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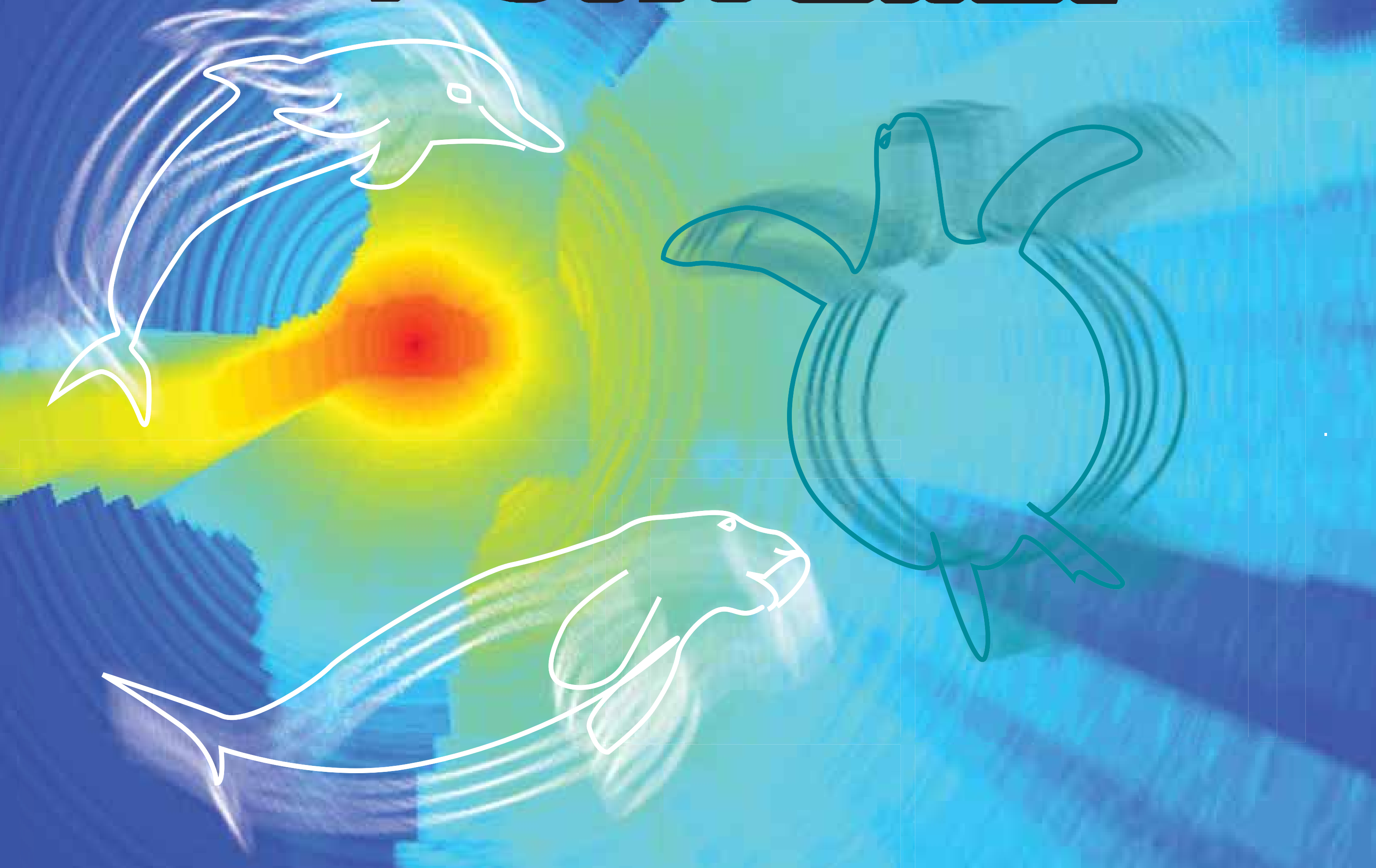

- **Driving piles underwater**
- **Duct directivity**
- **Mouth to ear response**
- **Controlling motor noise**

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PAPERS

TECHNICAL NOTE

Cover: Moreton Bay. See page 91. *Cover design: Heidi Hereth*

Message from the President

Well, it's a busy time of year! Last week I got back from the very well presented and attended Annual AAS Conference held in Adelaide and last night, I attended the Victorian Division end-of-year dinner, with a very interesting guest speaker, Dr. Robert Holmes who is a veterinarian with a PhD and Fellowship in animal behaviour. He talked about the sound produced by animals and what we can interpret from them and about the impact of sounds on animals. For more details, see Louis Fouvy's report herein.

This year's AAS Annual Conference President's Prize was awarded to: Dave Hanson, Jerome Antoni, Graham Brown, Ross Emslie "Cyclostationarity for ship detection using passive sonar: progress towards a detection and identification framework". See the Conference Proceedings for this and other good papers!

AAS President Awards Presidents Prize to Dave Hansen at AAS Annual Conference Banquet

Talking about awards at conferences, I would like to congratulate Professor Colin Hansen of the School of Mechanical Engineering at Adelaide University on the Rayleigh Medal, a medal awