to calculate the value of the index S_1 for a typical domestic bathroom, although the assumptions involved in its definition are not met in this case. The immersion index is about +4 dB, which is quite close to the value for a large cathedral. This perhaps explains the popularity of the bathroom environment with amateur tenors!

A SECOND ORDER INDEX

The index proposed above suffers from one very clear defect, which is that it does not allow for the influence of direct sound but collects it into a more generalized 'prompt sound'. It is possible to remedy this defect quite simply, but at the expense of additional complication in the calculation.

Referring to Fig. 2 for the case of a rectangular hall, the prompt sound received by a listener at point O can be divided into two parts, one of which is the sound propagated directly over a distance r and the other a more general early sound that has suffered a single reflection before reaching the listener. If I_{P1} is the directly propagated intensity from a source of acoustic power P, which we take to be a small source such as the mouth of an organ pipe radiating omnidirectionally into a half space, then

$$I_{P1} = P/\, 2\pi\, r^2 \,. \tag{7}$$

We must now determine the fraction of the source power that contributes to the more diffuse early sound that has suffered a single reflection before reaching the listener. To a reasonable approximation, only that sound that is reflected from the walls, ceiling or floor of the hall after travelling a distance less than $r/2$ along the hall meets this requirement – sound closer to the hall axis will either be experienced as direct sound or else be reflected to listeners nearer to the back of the hall. This is illustrated in the figure. If we define an equivalent circular hall with cross section of radius a so that $\pi a^2 = WH$, then the solid angle subtended at the source by the once reflected sound that can reach the listener at the point O is about equal to $2\pi r(r^2 + 4a^2)^{-1/2}$, so that the intensity in the once reflected prompt sound, assumed uniformly distributed over the hall cross section, is

$$I_{P2} \approx \frac{(1-\alpha_1)rP}{WH(r^2+4a^2)^{1/2}} \approx \frac{(1-\alpha_1)P}{WH}\left(\frac{\pi r^2}{\pi r^2+4WH}\right)^{1/2} \tag{8}$$

where α_1 is the area averaged absorption coefficient of the walls, ceiling and floor towards the front of the hall.

With this modification we can now define a second order immersion index S_2 by the relation

$$\begin{aligned} S_2 &\equiv 10\log_{10}\left(\frac{p_R^2}{p_{P1}^2+p_{P2}^2}\right) \\ &= S_1 - 10\log_{10}\left[\frac{WH}{2\pi r^2} + (1-\alpha_1)\left(\frac{\pi r^2}{\pi r^2+4WH}\right)^{1/2}\right] \end{aligned} \tag{9}$$

where S_1 is given by (5) or (6). Clearly the value of the index S_2 approaches that of index S_1 when $r > (WH)^{1/2}$, but for smaller values of r near the front of the hall the more refined S_2 is less than S_1 because of the increased contribution of direct sound. To aid in this comparison, Fig. 3 shows the quantity $S_2 - S_1$ as a function of the parameter $r/(WH)^{1/2}$. Specifically, $S_2 < S_1$ if $r < 0.52(WH)^{1/2}$ under the approximations we have adopted. For larger values of r, S_2 is always greater than S_1 because S_1 tends to overestimate the amount of once reflected sound in the prompt sound.

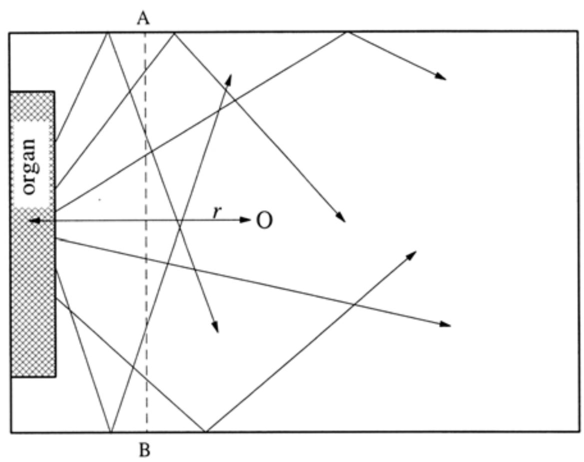

Fig. 2. Simple rectangular hall showing a listener at position O, a distance r from the organ. Only rays that strike the walls or ceiling to the left of the plane AB contribute to the prompt reflected sound. Other rays may contribute to the direct sound, and all ultimately contribute to the reverberant sound.

It is clear that the index S_2 gives more information about the acoustics of the hall than does index S_1, though it requires rather more labour to calculate, display and evaluate. In particular, Fig. 3 displays the dominance of direct sound at the very front of the hall, and even suggests an optimum listening position near $r = (WH)^{1/2}$, which is generally about one third of the distance from the front, at which the immersion index is high but, at the same time, the amount of direct sound is large, giving clarity. It is open to question, however, just how meaningful some of this extra information is, since it involves assumptions about wall and ceiling reflections that may well vary from one hall to another.

The relation of index S_2 to the various clarity indices that measure the ratio of directly propagated sound to reverberant sound in various parts of the hall is im¬mediately apparent. The main distinction, apart from a change in sign, is that the prompt once reflected sound is generally omitted or added into the reverberant sound. The intensity of the directly propagated sound, and thus the simple clarity index, therefore falls by 6 dB for each doubling of distance, so that the index does not achieve a saturation value. While this comment is not meant to denigrate the value of such a clarity index, the constant value of S_1 and the saturation behaviour S_2 confer desirable simplicity on these measures.

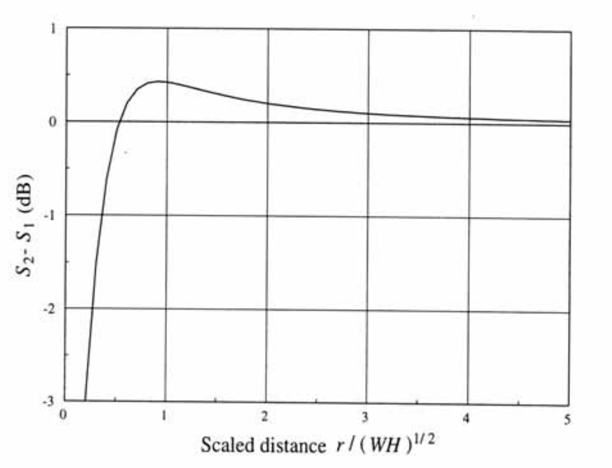

Fig. 3. Difference between the second order immersion index S_2 and the first order index S_1 as a function of distance r from the sound source, normalized in terms of the square root of the hall cross section WH.

NON RECTANGULAR HALLS

So far our discussion has dealt specifically with rectangular halls, or at least halls of constant cross section. While this class encompasses most concert halls, at least approximately, it is important to examine whether the index can be applied to halls of other shapes. Equation (6) has already suggested a way in which the first order index might be calculated for a hall of general shape. It is tempting to go one step further and use equation (5), with the cross sectional area WH taken as a function of distance from the front of the hall, to derive a spatially varying first order index for a hall of arbitrary shape, but a trial calculation for a fan shaped hall suggests that this overestimates S_1 near the front of the hall and underestimates it near the rear. It is probably necessary, when spatial variation of the index is being examined, to follow a course such as that adopted in the calculation of S_2.

In a hall with diverging side walls, the prompt wall reflections are directed much more towards the rear of the hall than they are in a rectangular hall. This has the effect of increasing the numerical coefficient 4 in the term $4WH$ in the denominator of equations (8) and (9) to a much larger value, depending upon the angle of divergence of the walls. While the walls contribute only part of the reflected sound, this modification will decrease the level of once reflected sound near the front of the hall and increase its level near the rear, thus tending to equalize the index S_2, apart from the effect of direct sound in the immediate front of the hall. Few fan shaped halls, however, have sidewalls that are simple planar reflectors, so that further detailed consideration, inappropriate for a simple index of the type proposed, is required.

The uncertainties involved in constructing appropriate modifications to the index S_2 in this way, however, bring into question its value as a design parameter in this case. For halls that are not well approximated by a simple rectangular shape, it may therefore be well to use only the simple uniform approximation (6) for S_1 and leave any more sophisticated index to the detailed design stage.

COUPLED VOLUMES

In many architectural designs, though perhaps not generally in concert halls, it is possible to consider the enclosure as consisting of two volumes more or less closely connected, rather than as a single volume. An example might be a rather long cathedral, with the nave linked to the chancel through a rather low or narrow tower crossing or, in the case of a concert hall, a reverberant enclosure purposely left behind or above the organ and coupled to the hall through a relatively small aperture. The network analog for such a situation is shown in Fig. 1(b). Each volume V_i can be considered as having an exponential decay constant $T^{(i)}_{60}$ that is derived by supposing the coupling aperture to be blocked by an ideal diffuse reflector. This information then defines the two resistive components R_i through the relation

$$R_i = T^{(i)}_{60} \,/\, (13.8 V_i). \tag{10}$$

The coupling resistance R_C is simply equal to the constant K of equation (2) divided by the area A_C of the coupling aperture. Thus

$$R_C = K / A_C = 0.012 / A_C \tag{11}$$

where metric units are assumed.

It is now straightforward to solve the network and calculate the energy densities E_1 and E_2 in both enclosures 1 and 2, assuming the sound source to be in enclosure 1. The results are

$$E_1 = \frac{(R_C + R_2) R_1 P}{R_C + 2R_1 + R_2}$$

$$E_2 = \frac{R_1 R_2 P}{R_C + 2R_1 + R_2} \tag{12}$$

and from these the reverberant sound pressures can be calculated using the relation $E = p_R{}^2/\rho c^2$ as before. The prompt sound pressure in enclosure 1 can be calculated just as for a simple enclosure, while the derivation for enclosure 2 follows the same path except that the power of the source is taken to be the prompt sound power I_1A_C entering through the coupling aperture, where I_1 is the prompt intensity at this aperture. Clearly there are additional problems if the aperture is in a side wall of enclosure 1, as would happen, for example, in the transepts of a cathedral, and this simple approach is then no longer adequate. Calculation of the immersion index for the two spaces now proceeds as before, and either S_1 or S_2 can be evaluated for each. The detail of the result for S_2 is too complicated to quote here, but for the first order indices we find

$$S_1^{(1)} = \left(\frac{E_1 cWH}{P}\right)^{1/2}$$

$$S_1^{(2)} = \left(\frac{E_2 cW^2H^2}{PA_C}\right)^{1/2} \tag{13}$$

where E_1 and E_2 are given by (12).

The analog network of Fig. 1(b) can also be used to calculate the form of the decay transient for an abruptly terminated sound input. It will consist, in general, of two superimposed exponential decays

$$A\exp(-t/\tau_1') + B\exp(-t/\tau_2') \tag{14}$$

where τ_1' and τ_2' are modified versions of τ_1 and τ_2, the extent of the modification depending upon the area A_C, of the coupling aperture. In enclosure 1, $A > B$, while in enclosure 2, $B > A$. The expressions for τ_1' and τ_2' can be calculated in a straightforward manner, but are algebraically complicated. Since this topic is of no immediate concern to us here, we shall not pursue the subject further.

DISCUSSION

It has been the purpose of this paper to propose a simply calculable index that has the potential to describe the sensation of auditory immersion of a performance space and thus to supplement other simple indices such as reverberation time and volume per seat. The first order index has the advantage of being a single number, with approximate level 0 dB, that can be calculated immediately if the volume, cross-section and reverberation time of the hall are known. The second order index varies spatially throughout the hall and gives additional information that may be of use.

Comparison of the simple first order index evaluated for several well known concert halls containing organs, and also for a Gothic cathedral, suggests that it may indeed be useful as a preliminary guide for assessing the sound of an organ in a projected building, before going to the very much greater labour of making detailed acoustic calculations. The index also appears to give useful information in the case of much smaller spaces.

Because of its computational simplicity, the first order index S_1 commends itself particularly for this purpose, when it is used together with other simple indices such as reverberation time. Only by practical trials, however, can its usefulness be established. A quick survey of modern books on the subject [8,9] shows that, while the field has advanced considerably since 1998, no simple immersion index of the type described here has been proposed. Perhaps, therefore, my suggestions may prove practically useful.

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A LOW COST (NEAR EXPENDABLE) UNDERWATER SOUND LOGGER USING AN MP3 PLAYER: TESTING IN SYDNEY HARBOUR

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A low cost, self contained, portable system based around a modified MP3 player has been constructed to record ambient underwater noise. High resolution acoustic data are stored in PCM WAV file format with a depth of 16 bits and 96 kHz sampling. The MP3 player, preamplifier, and battery are mounted in an underwater housing made of PVC pipe to which is attached a hydrophone on a short buoyed cable. This provides a near expendable package suitable for unattended deployments where equipment loss or damage are possible factors. System functioning was verified by several 24-hour deployments in Sydney Harbour, following which it was used for the international KONDARI ports and harbour security trial at Garden Island. Three characteristic sound regimes were observed: (1) a regular succession of ferry passage events from morning till midnight; (2) irregularly occurring high energy vessel events other than ferries; (3) lower sound levels from midnight to six a.m. dominated by snapping shrimp, and by pumps and ancillary engines on moored vessels and dockyards.

Figure 1. Components of the MP3 based underwater ambient noise recording system. The underwater housing is 16 cm in diameter and 42 cm in length.

INTRODUCTION

A low cost portable underwater noise recording system based on an MP3 player has been constructed. System functioning was verified by short term recordings made in central Sydney Harbour, prior to use of the system in the international KONDARI ports and harbour security trial at Garden Island. The MP3 player is used to record acoustic data in high resolution PCM WAV file format, not the lossy MP3 format, which is based on perceptual noise shaping and other compression algorithms. This paper describes the system and its development in brief. The nature of the acoustic time series obtained for system verification suggests they are typical of the central harbour ambient underwater noise conditions, and although uncalibrated for absolute levels, they are presented as being of interest in their own right. It is intended to calibrate the system at the DSTO calibration facility at Woronora Dam.

Measuring underwater noise in Sydney Harbour and other busy port areas provides special problems compared to land based work. The first problem is accessing the site, which usually involves a boat journey. Measurements at a fixed point may be made by anchoring, and lowering hydrophones into the water column, or a vessel can simply drift during measurements. However, in high density vessel traffic areas this technique may not be viable. If hydrophone cables cannot be run to shore in such cases then underwater housings must be constructed and deployed to enable measurements to be made. In busy waterways the instrument package must be deployed unattended on the seabed, without the luxury of a surface buoy for recovery. Retrieval of the package is by diver, or by use of an electro-mechanical acoustic release, which is triggered externally by a coded acoustic pulse to allow a small buoy and tether to rise to the surface. There is always the possibility that the package may be lost, either by simple failure to locate it, or by damage from boat or fishing operations. Trawling is no longer undertaken in Sydney Harbour, removing one potential problem for the present deployments. In some areas wave and current action must be considered, especially if these may cause burial of equipment in the seabed. Equipment cost then becomes a factor, with near disposable kit a desirable option. Details of the relatively low cost underwater noise recording system developed to meet this requirement are presented in Section 2. Descriptions of underwater sound regimes for Sydney Harbour gained from the system testing follow in Sections 3 and 4.

INSTRUMENTATION

The underwater ambient noise recording system consists of an ITC 1032 omnidirectional hydrophone, a custom built low noise preamplifier, and a modified MP3 player (Figure 1). The hydrophone is buoyed to float about half a metre above the seabed. In the open sea the package can be left to drift on the surface. The underwater housing is a section of PVC pipe with removable end caps sealed by double O-rings. The preamplifier is a custom design with a gain of 20 or 40dB, constructed with low input capacitance, low noise, and low power consumption as the design constraints. High resolution PCM WAV file format is recorded, with a depth of 16 bits. The MP3 player forms a convenient logging platform, having a real-time clock, and on-board programming capability. Exploiting these off-the-shelf capabilities enables simpler system development and reduced costs. For example, the system is set up for use by simple menu driven commands accessed through the MP3 screen displays.

The system employs the highest sampling rate of the MP3 player of 96 kHz. Lower sampling rates may also be selected e.g. 8 kHz. The two stereo channels record the same signal, but to maximise dynamic range gains of 0dB on the first channel and 12dB on the second are used. This allows one channel to avoid overload during energetic events, and allows the other channel to record useful signals during quiet times. The 12 dB for this particular system was chosen after preliminary field tests. The redundancy in the recording provides verification of more unusual transient events, and possible allowance for some hardware failures. The system was designed for burst sampling, e.g. 1 minute sampling in every 10 minutes. This allows a representative time series to be obtained whilst minimising the amount of data recorded. Other logging selections are 5, 10, 20, 60, and 120 seconds at intervals of 2, 5, 10, 20, or 60 minutes. However, if e.g. 120 seconds is selected at 2 minute intervals, the system will generate an error message, and request another setting. An error message is also output if the selections would cause the buffer length to be exceeded. The specifications are:

Preamplifier
Noise: -166dBVrms/√Hz
Input capacitance: 25pF
Gain: selectable 20dB and 40dB

MP3 player
Model: Iriver H10 20GB
Storage: 20GB 1.8 inch hard disk (a modification allows use of a compact flash card, with consequent saving in power usage, and elimination of moving parts)
Software: Rockbox (www.rockbox.org) with a custom recording plugin. Rockbox supply open source replacement firmware for portable audio players.
Gain: selectable -12dB to 12dB in 3dB steps
Channels: Two with independent gain settings
CODEC: Wolfson WM8731
Power Supply: a 12V, 12Ah rechargeable switchmode lead-acid battery provides several days of recording
Data format: 16 bit PCM WAV. Maximum individual recording length is restricted by the available internal RAM (approximately 30MB) as the entire recording is buffered before being written to disk.

An earlier recording system used a DAT tape recorder, but the advantages of tape were offset by system unreliability. The MP3 player is cheaper than the DAT recorder, and purchase of DAT tapes is not required. Data archiving requirements are modest, as twenty-four hours recording of one minute in every ten sampled at 96 kHz and 16 bits in stereo fits comfortably on a DVD.

Frequency Response
The system has not been fully calibrated, so this information is imperfectly known. The recommended operational hydrophone frequency range is 10 Hz to 45 kHz. A hydrophone calibration using the reciprocity method confirms good receive voltage response from 500 Hz to 40 kHz, with a 10 db rolloff from 40 to 50 kHz, and upper 3db point at 42 kHz. Linearity of response is specified by the manufacturer from 500 Hz to less than 10 Hz, depending on the input impedance of the preamplifier. Preliminary measurements indicate a lower frequency 3 db system response of 23 Hz. Testing of the MP3 itself indicates linear response at 96 kHz sample rate from DC to 25 kHz, with the upper frequency 3db point at 42 kHz. The combined hydrophone and MP3 information indicates the 3 db upper frequency point of the system as 33 kHz.

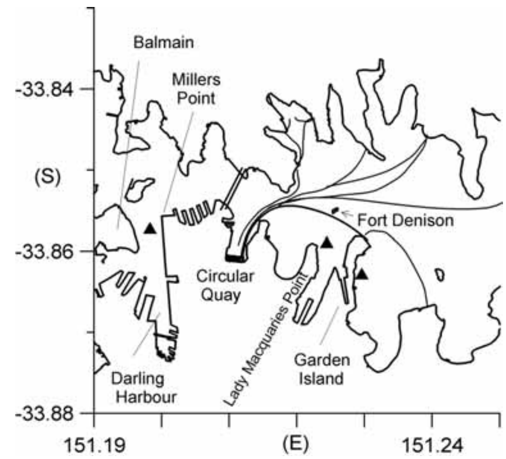

Figure 2. The three measurement sites in Sydney Harbour (▲) and general routes of ferries passing near Lady Macquaries Point, and east of Garden Island.

RESULTS OF SYDNEY HARBOUR TESTING

The system was deployed at various Sydney Harbour locations (see Figure 2) to verify mechanical and acoustical functioning in the field. Measurements from three sites are presented. These were made northwest of Garden Island in November 2008; west of Millers Point in January 2009; and east of Garden Island. Several days of recordings were made at the last named site for the international KONDARI ports and harbour security trials (see e.g. the web site http://www.dsto.defence.gov.au/news/5580/).

Northwest Of Garden Island
On November 25, 2008 the system was deployed on the seabed at 14 m depth for a 24 hour period between Fort Denison and Lady Macquaries Point, a well known Sydney landmark (Figure 2). One minute was recorded in every ten minutes. Ideally, concurrent visual or video observations would be made of ship traffic and other activities, to correlate major sound events arising from unscheduled shipping movements to events in the recordings. However, this was not carried out for the

present data. An assessment of underwater sound conditions and frequency of events over a typical day in this busy working harbour was required, rather than detailed knowledge of the causes.

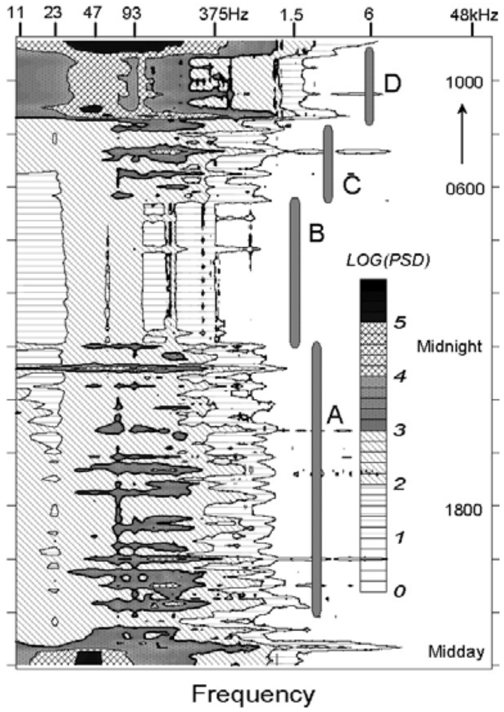

Figure 3. Logarithm of PSD (Power Spectral Density, uncalibrated) as a function of frequency and elapsed time for a day's recording off Lady Macquaries Point on 25 November 2008. Letters A-D identify particular persistent noise conditions or sound sources. A = ferry traffic, B = moored vessel noise and equipment at Garden Island, C = ferry traffic, D = local vessel traffic.

A plot of power spectral density with frequency and elapsed time (Figure 3) summarises the time history of underwater noise at the site. Selected WAV files were replayed on a PC fitted with a DVD reader to identify the major noise events noted in this record. Three main sound regimes are experienced at the site. These are (1) a succession of regularly spaced episodes of higher sound levels from early morning till midnight generated by ferries, (2) a relatively quiet period from midnight till 6 a.m. dominated by sound from moored vessels and industrial machinery at Garden Island dockyard, and (3) higher sound levels from local ship traffic other than ferries, e.g. barges, passing close to the recorder.

From deployment at 1 p.m. to midnight a succession of higher noise power events centred on 50 to 375 Hz reflects the passage of screw-driven ferries, Jetcats (high speed catamarans), and other vessels. Movements of larger vessels (generally ferries) cease at midnight. Noise levels at frequencies below 23 Hz begin to drop after 8 p.m. and reach a minimum by 10 p.m. except for two high energy vessel events near 11 p.m. and midnight. These low frequencies reappear after 6 a.m. the following morning when industrial machinery is switched on at nearby Garden Island dockyard. Almost constant sound levels are seen in particular frequency bands from midnight till 6 a.m. at frequencies below 93 Hz, coupled with a regularly spaced series of events near 300 Hz. This latter signal is attributed to pumps and other ancillary engines on vessels moored at Garden Island, or on land based facilities. A near continuous background low level broadband crackle is attributed to snapping shrimp. This location is therefore never truly quiet, but is always subject to both natural and anthropogenic noise sources.

Particular types of craft may be identified by their acoustic signatures. The records of passing Jetcats and screw-driven ferries in Figure 3 are associated with peaks in noise power levels near 47, 71, 95, 132, and 184 Hz, although it is not immediately clear which ferry type is responsible for which frequency, as different models appear to have some energetic frequency bands in common. In between the passage of ferries, a distinctive drop in sound levels about 23 Hz is noted.

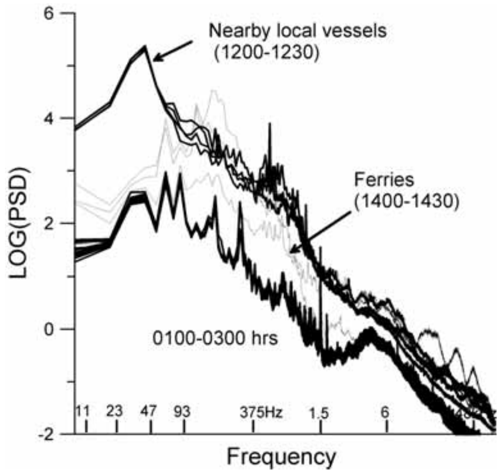

Figure 4. Examples of characteristic acoustic spectra off Lady Macquaries Point for the quietest times in the record (0100-0300 hrs), for the passage of ferries (1400-1430 hrs), and for vessels passing close to the measurement site (1200-1230 hrs).

Examples of individual spectra in Figure 4 provide a different view of some of the ambient noise regimes and events observed in Figure 3. For example, peaks in sound levels at particular frequencies are more readily identified. The quietest part of the record for 0100-0300 hours shows a sound level decreasing at frequencies below 23 Hz, and several distinctive peaks attributed to Garden Island machinery and vessels. The noise levels from ferries for 1400-1430 hours in Figure 4 often do not surpass the peaks in the quietest part of the record for 0100-0300 hours. This is not the case for other vessels near the measurement site, as shown for 1200-1230 hours, but whether or not because they are closer or louder is unknown.

Millers Point

A 24 hour recording of one minute at five minute intervals between Millers Point and Balmain shows similar results to the site at Lady Macquaries Point (Figure 5). Three ferry routes pass this location transiting in and out of Darling Harbour, to parts of Balmain, and upriver to the west to Parramatta. The measurement site is closer to the ferry routes than at Lady Macquaries Point, leading to more high frequency detail in the acoustic record. The data from this site were used to plan optimal deployment times for other types of underwater surveying equipment.

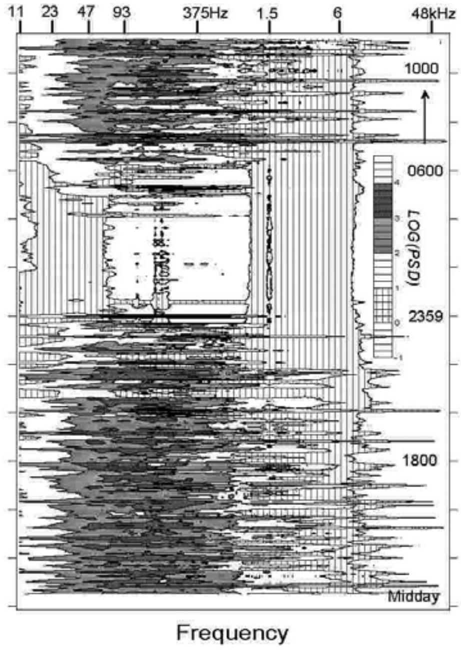

Figure 5. Logarithm of PSD (Power Spectral Density, uncalibrated) as a function of frequency and elapsed time for 24 hours acoustic recording between Millers Point and Balmain on 20 January 2009.

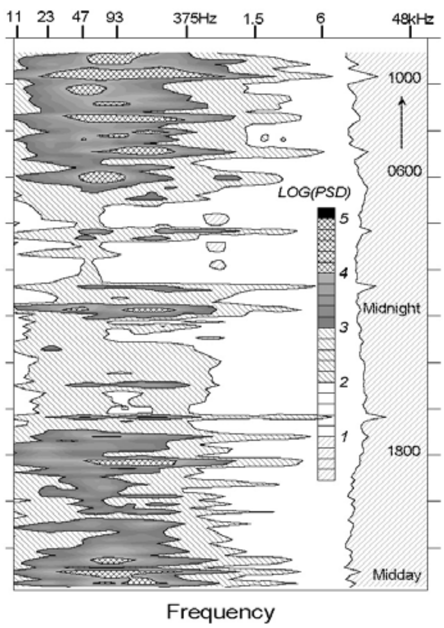

Figure 6. Logarithm of PSD (Power Spectral Density, uncalibrated) as a function of frequency and elapsed time for 24 hours acoustic recording east of Garden Island on 09 February 2009.

East Of Garden Island

The system was deployed off Garden Island as part of the international KONDARI ports and harbours security trial. One day on 09 February 2009 from a three-day record sampling one minute in every ten to the East of Garden Island (see Figure 2) is presented (Figure 6). Only one ferry route passes near this location, travelling between Circular Quay to the West and Watsons Bay in the eastern harbour, leading to fewer major events in the record than at Lady Macquaries Point and Millers Point. Ferry services on this route commence near 0700 hrs, and the last evening ferry docks at Circular Quay at 1932 hrs. The site is about 1.7 times farther from the closest point of ferry approach than the other two test sites, which is expected to reduce received ferry sound levels. According to ferry timetables, an apparent broadening in time of ferry signatures compared to the two other test sites may be caused by inbound and outbound ferries on this route regularly passing each other near Garden Island. Unlike the other two sites there is no well defined structure in the nighttime portion of the record, attributed to the absence of larger moored vessels east of Garden Island, and the absence of major wharves.

DISCUSSION

Underwater ambient noise recordings have been made at several locations in central Sydney Harbour with a low cost, purpose built, low noise amplifier coupled to an MP3 player. The acoustic equipment is housed in a sealed section of weighted PVC pipe, which is placed on the seabed. The acoustical recordings indicate successful system mechanical, acoustical, and software functioning. The on board capabilities of the MP3 recorder provide a range of sophisticated functions accessed through the open source Rockbox firmware project (see the web site www.rockbox.org).

West of Garden Island near Lady Macquaries Point the harbour is never completely quiet. Continually active snapping shrimp provide a broadband background noise floor, which has been rather aptly described as sounding like the frying of bacon. Machinery and moored vessels provide a regular and continuous source of sound, with higher intensity during daylight hours. Regularly scheduled ferry and tourist vessel events add to this background from early morning till midnight. Some component of ferry noise is received almost continually during this period, both from ferries passing near Garden Island landwards of Fort Denison, and from particular energetic frequency bands from more distant ferries elsewhere in the harbour. Most ferry traffic arrives at or departs from the busy Circular Quay area, which is a relatively short distance to the west of the measurement site (see Figure 2). Highest sound levels at the measurement site originate from occasional shipping other than ferries. The same general underwater sound regime is seen west of Millers Point. The area east of Garden Island experiences fewer ferries, and larger moored vessels are generally absent, resulting in a quieter ambient noise record. Engineering works, strong winds, and times of rain will increase underwater sound levels. Otherwise the acoustic results are expected to represent the typical underwater sound events and regimes of the middle reaches of Sydney Harbour.

ACKNOWLEDGEMENTS

Mr Tony White of DSTO is thanked for assistance with deployment and recovery of equipment. Mr Mark Savage of DSTO provided hydrophone calibration.

DESIGN AND TEST OF A FEEDBACK CONTROLLER FOR ATTENUATING LOW FREQUENCY NOISE IN A ROOM

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A single-channel feedback control system is presented for global noise attenuation inside a room. The controller design is based on the compensation filter approach of classical control theory. To demonstrate the global noise attenuation capability of this simple control scheme experimental results of noise control in an office are presented. They show that the positioning of the error microphone relative to the control loudspeaker greatly affects the global noise attenuation performance. If the microphone is placed too close to the control source, the microphone signal is dominated by the near field and little global noise reduction is achieved. If the error sensor is placed further away where the near field has little effect on the microphone signal, noise reductions of about 10 dB can be obtained over low frequency ranges.

INTRODUCTION

Active noise control technology is an attractive solution for the attenuation of low frequency noise in enclosures. Over the past decade, it had many successful applications including the control of sound pressure level in 1-D acoustic ducts [1], inside the fuselage of passenger jets [2], vibration suppression that reduces the structural acoustic coupling and therefore reduces the interior noise in cars [3], and active control for noise suppression in payload fairings [4]. In terms of control strategies, feedforward control has been widely used. While feedforward control has many advantages, its success relies on the availability of causal reference signals which have to be highly correlated to the noise to be cancelled [5]. For some applications such as the attenuation of random noise in office spaces or bedrooms, such reference signals are either not available or very expensive to obtain. In these situations, feedback control can be an alternative solution.

Progress on feedback control of the sound field in enclosures has been made in the area of prediction and control of noise radiated into enclosures using structural sensing [6]. Such a control scheme was developed for applications where neither a coherent reference signal, nor the sound field to be controlled were available. A method for the selection of the sensor and actuator positions was given based on a transformation of the problem into radiation modes. An optimal feedback control approach which allows the control of radiated pressure into a defined subvolume of the cavity using only structural actuators and sensors was also demonstrated. The main drawbacks of the method were its inability to model complex structures with a large number of modes accurately, failure to include robustness due to uncertainties and the need for system identification.

Techniques of state feedback control of sound fields in enclosures were reviewed by Samejima [7], who also investigated theoretically and experimentally the use of state feedback to achieve a desired modal distribution of the enclosed sound field. Pole allocation was employed to obtain the state feedback gain vector such that the roots of the closed-loop system have the desired modal distribution. This method can be difficult to employ in irregular shaped rooms and was used mainly for changing the acoustic resonances inside a cavity.

A method was also proposed by Yuan [8] to improve active noise control in enclosures. A virtual sensing technique by using two judiciously placed microphones was suggested in order to predict a virtual signal. However, an exact mathematical model between the virtual and physical sensors was required over the entire frequency band of interest for broadband control in a lightly damped enclosure. This can be impractical for rooms. No discussion of the global attenuation characteristic of the control strategies was provided in the paper.

Al-Bassyiouni and Balachandran [9] proposed a zero spillover scheme for active structural acoustic control inside a three-dimensional rectangular enclosure into which noise is transmitted through a flexible boundary. Piezoceramic patches mounted on the flexible boundary were used as actuators and microphones placed inside and outside of the enclosure were used as sensors. The technique took into account the effect of inherent acoustic feedback in the design of the control scheme. The results showed that significant attenuations can be obtained at the error microphone and near the collocated microphone locations, and that a good attenuation can be obtained over a large area of the enclosure in the presence of tonal and broadband disturbances. The experiments also demonstrated that the energy levels in the flexible panel increased significantly when applying the control scheme. The control algorithm used did not take into account the robustness of the control system to any possible changes in ambient conditions and other factors.

More recently, several feedback control schemes have been applied to control noise inside cavities. For minimising sound radiation, Hong and Elliott [10] examined closely spaced local feedback control systems on a honeycomb panel using an accelerometer and a piezoceramic actuator. It was found that the global control performance was affected by local coupling of the control channels, and the multichannel system did not yield a significant improvement in performance because of a decreased gain margin. Creasy et al. [11] described a method for adaptive energy absorption in acoustic cavities based on an adaptive scheme consisting of a self-tuning regulator that has the ability to target multiple modes with a single actuator. The inner control loop of the regulator used positive position feedback in series with a high and low-pass Butterworth filters for each controlled mode. The outer loop consisted of an algorithm that locates the zero frequencies of the collocated signal and uses these values to update the resonance frequency of the positive position feedback filter and the cut-off and cut-on frequencies of the filters. Experimental results show the robustness of the method in the presence of changes in the resonance frequencies of the system and the reduction of spillover. de Oliveira et al. [12] proposed a methodology to derive a fully coupled mechatronic model that deals with both the vibro-acoustic plant dynamics as well as the control parameters. The inclusion of sensor and actuator models was investigated since it can cause limitations to the control performance. The proposed methodology provided a reduced state-space model derived from a fully coupled vibro-acoustic finite element model. Experimental data on a vibro-acoustic vehicle cabin mock-up were used to validate the model reduction procedure. A collocated sensor/actuator pair was considered in a velocity feedback control strategy. The results showed that an optimal design could only be achieved when considering structure and control concurrently. Although the method is useful at the design stage its application to active noise control in existing rooms is impractical.

In this paper, a practical feedback control system for noise attenuation in a room is presented. It is aimed towards the development of a simple active noise control unit for household use (such as bedrooms) and offices in workplaces. The control system is designed based on classical control theory [13], and the controller can be realised with analogue electronic circuits developed for feedback noise control ear defenders, or with digital signal processors employed for traditional active noise control. The objective of the paper is to demonstrate the feasibility of active control of low frequency noise in rooms using classical feedback control theory and to investigate the effect of room acoustics on the performance of the noise control. Control performance measures such as system stability, control bandwidth and global controllability are studied and supported by experimental results.

1. DESIGN OF THE FEEDBACK CONTROLLER

Figure 1 shows the feedback control system considered in this paper. The aim of the control is to achieve global noise attenuation in a room with a control system consisting of a single loudspeaker, a single microphone and a controller.

The heart of the control system is the controller in which a control signal is generated using the error signal from the microphone and a compensator to be designed. There are several ways to design a controller in a feedback control system, ranging from a traditional PID (Proportional-Integral-Derivative) control approach based on classical control theory to a more advanced state-variable approach based on modern control theory [13].

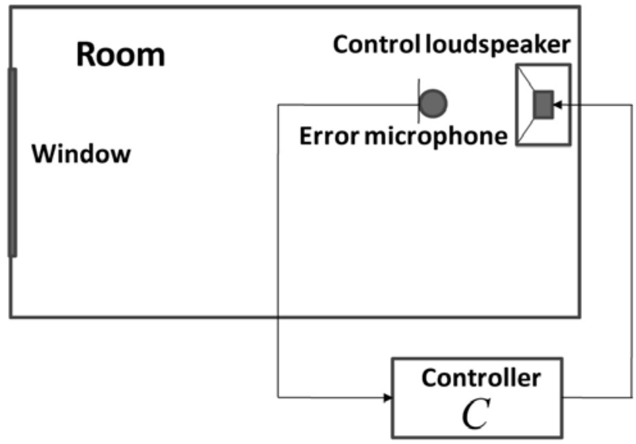

Figure 1 Schematic of the feedback control system.

In this paper, a compensation filter approach based on classical control theory is employed, as it is easy to implement with less expensive analogue circuits. The compensator to be used as the controller has a basic form

$$C(s) = K\left(\frac{s^2 + 2\xi_z\omega_z s + \omega_z^2}{s^2 + 2\xi_p\omega_p s + \omega_p^2}\right), \tag{1}$$

characterised by an angular frequency ω_z and damping ratio ξ_z for the zeros, and ω_p and ξ_p for the poles. In Eq. (1), K is the frequency independent gain and s is the Laplace variable. This form of compensator enables the gain of the open-loop system to be made high in the frequency region where attenuation is wanted while the phase recovers to zero at high frequencies. In order to obtain the best possible performance, the compensator parameters have to be optimised.

In the optimisation of the compensator parameters, the objective function to be minimised can be chosen as an energy term representing the amount of acoustic energy at the error microphone. However, for a practical system the minimisation of energy is not the only performance criterion that has to be taken into account. In fact there are two other important factors that need consideration. These are the Nyquist stability criterion and a term associated with fluctuations (uncertainties) in the open-loop frequency response caused by any changes in the physical system. A simple approach to the optimisation problem is to use multi-objective optimisation which enables a clear and easy problem formulation as well as preferences to be entered into the numerical design. The three objectives to be minimised in the compensator design make a vector of objectives which must be traded off in some way.

The Goal-Attainment method [14] is used here since it is very practical and requires less guessing on the part of the designer than other methods. This method involves expressing a set of design goals $\mathbf{f}^*=\{f_1^*, f_2^*, ..., f_m^*\}$ which is associated with a set of objectives $\mathbf{f}(\mathbf{x})=\{f_1(\mathbf{x}), f_2(\mathbf{x}), ..., f_m(\mathbf{x})\}$. The formulation of the problem allows the under or over-achievement of the objectives. This enables the designer to be relatively imprecise about initial design goals. A vector of weighting coefficients, $\mathbf{w}=\{w_1, w_2, ..., w_m\}$ controls the amount of under or over-achievement of the goals and lets the designer select the relative trade-offs between objectives. Before the Goal-Attainment method is used, the objective functions that determine the performance of the feedback system have to be defined. The three objective functions are terms related to the energy at the error microphone, the Nyquist stability criterion and the stability margins.

Energy-related objective function

In the optimisation of the compensator parameters, the open-loop transfer function of the control system without the compensator, H, is measured. Using the measurement data and Eq. (1), the energy-objective function can be written as [15]

$$f_1\left(K,\xi_z,\omega_z,\xi_p,\omega_p\right)=\sum_{i=1}^{N}\frac{W_i}{\left|1-C(\omega_i)H(\omega_i)\right|^2}, \quad (2)$$

where W_i is a frequency weighting window which allows emphasis at important frequencies.

Stability-related objective function

The second objective function takes into account the stability of the closed-loop system which needs to satisfy the Nyquist stability criterion. For the case at hand and for systems which are stable in open-loop, it states that systems whose open-loop loci do not encircle the (1,0) point in the complex plane will be closed-loop stable. The stability-related objective function can be defined by using an exponential function as [16]

$$f_2\left(K,\xi_z,\omega_z,\xi_p,\omega_p\right)=e^{\left[\alpha\left(\mathrm{Re}_{\max}-\beta\right)\right]}, \quad (3)$$

where Re_{max}, a function of the compensator parameters, is the maximum of the positive intercepts with the real axis of the Nyquist plot (of the compensated open-loop transfer function), and α and β are positive constants adjusted empirically. Typical values used are α=3 and β=0.5 [16], which indicates that the maximum positive intercept of the real axis in the Nyquist plot is desired to be 0.5.

Fluctuation-related objective function

In order to prevent any instability due to any fluctuation in the system response, a fluctuation-related objective function has to be minimised. This term is based on the gain and phase margins chosen as safety limits by which the system behaviour can deviate from a mean behaviour without causing instability. The fluctuation objective function is chosen as [16]

$$f_3(K,\xi_z,\omega_z,\xi_p,\omega_p)=\sum_{j=1}^{M}e^{\frac{\left(\Phi-|\phi_j|\right)}{\gamma}} \quad (4)$$

where Φ is the predefined phase margin, φ_j are the phase shifts with magnitude less than or equal to the phase margin and γ is a constant which allows the magnitude of f_3 to be adjusted so that it becomes comparable to the values of the two other objective functions when the optimisation is successful. In a practical situation a phase margin of 45° and a gain margin of 6 dB are often used. The weighting constant γ is then chosen to be 45 [16].

The three objectives are minimised simultaneously in order to obtain the optimal coefficients of the second order minimum phase filter. Several such filters can be cascaded together to improve the performance of the control system if needed.

2. EXPERIMENTAL INVESTIGATION

In order to demonstrate the feasibility of feedback control of noise using the controller designed in Section 1, experiments were conducted in an office of volume 4.0×2.9×3.0 m^3. These experiments also allowed a study of the effect of room acoustics on the locations of the sensor and actuator and hence on the control performance. The office is furnished with two desks and two filing cabinets and its floor is covered with carpet. The averaged reverberation time of the room below 200 Hz is about 1.5 seconds. There is a full-size window on one of the walls through which noise is transmitted from a nearby workshop. In the experiments, the primary noise was either generated internally by a loudspeaker standing next to the window or transmitted into the room from the workshop. The control loudspeaker was located at the opposite end of the room. Figure 2 shows a typical plot of the magnitude of the low frequency acoustic response of the room. It can be seen that the room modes (or room resonances) are all well damped except for the first mode at 42 Hz. Thus, according to Nelson and Elliott [17], global noise attenuation with a single control source is only possible around that frequency region. Or quantitatively, the required control bandwidth is about 50 Hz.

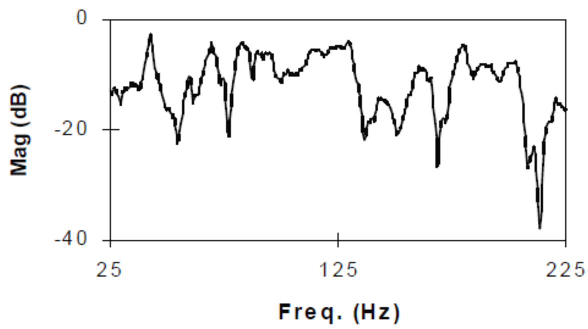

Figure 2 Typical acoustic frequency response of the room.

The performance of the control system is evaluated based upon the sound pressure measurements. The sound pressure spectra were measured at 15 locations distributed evenly along the diagonal line of the room. The measured spectra were then averaged over these 15 locations to form a global index. The comparison of the index with and without control indicates the global control performance of the system.

The locations of the control loudspeaker and error microphone are always important for effective control. It is well known that in order to meet the requirements of controllability and observability, the control loudspeaker and error microphone should not be located on the node-lines of those room modes to be controlled. However, in order to have effective global attenuation with feedback control, other considerations are also required. For instance, in order to minimise the effect of control spillover, it is desirable to have the control loudspeaker located on the node-lines of the room modes which cannot be controlled.

One of the issues to be investigated in the experiments is the relationship between stability, control bandwidth and global controllability. From the stability and control bandwidth point of view, the error microphone should be placed as close as possible to the control loudspeaker. However, this often leads to local control rather than the required global control due to a very strong near field in the vicinity of the control loudspeaker. This can be illustrated by the following examples.

In the first example, the error microphone was located 13 cm away from the control loudspeaker. Figure 3 shows the open-loop frequency response of the uncompensated system of this arrangement. It can be seen that the two phase cross-overs (phase cross-over being 0° in the convention adopted here) in the frequency region of interest are well apart (30 and 740 Hz). This provided a good margin for the compensator design, as the required control bandwidth is merely 50 Hz. As a result, it was possible to design a compensator consisting of two second order filters cascaded together.

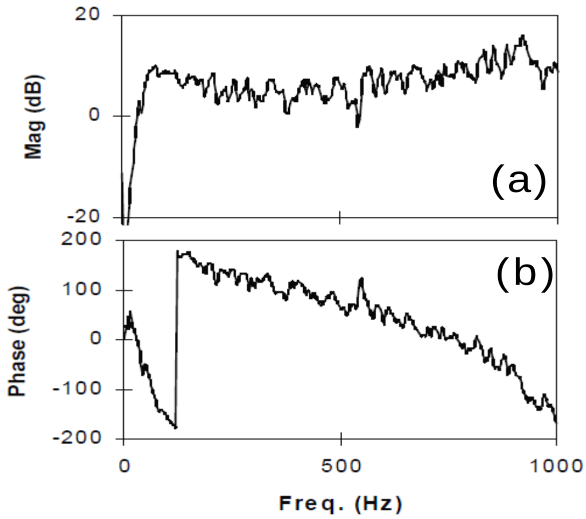

Figure 3 Magnitude (a) and phase (b) of the measured open-loop frequency response of the uncompensated system in Example 1.

Figure 4 shows the control result achieved from the 4th order compensator design. The primary noise in this example was generated by a loudspeaker inside the room. As expected, very good attenuation was obtained over the frequency range from 25 to 75 Hz at the location of the error microphone (see Fig.4.a). Around 42 Hz, an attenuation of more than 30 dB can be seen. However, because the error microphone was very close to the control loudspeaker the near field dominated the sound field and significant global attenuation was not achieved (see Fig.4.b). This is because in the near field the pressure and particle velocity are in quadrature and the sound power radiated is not necessarily reduced by minimising the sound pressure at the error microphone. In fact, it is the radiated sound power that affects the global control result and this is best minimised by an error microphone further from the control loudspeaker. It is also important that the error microphone is not too far so that the reverberant field dominates the measured sound field. The following examples illustrate these points.

In the second example, the error microphone was moved 28 cm away from the control loudspeaker. Figure 5 shows the open-loop frequency response of the uncompensated system.

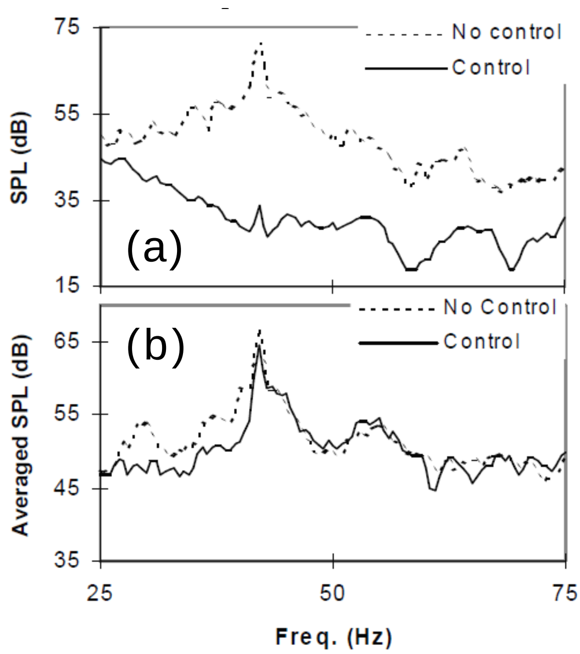

Figure 4 Control results with the error microphone 13 cm from the control loudspeaker. (a) SPL at the error microphone, (b) Averaged SPL in the room.

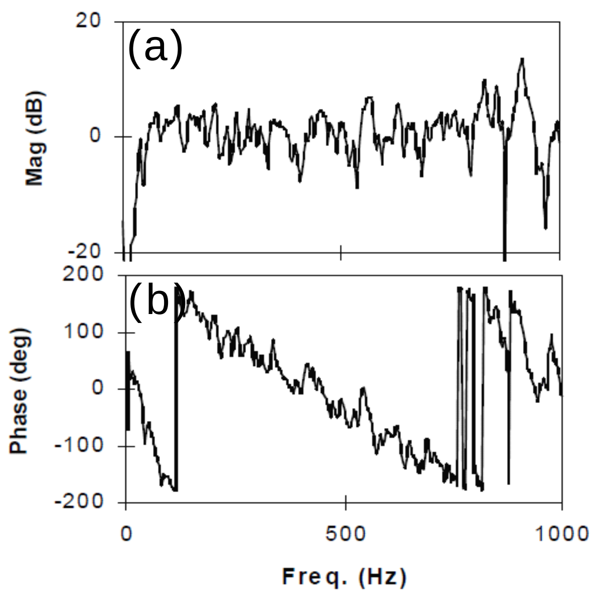

Figure 5 Magnitude (a) and phase (b) of the measured open-loop frequency response of the uncompensated system in Example 2.

It can be seen that the frequency span between the phase cross-overs (30 and 380 Hz) becomes smaller but nevertheless is still wide enough to accommodate a compensator consisting of two second order filters. Thus, good attenuation of more than 10 dB was still obtained over the frequency range from 30 to 65 Hz at the location of the error microphone (see Fig.6.a). As the error microphone was now further away from the control loudspeaker, the near field played little part in the sound field at the error microphone. In this case the radiated sound power is minimised. Consequently, global attenuation was obtained over the frequency range from 30 to 45 Hz (see Fig.6.b). Around 42 Hz, a global attenuation of more than 10 dB was achieved. Figure 7 shows the control result using the same configuration but with the primary noise coming from the nearby workshop through the closed window. Again, good attenuation was obtained at the location of the error microphone and some global attenuation was achieved below 45 Hz.

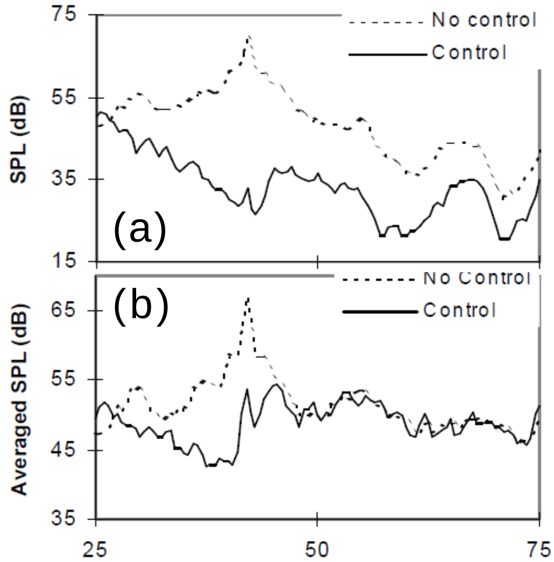

Figure 6 Control results with the error microphone 28 cm from the control loudspeaker. (a) SPL at the error microphone, (b) Averaged SPL in the room.

Moving the error microphone further away from the control loudspeaker can possibly extend the bandwidth of global attenuation to a higher frequency. However, this extension is limited by two factors.

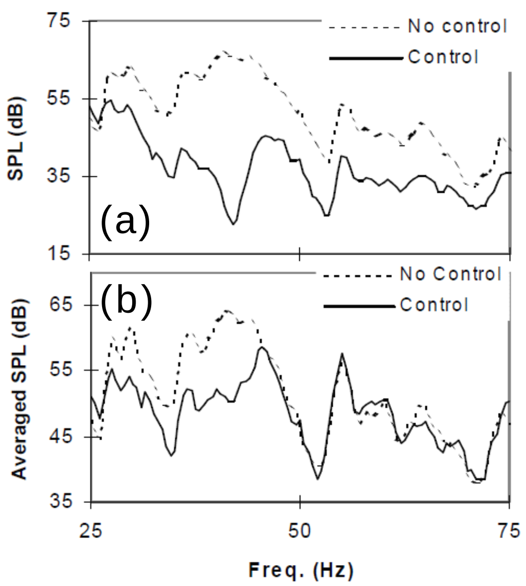

Figure 7 Control results with the error microphone 28 cm from the control loudspeaker and the primary noise from the nearby workshop. (a) SPL at the error microphone, (b) Averaged SPL in the room.

First, the bandwidth is confined by the acoustic characteristics of the room (e.g. modal overlap). In this particular case, the upper frequency limit of global attenuation achievable with a single control source is about 50 Hz. Secondly, as the error microphone is located further away from the control loudspeaker, the stable bandwidth (the frequency span between phase cross-overs) of the uncompensated system decreases. This will reduce the margin for the compensator design thereby limiting the control bandwidth of the compensator and its achievable attenuation as well. This can be illustrated by a last example.

In this example, the error microphone was located 170 cm away from the control loudspeaker, thereby eliminating the near field effect of the loudspeaker on the control. Figure 8 shows the open-loop frequency response function of the uncompensated system.

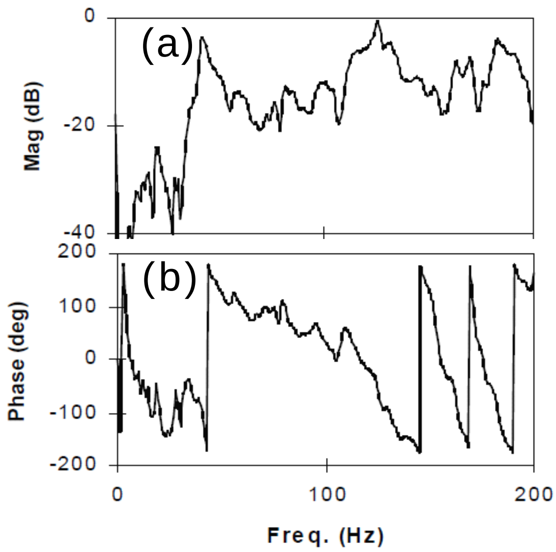

Figure 8 Magnitude (a) and phase (b) of the measured open-loop frequency response of the uncompensated system in Example 3.

It can be seen that the frequency span between the phase cross-overs (7 and 104 Hz) became smaller. This greatly reduced the margin for the compensator design. In this case, the order of compensator had to be limited to two to have a reasonable result.

Figure 9 shows the control result obtained from the 2nd order compensator design. It is clear that the bandwidth and the amount of attenuation were greatly reduced at the location of the error microphone, compared with Figs. 4.a and 6.a. However, as far as global attenuation is concerned, the result was still significant. The bandwidth and the amount of attenuation were quite similar to those at the location of the error microphone. Indeed, some global attenuation can now be seen between 40 and 50 Hz. The lack of global attenuation at lower frequencies is clearly due to the fact that the bandwidth of the uncompensated system is not wide enough. Between 50 to 60 Hz an increase of noise level can be noticed. This can be expected from practical feedback control systems around the phase cross-over frequencies of the compensated system as a direct result of Bode's integral theorem and spillover [18]. The increase of noise can be reduced by reducing the overall gain in the open-loop frequency response. However, this will also reduce the peak noise reduction as well as the control bandwidth.

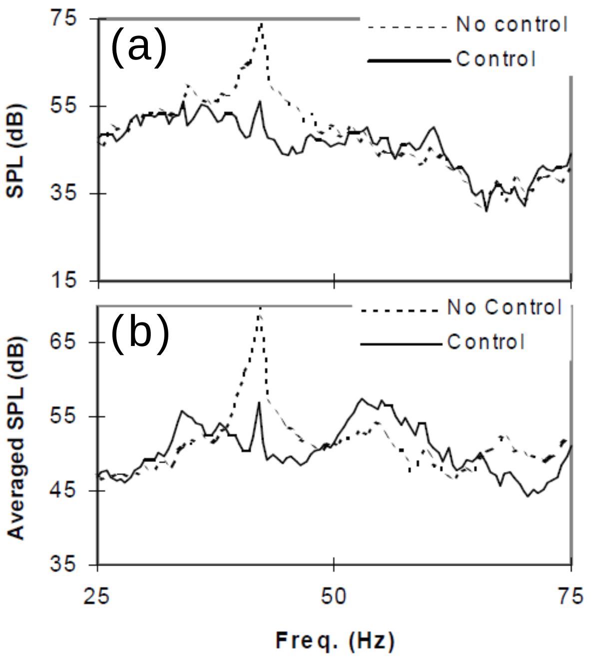

Figure 9 Control results with the error microphone 170 cm from the control loudspeaker. (a) SPL at the error microphone, (b) Averaged SPL in the room.

3. CONCLUSIONS

The single channel feedback control system presented in this paper can achieve a global noise reduction of the order of 10 dB. However, the control bandwidth depends on the relative position between the error microphone and control loudspeaker.

When the error microphone was placed 13 cm away from the loudspeaker significant global noise reduction was not achieved. This is despite excellent local control at the error microphone. In this case the near field dominates the sound field and minimising the sound pressure did not lead to attenuation of the radiated sound pressure in the far field.

When the error microphone is moved 28 cm from the loudspeaker the effect of the near field on the sound field is reduced. The radiated sound power is now minimised and both local and global control can be obtained. The peak attenuation at 42 Hz was about 10 dB and the control bandwidth between 30 to 45 Hz.

Finally when the error microphone is moved further away at 170 cm from the control loudspeaker the effect of the near field is eliminated. A peak global reduction of about 10 dB is achieved and the control bandwidth reduced to between 40 and 50 Hz with small increases adjacent to this band.

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News

IAC Colpro now a sustaining member

IAC (Industrial Acoustic Company) was founded in 1949 and has become a major international provider of noise and acoustic control products. Based in Sydney, IAC Colpro is a 100% owned subsidiary that provides the full range of IAC products and services. They include audiometric rooms, acoustic doors, acoustic enclosures and barriers, HVAC acoustic louvres and silencers, and engine and gas turbine exhaust and air inlet silencers.

For more information contact IAC Colpro, 156 Bungaree Road, Pendle Hill NSW 2145. Tel: 02 9896 0422 Fax: 02 9896 0482

Web: www.colpro.com.au

ICA Early Career Award

The ICA has called for nominations for the prestigious ICA Early Career Award to be presented at the ICA 2010 in Sydney.

This award is presented to an individual who is relatively early in his/her professional career (about 10-15 years of active career), who has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics or both and who has been active in the affairs of Acoustics through his/her National Society, other National Society(ies), Regional or International organisations.

The award consists of an award certificate, medal and an honorarium. The honorarium is determined by the Board, and for the 2010 Congress is Euro 1,000 plus up to Euro 1,000 for travel to the congress. More details are available from the ICA webpage. The deadline for submission is 21 December

Possible Acoustic Relief for Restaurant Patrons

The Phonographic Performance Company of Australia has recently announced a proposal to change the fees paid by restaurants and cafes for background music. Currently the annual licence fee is $62 a year for a restaurant or café with a seating capacity of 60. The new fee structure is based on the average costs of meals, the number of meal sessions, and also the seating capacity. A figure over $15,000 a year is thus quoted for a moderately priced licensed restaurant of 120 seats open for 3 meals a day. The new tariffs may make the cost of playing music in restaurants and cafes prohibitively expensive.

Readers who enjoy a quiet discussion or chat over a meal will probably not be dismayed if the proposed licensing changes do indeed reduce the amount of background music played in restaurants and cafes.

see http://www.ppca.com.au/documents/PPCAFactSheet090511.pdf

New distributor for Norsonic

Belcur Pty Ltd has now been appointed the sole Australian distributor for Norsonic Innovative Sound Instrumentation from 1 July 2009. Norsonic is one of the world's leading manufacturers of precision measurement instruments for sound and vibration applications. For more than 40 years the specifications for their successful instruments have been based on the requirements of regulations found in EU North America, Australasia and other industralised countries.

Norsonic have a policy of listening to their customers and implementing their needs. The Norsonic retrofit policy is also a fundamental part of their business concept. Most of their instruments are of modular design. Consequently the instrument can be easily updated by a local service centre or the factory if required by new standards or new technology. Hence, early customers can have an instrument as modern as our newest customers.

Belcur representatives have fifteen years experience in acoustics and vibration; their mission is to provide the highest possible standard of service and support for existing and future Norsonic customers.

Norsonic Instruments are backed by a 3 Year Warranty.

More information www.norsonic.com

or contact tel: 07 3820 2488

email : belcur@optusnet.com.au

NSW Construction Noise Guideline

The *Interim Construction Noise Guideline* was released on 21 July 2009 for use in New South Wales. The Guideline deals with the assessment of noise from construction activities and advises on best practice approaches to minimise noise impacts. It is specifically aimed at managing noise from construction works regulated by the NSW Department of Environment and Climate Change (DECC) and will be used to assist in setting statutory conditions in licences or other regulatory instruments.

Some councils have their own policy to manage construction noise. Other councils that do not have the resources to develop policy often seek guidance from DECC. The Guideline will assist these councils in their decision-making on construction approvals. The Guideline will be reviewed after three years to ensure that it continues to meet industry, government and community needs.

DECC would like to acknowledge the contribution of AAS members, Stuart McLachlan, Roger Treagus and Chris Schulten in developing the guideline.

The Interim Construction Noise Guideline can be downloaded from www.environment.nsw.gov.au/noise/constructnoise.htm. For a hard copy contact DECC's Environment Line on 131 555.

SAI Global Award for Heggies

At a recent gala event held at the Rydges Hotel in Brisbane, SAI Global announced Heggies Pty Ltd (Heggies) as the joint winner of the 2009 National Quality Assurance Excellence Award for companies with 50 to 500 staff. Managing Director, Richard Heggie said that the company was delighted to receive such high recognition, which demonstrates the robustness and maturity of its Quality Assurance system, processes and procedures.

Heggies consultants provide specialist engineering and scientific solutions for improving and sustaining the environment. The company has been operating since 1978 and its paper-less ISO9001 QA System was first accredited by SAI Global in 1990. The system is closely integrated with all facets of the business. With 130 staff members and nine offices across Australia and Southeast Asia, Heggies chose to develop a fast, reliable and accessible system, using its own Intranet as the document/records management and process delivery vehicle.

Thermo Fisher Scientific acquires Biolab Pty Ltd

Biolab is a leading and largest supplier of scientific, environmental and healthcare instruments and consumables in Australasia. Biolab was previously known as Davidson and carries a wide range of sound and vibration measuring instrumentation. Fisher Scientific is part of Thermo Fisher Scientific, and is a leading global channel for a complete range of laboratory research, healthcare and safety products and services. This acquisition will enable customers throughout Australasia to benefit from the highly complementary product offering, services and convenience that each brings to this transaction. The result is an even broader spectrum of product and service offerings

Further information from industrial@biolab.com.au and www.davidson.com.au

RSA merges with Heggies

Heggies Pty Ltd (Heggies) is proud to announce the recent completion of the merger of RSA Acoustics to its Sydney operations. RSA Acoustics commenced operating in 1989 and has been providing acoustic consultancy services to industrial, commercial and residential clients for 20 years. RSA Acoustics' founder, Rodney Stevens, has been an acoustic practitioner for over 30 years, commencing the early part of his career with the NSW State Pollution Control Commission (now NSW DECC). Rodney is a member of the Australian Acoustical Society (AAS) and is the Treasurer for the NSW Divisional Committee. Rodney and his team moved into Heggies' Sydney Office immediately post-acquisition, working closely with all three of Heggies' Sydney based Noise and Vibration Division Managers.

Prizes for best papers at AAS 2009

There will be four $250 prizes for best papers presented at this year's AAS conference in Adelaide. They will be awarded in the following areas:

- Underwater Acoustics
- Environmental Acoustics
- Industrial Acoustics
- Research

Some electric cars will be noisy

Apart from their environmental advantages, electric cars are also wonderfully quiet.

However it seems that noise, and plenty of it, is an essential part of the motoring experience for some drivers, particularly those who own high performance or executive cars. Consequently Brabus, a German automotive performance specialist, has developed an electronic system to give electric cars, such as the Tesla, a more exciting sound using an amplifier and multiple speakers. The occupants on-board the Brabus Tesla Roadsters can choose from several simulated engine sounds including that of a typical V8 combustion engine, a racecar engine and two futuristic soundscapes named 'Beam' and 'Warp.' The volume of the sound is dependent on the momentary power output of the electric motor. See www.brabus.com/index_en.html

One positive outcome might be improved pedestrian safety. Electric cars can be so quiet that they pose a danger to visually-impaired pedestrians. They can also be a problem to pedestrians who have deliberately chosen to impair their hearing by using mp3 players or mobile phones. Indeed a recent paper presented at the 157th ASA Meeting in Portland in May confirmed that it took longer to determine the direction of approach of an electric car compared to a conventional engine – see www.acoustics.org/press/157th/rosenblum.html

Study confirms problems with open plan offices

In recent years there has been an increasing trend to use cheaper open plan offices instead of private, closed offices. However a recent review article by Vinesh Oommen, Mike Knowles and Isabella Zhao published in the Asia-Pacific Journal of Health Management confirms their negative effects on employees. As might be expected the high noise levels and lack of acoustic privacy play a major role in employee's dissatisfaction.

Noise Guide for Local Government

DECCW (The Department of Environment, Climate Change and Water) have now released the new Part 2 of the Noise Guide for Local Government. This Part provides guidance on how to assess noise impacts and includes advice on how to measure noise levels.

The revised Part 2:

- Explains the three tests associated with noise pollution, the audibility test, the duration of noise test and the offensive noise test,
- Gives more advice about conducting the offensive noise test and gives examples of offensive noise,
- Provides a clearer explanation of intrusive noise and the difference between intrusive and offensive noise,
- Indicates how noise levels can be defined for Noise Control Notices and Prevention Notices,
- Describes in greater detail both the techniques and the traps in measuring noise and the correct sequence of steps for measurements in the field,
- Gives advice on the use and suitability of specific noise descriptors with an associated case study for illustration.

The new Part 2 can be downloaded from the DECCW website at http://www.environment.nsw.gov.au/noise/nglg.htm

A new Part 4 on "Regulating Noise Impacts" was posted to the "noise" page of the DECCW website in March 2009.

Any questions about the Guide may be directed to Roger Treagus at DECCW on (02) 9995-5784

Bassett rebranded as AECOM

Bassett Acoustics in Australia has recently been rebranded as AECOM. Bassett has been part of AECOM since October 2004.

AECOM is a global provider of professional technical and management support services to a broad range of markets, including transportation, facilities, environmental and energy. With 43,000 employees around the world, AECOM is a leader in all of the key markets that it serves. AECOM provides a blend of global reach, local knowledge, innovation, and technical excellence in delivering solutions that enhance and sustain the world's built, natural, and social environments. A Fortune 500 company, AECOM serves clients in more than 100 countries. More information can be found at www.aecom.com or for additional information on the AECOM acoustic group in Australia contact matthew.stead@aecom.com

Heggies move in Perth and Townsville

As part of their continuing development of services in WA and North QLD, and ongoing commitment to clients, Heggies have moved to larger premises in Perth and Townsville. The new contact details are: 125 Edward Street Perth WA 6000. Phone: 08 93700100

Unit 5, 286 Ross River Road Aitkenvale QLD 4814. Phone: 07 4725 2119

Pyrotek Soundguard helps reduce noise in schools

Pyrotek Soundguard now offers a "Kill noise in schools" information pack. A recent example involves their installation of customised acoustic panels in the art and music rooms of the Lecceto Arts Centre in St Augustine's College at Brookvale. The panels used Sorberfoam acoustic foam mounted within a timber frame before laminating with an acoustic membrane of Sorbertex P52, and reduced sound levels that had previously raised OHS concerns. It is hoped that some of Prime Minister Rudd's promised $14.7 billion in school upgrades will be spent in similar improvements to students' acoustic environments.

Pyrotek Soundguard offers their "Kill noise in schools" info pack by calling 1300 136 662 or via www.soundguard.com.au/pack.htm.

New Products

Pulsar Quantifier Sound Level Meters

Pulsar Instruments announce their 'Quantifier' range of sound level meters that are designed as practical and easy-to-use sound level meters for industrial, environmental & general noise measurements. The models 91 and 92 are simple to use and suit a wide and varied range of applications. With 1:1 octave band filters, the models 93 and 94 are the complete solution for noise at work measurements and the models 95 and 96 are perfect for comprehensive industrial and environmental noise measurements with 1:1 & 1:3 octave band filters. 'Analyser' software is provided as standard with the entire range. The measurement kit, includes everything in a high quality carry case giving the confidence to know one has everything required to carry out risk assessments.

Information www.soundlevelmeters.net.au or bryan@soundlevelmeters.net.au

Rion NL27 Sound Level Meter

Rion's latest offering is the NL-27 compact sound level meter. Compliant with class 2 specifications according to IEC 61672-1:

2002 Class 2 and featuring a wide 107dB dynamic range. Pocket sized with easy measurement of SPL, Leq, Lmax, LE in A or C weighting and LCPeak this meter is ideal for the occupational health and safety professional.

Information Morgan at Acoustic Research Laboratories on 02 9484 0800 or www.acousticresearch.com.au

Random Incidence Microphone with Extended Frequency Capability

The Industrial Technologies division of Biolab (Aust) Pty Ltd together with PCB Piezotronics, Inc, introduce model 377A21, a prepolarized, 1/2" random incidence (RI) response microphone. Operating from ICP sensor power, this new microphone features an extended frequency capability.

Most half inch RI response microphones are limited to a maximum frequency of 12 kHz and can not be used in applications where the full audible range was required. PCB® model 377A21 has a linear frequency range of 4 Hz to 20,000 kHz (+/- 2 dB.) that eliminates this limitation. The 377A21 is specifically designed for applications where the user is looking to measure diffuse sounds, or the total sound coming from different locations. The 3% distortion factor of this microphone, which is the maximum amplitude that a microphone can measure and still be linear, was increased from 146 dB (re 20µPa.) to 155 dB. The increased amplitude, in combination with the low noise floor rating of 20 dB(A), make this microphone an excellent choice for specific aerospace, automotive and military applications.

Information, from Biolab (Aust) Pty Ltd, tel 1-300-SENSOR (736 767) www.biolab.com.au or www.davidson.com.au

Brüel & Kjær announce PULSE Reflex

PULSE Reflex is designed to be a major evolutionary step in the development of a most popular sound and vibration platform. And with its new applications – Core, Modal Analysis and Building Acoustics – PULSE Reflex extends the measurement and analysis chain to include world-class post-processing. PULSE Reflex has an intuitive user interface, powerful data management and enhanced reporting opportunities to increase productivity. Designed in close cooperation with end-users from diverse sectors of industry across the world, PULSE Reflex has ease of use and high productivity concepts at its core. A new graphical user interface optimises the workflow, enables data sharing across applications, visualises results quickly, and create high quality reports with ease.

Related operations are grouped together in dedicated regions of the screen and the user controls whether they are visible or hidden to give maximum focus and screen space for whatever task is performed. PULSE Reflex provides a versatile, but simple-to-use notes section that enables the user to record every thought as it occurs. Additionally, it is possible to include sounds, photographs, or videos and link them to items in the project to provide a lasting reminder of how or why critical decisions were made.

Information from www.bksv.com/pulsereflex or Brüel & Kjær Australia at auinfo@bksv.com or 02 9889 8888.

Meeting Reports

VICTORIA DIVISION

The two most recent technical meetings held in 2009 by the Victoria Division were largely concerned with aspects of monitoring and evaluating aircraft noise.

Monitoring environmental noise

At the earlier meeting, held on March 21 in the SKM theatrette, Armadale, at which 9 were present, Nick Prgomelja, an Application Engineer of Brüel & Kjær Australia, spoke about the most recently available B & K instruments and systems for logging real time sound levels of environmental noise. Such environmental noise, while including aircraft noise, also includes industrial, railway, road traffic and entertainment venue noise.

This B & K Environmental Noise Management [ENM] system, type 7843, allows the noise measuring monitor to be left unattended in its monitoring location or station, and controlled from a remote office or laboratory location, where also the noise signals are received, further analyzed, and evaluated.

The B & K type 7843 system comprises two units, the ENM Server 7843A located alongside the noise measuring terminal, and the ENM Client 7843C at the operator's remote station, the two being joined by phone line or similar means to provide both a one-way http connexion for relaying the real time noise data to the ENM Client, and a two-way net.tcp connexion to enable the remote "client" to control the operation of the noise monitor. With the monitoring of noise from entertainment venues, there is the added facility of being able to directly control the level of the amplified sound so that it does not exceed a predetermined limit.

The monitoring microphone is directly connected into the system via a type 2250 or 2270 hand-held analyzer, which is installed within a type 3639A/B or 3639E/G noise monitoring terminal [NMT], the 3639A/B unit being designed to contain a type 2250 analyzer, and the 3639E/G unit a type 2270 analyzer.

The type 2250 analyzer, for example,

[a] is equipped with a large high-resolution, touch-sensitive color screen, automatic detection of and correction for a windscreen, and for use with a variety of software modules,

[b] provides for data storage on plug-in memory cards [of capacity at least 128MB], and a 3 Hz to 20 kHz broadband linear frequency range, and

[c] enables operations such as real-time frequency analysis in 1/1- or 1/3-octave bands, the logging of broadband and spectral data to obtain a time history for subsequent analysis, the sound recording of a measured signal during all or parts of a measurement, and can document measurements using text and voice annotations.

The type 2270 analyzer includes several additional features, including dual-channel measurement, an integrated digital camera for reporting, timers for automatic start of measurement, CCLD input for vibration or sound measurement, the making of building acoustic measurements, and FFT analysis of sound and vibration.

This brief description of the type 7843 ENM system is of its basic principles, characteristics and uses. Further detail needs to be obtained from the B & K Product Data sheets and via the B & K website.

At the end of this presentation, the chairman thanked Nick Prgomelja for his interesting and informative talk, the thanks being carried by applause from those present.

Managing aircraft noise

At the later meeting, held on July 22 at the offices of Lochard, Caulfield North, at which 13 were present, Mike Rikard-Bell, engineer, and a co-founder of Lochard, described aspects of Global best practice in airport noise management. He stated too that Lochard is now part of the B & K group of companies.

Environmental noise management at airports has mostly followed the conventional wisdom that because annoyance varies directly with noise exposure, less noise means less annoyance. It obviously involves objective considerations including quieter aircraft, noise monitoring, night time noise curfews, changed landing procedures such as the continuous descent approach, fines for aircraft exceeding specified noise limits, and appropriate land use noise zoning around airports.

These measures have all been implemented – examples were given – but with only moderate success. The numbers of noise complaints, instead of decreasing, have actually increased. Human psychology must therefore be more seriously taken into account.

Analyzing the nature of these complaints has shown, for example, that while complaints

ICA 2010

This is the twelfth in a series of regular items in the lead up to ICA in Sydney in August 2010.

The ICA 2010 and associated meetings are just a year away, 23-27 August 2010, and so there is much work to do from now. The web page is being updated and the registration form and call for papers will be available soon. The ICA 2010 incorporates the AAS 2010 annual conference and provides the opportunity to showcase the work that is being undertaken in Australia on the wide range of acoustics and vibration topics.

In conjunction with the main congress of the ICA there will be four associated meetings in the week following:

- International Symposium on Musical Acoustics will overlap with the main congress and follow on for specialist sessions in the Blue Mountains, near to Sydney,
- International Symposium on Room Acoustics to be held in Melbourne from 29 August
- Acoustics and Sustainability to be held in Auckland, New Zealand from 29 August
- Non Linear Acoustics and Vibration to be held in Singapore from 30 August

Some deadlines for the main ICA congress are:

Deadline for receipt of abstracts	22 February 2010
Advice of acceptance of abstract	19 March 2010
Deadline for early bird registration fee	21 May 2010
Deadline for full papers	30 May 2010
Program schedule available	21 June 2010
ICA 2010	23-27 August 2010

The web page will be the main source of information on the ICA and the associated meetings www.ica2010sydney.org. As well as planning to attend this outstanding congress we ask that all members of the AAS assist with the promotion among your national and international colleagues.

Marion Burgess, Chair ICA 2010

from those beneath flight paths are decreasing, they are increasing beside flight paths and because of off-track flights. Noise has also been identified as a proxy for more general concerns such as adverse health effects from poor air quality. An NAL study in the 1980s resulting in two closely similar curves of the percent highly annoyed for various average Day-Night sound levels from 40 to 100 dB[A], showed that this rose from 0% at 40 dB[A] to 10% at 65 dB[A], and then rose almost uniformly to 80% at 100 dB[A]. Further study revealed that only 13% of this variability was explained by the factor of noise exposure, with 60% by other factors.

However, while in the past the prime focus in designing airports has been on their physical capacity, the environmental capacity is only now coming to be recognized as a most important and often limiting factor. Thus has been developed Lochard's "Law" of environmental capacity,

$$E = T - I,$$

where E = environmental capacity, T = community tolerance, and I = environmental impact.

Environmental capacity is generated by

[a] reducing environmental impact by proactive management of operations, and

[b] increasing community tolerance through collaboration with all interested parties.

This requires a balanced program of

[a] Impact reduction initiatives such as replacing noisy aircraft with newer, quieter craft, adherence at airports to specified landing, take-off and engine maintenance and testing procedures, pilot working groups, noise budgets, surcharges, curfews and access limits, and

[b] Tolerance building initiatives such as noise and information sharing, community forums, self-service on-line interaction, roundtable discussions and community education.

Four generations of environmental management were identified - as Ignorance, Measurement, Abatement and Environmental Capacity, the corresponding degrees of maturity being passive, reactive, proactive and collaborative. At present, most airport managements are at stages two or three.

There is also a second law - Lochard's "Law" of Annoyance,

$$A = P - X,$$

where A = annoyance, P = perception and X = expectation.

Much, therefore, needs to be done in setting people's expectations and calibrating their perceptions. There's much more to annoyance than noise dose response.

The way forward now is to;

[a] continue to reduce noise exposure [limited by the law of diminishing returns],

[b] work hard on setting expectations,

[c] work even harder on calibrating perceptions,

[d] do it early and often.

In the ensuing discussion, the matters raised included;

[a] problems with land developers,

[b] noise complaints being directed to airport authorities instead of developers,

[c] the influence of continuous monitoring in increasing community tolerance, and identifying noise sources,

[d] distinguishing aircraft noise which is around only 5 dB[A] above background is now more complex than it was when the difference was 10 to 20 dB[A],

[e] engine testing involved a low frequency noise problem causing chest pain because of a body resonance at 31 Hz.

At the close of this meeting, Norm Broner, on behalf of all present, thanked Mike Rikard-Bell for his most interesting and informative talk and discussion, with applause from those present.

Invitation

To advance the science and art of Acoustics, the AAS Victoria Division invites consultants, manufacturers and agents to use, from time to time, the Society's regular technical meetings as a forum for describing their acoustical work, or the acoustical properties, characteristics and uses of their products, equipment and instruments. The Victoria Division secretary may be contacted by mail at c/o N Broner, SKM, 590 Orrong Rd, Armadale, Vic, 3143, by email at a.robinson@marshallday.com.au or by telefone at [03] 9416 1855.

Louis Fouvy

Future Meetings

AAS Annual Conference 2009

This will be held in Adelaide, 23-25 November 2009, with the theme "Research to Consulting". Keynote speakers include Dr David Rennison, Prof Jie Pan and Dr Brian Ferguson. Papers on all topics in acoustics are welcome and the submission details available from the website. A technical exhibition will be open for the duration of the conference. The social program will include a welcome function, dinner and farewell meal.

For more information on the conference follow the links from AAS webpage www.acoustics.asn.au/joomla/acoustics-2009.html

FASTS

Science Governance

The Federation of Australian Scientific and Technological Societies (FASTS), of which the AAS is a member, will be releasing a paper on the Governance of Science by Tony Coyle. This paper poses the question : are the current informal governance of science (professional societies, peer networks) are adequate to deal with complex 'wicked' problems? FASTS will be holding a workshop on this topic in Sept/Oct and any AAS members who may be interested in attending should advise the AAS General Secretary.

Women in Science

On September 12th FASTS is holding a workshop at Parliament House to consider the final recommendations in a new paper on Women in Science, commissioned from Sharon Bell. A key theme is the loss of productivity from systematic under-representation of women in the scientific workforce in industry and academia. Any AAS members interested in attending should advise the AAS General Secretary.

Preparedness R&D

FASTS will soon release a revised discussion paper by Mark Matthews on preparedness R&D. This is a follow up to the influential FASTS paper that was picked up by the Productivity Commission in its report on public support for science and innovation. This update will argue for an explicit treatment of 'preparedness' in the Govt's post-Cutler review policy framework and will also consider the human capital dimensions of 'preparedness R&D'.

Changing nature of science

This is an emerging project FASTS is undertaking jointly with the deans of science. The impetus is dissatisfaction with much of the commentary on skills shortages in science and technology. FASTS felt a key problem with the skills shortage literature was a poorly articulated sense of how the nature of the technical workforce has evolved over the last 30-50 years in the face of the IT revolution and globalisation. It is worth exploring the question of what role technically trained people now play in a developed economy. How have the required skill sets and roles changed over time? Do developed economies need fewer science trained people but in higher level and more specialised areas? How has instrumentation changed the productivity of science? How do employers now identify and describe the kind of science background that they require of their recruits, and the roles that they have in mind for them? How has this changed over time? The FASTS/ ACDS workshop is intended to open up a more sophisticated and generative way to think about skills and skill requirements. It is intended that a key outcome of the workshop will be terms of reference for one or more well defined research programs/projects to advance these issues. AAS members interested in attending should advise the AAS General Secretary

Risky Business

The following opinion piece from the FASTS website is by Ken Baldwin, President of FASTS. It was published as 'Risky Business' in Campus Review on 6 July 2009.

Last year, the Federation of Australian Scientific and Technological Societies (FASTS) and the Australian National University jointly held a forum on risk-aware research. (See the report in the August 2008 issue of Acoustics Australia. Ed)

Our starting point was a conviction that policy makers and the research sectors needed to ensure that investment encourages risk-aware, not risk-averse, research so as to provide the robustness and flexibility to meet future threats and opportunities.

It immediately became apparent that risk in research had many definitions. We therefore developed – perhaps for the first time – this 'taxonomy of risk' to provide a common starting point for discussion:

- "Transformative" risk: the risk that new ideas which might transform the way we think might be delayed or not accepted because they violate existing interests or views.
- "Capacity" risk: research capability or funding which is insufficient to perform the required task – for example, by not fully funding the true costs of research.
- "Failure" risk: the risk that the research might fail to meet a particular objective, although this is not a real 'failure' as it can provide useful information on other avenues for research.
- "Collaborative" risk: the risk that research is constrained by 'safe' choices of collaborators – a very real issue in multi-disciplinary research.
- "Precedence" risk: being beaten to the punch.
- "Regulatory" risk: the risk (particularly to institutions) that researchers transgress ethical, regulatory and commercial constraints on research.

The forum focused primarily on transformative risk as the key challenge to be addressed by research funding programmes.

However, it is generally recognised that the peer review process is often risk-averse when it comes to evaluating transformative research. For example, a recent US National Science foundation (NSF) report notes that transformative research does not "fare well wherever a review system is dominated by experts highly invested in current paradigms or during times of especially limited budgets that promote aversion to risk."

To counter the conservatism of peer review, investigators can game the situation by putting forward "safe" proposals that are incremental extensions of existing work, and for which they may have already achieved results. The peer review process might look favourably on their track record, view the chances of success of their conservative approach as high, and recommend funding. Investigators then take the opportunity to pursue more speculative research, already having the incremental results "in the bag".

This gaming of the system mitigates against the conservative nature of peer review by unwittingly providing researchers with in-built agility. However, allowing perverse behaviour to counter inherent shortcomings is bad policy.

So how can risk-aware research be encouraged? The forum identified a number of key factors, including:

- Aggregation: It is critical to recognize that risk cannot easily be borne at the individual project grant level, particularly in tight funding environments. Larger research programmes such as ARC Centres of Excellence or NHMRC program grants can provide the flexibility and agility needed to encourage transformative risk by allowing discretionary funds for risky projects.
- Diversity: We need an explicit portfolio approach which provides a spectrum of funding bodies and programs with qualitatively different risk profiles, objectives and evaluation measures.

  A plethora of small programs clumping on similar selection criteria and objectives does not provide diversity but rather adds opportunity costs and other inefficiencies.
- Flexibility: It is unwise to impose narrow and inflexible objectives. Programs need to ensure that funding rules allow research to change direction if the need arises. This is sometimes achieved by not requiring a project to report against the original objectives, thereby encouraging researchers to grasp opportunities and move in new directions.
- Rewards: Research contracts should encourage the handing back of funds in cases where research reaches a dead end. This should be treated as good professional practice, where favourable consideration is given to providing additional funds for future successful applications by the same investigators.

The global financial crisis has shown that we need to understand and evaluate risk in order to create a robust economic system. Similarly in research, we need to provide the agility to create flexibility in research funding programmes so that we can encourage transformative risk and enable science to develop new frontiers.

For example, US agencies like DARPA (Defence Advanced Research Projects Agency) and its recently established Intelligence sibling IARPA, fund high-risk research aimed at identifying potentially disruptive threats and challenges. The NSF SGER fund (Small Grants for Exploratory Research) is also aimed at testing new, high-risk ideas and is particularly useful for early career researchers.

In this sense, programmes that encourage risk in research are as much about developing human capacity to push forward the boundaries of knowledge as they are about the research outputs themselves. International experience in encouraging risk-aware research – in Australia, the US and elsewhere – can help scientists and engineers investigate bold new ideas, and free us all to create an exciting future.

Book Review

"Environmental Noise Barriers – A guide to their acoustic and visual design"

2nd Edition

Ben Kotzen and Colin English

Spon Press, Taylor Francis 2009, 258 pp hard cover ISBN10:0-415-43708-3 and ebook ISBN10:0-203-93138-6

In the ideal world all environmental noise sources would be controlled at the source, but in the real world of today other methods of noise control are necessary. Barriers between noise sources and the surrounding communities are a common means of noise control. A well-designed acoustic barrier can achieve the required performance goals at a reasonable cost while providing visual and social benefits to the community. The range of practitioners involved in the process of barrier selection, design and installation includes clients, acoustical engineers, landscape architects, manufacturers as well as those in the local and central authorities responsible for highways, environment or planning.

This book, entitled Environmental Noise Barriers, is a unique one-stop reference for all involved in the barrier design process. It is an extensively revised new edition of the popular book of the same name first published in 1999. While it is primarily directed at the European market with updates in line with UK and EU legislation, it provides an international perspective of barriers for noise control.

The sections of the book deal with: Defining the need for barriers; Acoustic performance of barriers; Barrier morphology and design; Types of barriers and barrier material; Climbing plants and other plants for use on barriers; Engineering safety, environmental and cost considerations; and Contemporary issues, developments and considerations. Throughout the book are colour photographs of barriers from around the world which illustrate the range of solutions that have been developed. Some of the barriers included in the first edition have been revisited with critical comments on those that have not 'aged' as well as expected.

While the basic design requirements for a barrier are essentially unchanged, this book discusses and presents examples of barriers that do more than just meet those requirements. There are those that are particularly designed to be a visual feature in the environment. There book also discusses innovative barriers that are designed to meet a range of needs including barriers with features to reduce air pollution, barriers incorporating photovoltaic panels and barriers integrated with building design. This latter type of barrier is not just a screening building but actually incorporates building components into the barrier such as showrooms and shops.

It is the range of photographs and illustrations that makes this book a pleasure to read through. While most of the barriers discussed are in Europe there are some featured from our side of the world – the Tolo Highway in Hong Kong, and the Craigieburn Bypass in Victoria.

This book is highly recommended as a resource for all involved with the design and planning of barriers to reduce noise. As well as providing guidance on the basic design, the discussion and photographs would be valuable to use as reference for the whole team working on any development of a barrier design. It would also be an excellent reference for anyone involved in teaching noise control either in formal courses or for seminars, workshops etc on the topic.

Marion Burgess

Marion Burgess is a research officer with the Acoustics and Vibration Unit of UNSW at the Australian Defence Force Academy. She has been involved with acoustics long enough to see the barrier develop from a boring wall to the landscape features of today.

VALE CARLEEN HUTCHINS

VIOLIN MAKER EXTRAORDINARY

To most musicians there are just four violin-like instruments – violin, viola, cello, and double-bass – and there has been no change since the time of Bach. Before that there were viols, which were somewhat different in shape, size and playing technique. But will there ever be another change?

Until fifty years ago the answer seemed to be "No", but then Carleen Hutchins, a science teacher and violin maker in New York, in consultation with Harvard physicist Frederick A. Saunders, came up with the idea of a family of instruments all modelled on the proportions and acoustics of a Stradivari violin. It took ten years to realise the dream but by then Carleen had produced the eight instruments of the New Violin Family. The smallest was tuned an octave higher than a standard violin, but in order to make fingerings practical its fingerboard needed to be not much shorter so that great string tension was required. The largest was tuned to the same pitch as a standard double bass but was much larger in body size to give a very full tone. Carleen made six sets of these instruments during her lifetime and several other sets have been made by other luthiers. The New Violin Family Association http://www.newviolinfamily.org/eight.html, formerly the Catgut Acoustical Society which Carleen founded in 1963, promotes continuing interest in them, and several recordings have been produced. As well as inventing the new Violin Octet members, she is known for her work on plate tuning and other techniques for the scientific handcrafting of string instruments.

Sadly Carleen Hutchins died on Friday August 7 at the age of 98. She received many awards during her career, including a Guggenheim Fellowship and, from the Acoustical Society of America, the inaugural Silver Medal in Musical Acoustics in 1981, and an Honorary Fellowship in 1998, an honour given to only 17 people, beginning with Thomas A. Edison. She was a most pleasant and outgoing person and a great scientific luthier whose contribution to string instrument acoustics will have an enduring influence. An excellent short biography is given in the New York Times web page http://www.nytimes.com/2009/08/09/arts/music/09hutchins.html

Neville Fletcher

Sustaining Members

The following are Sustaining Members of the Australian Acoustical Society. Full contact details are available from http://www.acoustics.asn.au/sql/sustaining.php

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VIPAC ENGINEERS AND SCIENTISTS
www.vipac.com.au

Diary

2009

6 – 10 September, Brighton
Interspeech 2009
www.interspeech2009.org

19 – 23 September, Rome
IEEE 2009 Ultrasonics Symposium
pappalar@uniroma3.it

23 – 25 September, Beijing
WESPAC X: 10th Western Pacific Acoustics Conference
www.wespacx.org

23 – 25 September, Xi'an
Pacific Rim Underwater Acoustics Conference (PRUAC)
lfh@mail.ioa.ac.cn

5 – 7 October, Tallinn
International Conference on Complexity of Nonlinear Waves.
www.ioc.ee/cnw09

26 – 28 October, Edinburgh
EURONOISE 2009
"Action on Noise in Europe"
www.euronoise2009.org.uk

11 – 13 November, Miyagi, Japan
1st International Workshop on Principles and Applications of Spatial Hearing.
www.riec.tohoku.ac.jp/IWPASH/

18 – 20 November, Kyoto, Japan
30th Symposium on Ultrasonics Electronics.
www.use-jp.org/USE2009/en/index.html

23 – 25 November, Adelaide
AAS Annual Conference 'Research to Consulting'
www.acoustics.as.au

2010

06 – 09 January, Sanya, China
2nd International Conference on Vibro-Impact Systems
www.neu.edu.cn

15 – 19 March, Dallas
International Conference on Acoustics, Speech, and Signal Processing.
icassp2010.org

09 – 11 June, Aalborg
14th Conference on Low Frequency Noise and Vibration.
lf2010.org/

13 – 16 June, Lisbon
INTER–NOISE 2010
www.internoise2010.org

24 – 27 May, Sydney
OCEANS'10 IEEE
www.oceans10ieeesydney.org/

29 – 31 August, Auckland
Acoustics and Sustainability
TBA

30 – 31 August, Singapore
Non Linear Acoustics and Vibration
www.ica2010sydney.org

23 – 27 August, Seattle

23 – 27 August, Sydney
ICA2010
www.ica2010sydney.org

26 – 31 August, Sydney
ISMA 2010 International Symposium on Musical Acoustics
isma2010.phys.unsw.edu.au/

29 – 31 August, Melbourne
ISRA 2010 International Symposium on Room Acoustics
www.isra2010.org/

11th International Conference on Music Perception and Cognition
TBA

26 – 30 September, Makuhari
Interspeech 2010 - ICSLP.
www.interspeech2010.org

11 – 14 October, San Diego
IEEE 2010 Ultrasonics Symposium.
bpotter@vectron.com

19 -– 20 November, Brighton
Reproduced Sound 25
www.ica.org.uk/viewupcoming.asp

2011

22 – 25 May, Prague
International Conference on Acoustics, Speech, and Signal Processing (IEEE ICASSP 2011).
www.icassp2011.com

24 – 28 July, Tokyo
19th International Symposium on Nonlinear Acoustics (ISNA 19)
Web: TBA

27 June – 1 July, Aalborg
Forum Acusticum 2011
www.fa2011.org

27 – 31 August, Florence
Interspeech 2011
www.interspeech2011.org

05 - 08 September, Gdansk
2011 ICU International Congress on Ultrasonics.
Web: TBA

4 –7 September, Osaka
INTER-NOISE 2011
office@ince-j.or.jp

Meeting dates can change so please ensure you check the www pages. Meeting Calendars are available on http://www.icacommission.org/calendar.html

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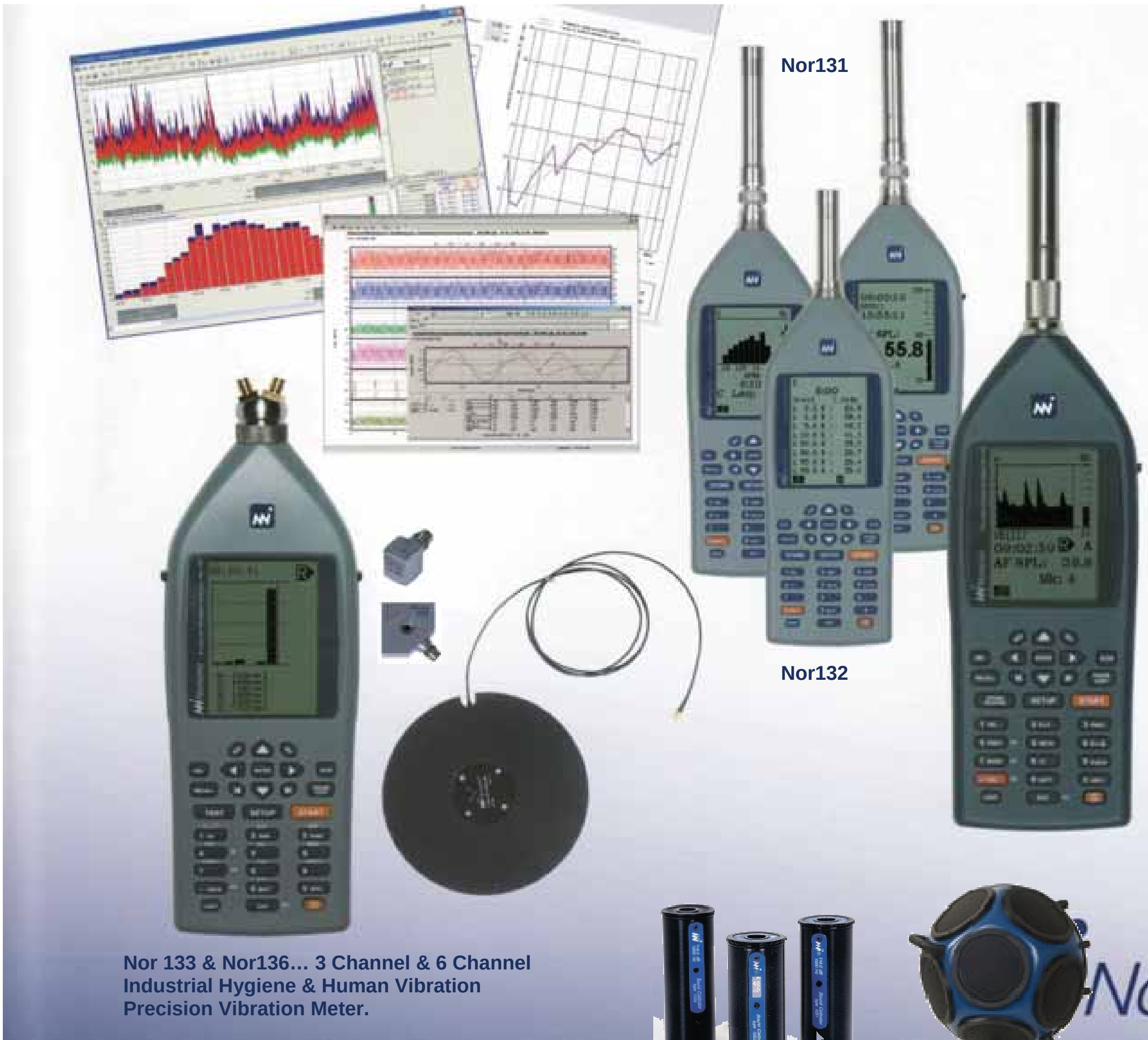

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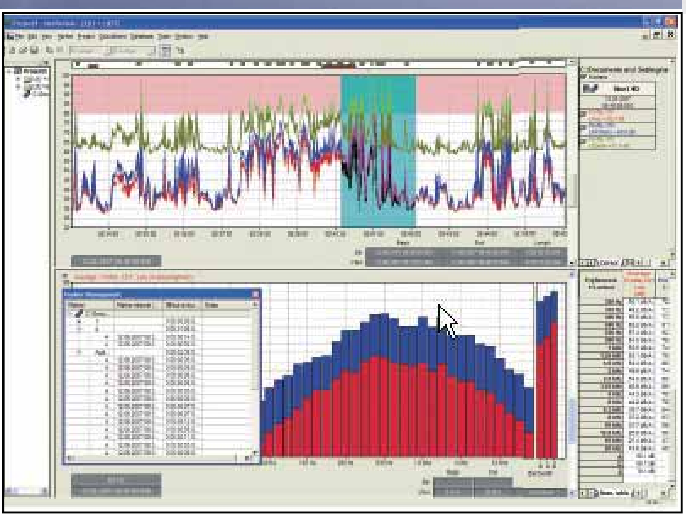

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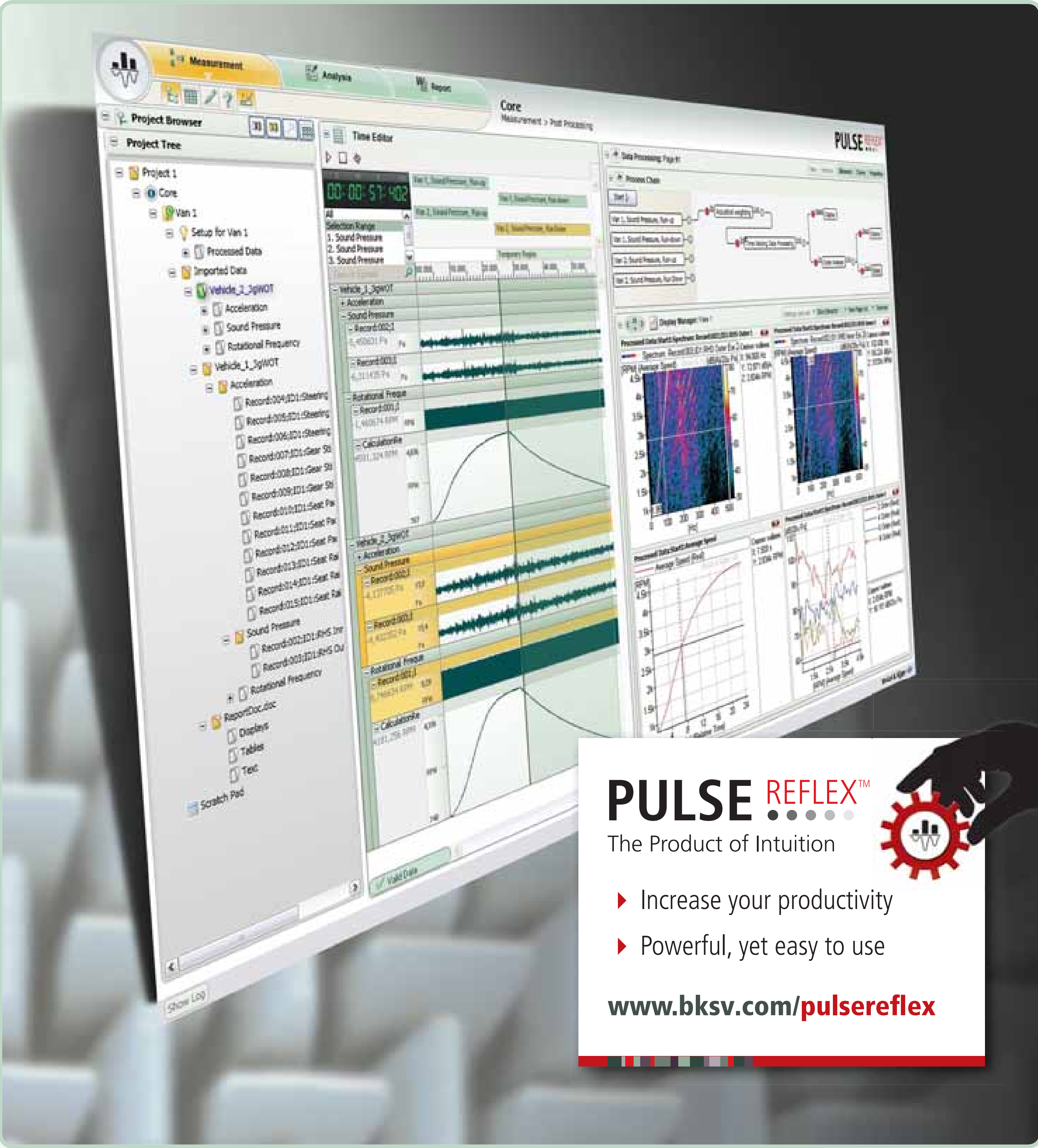

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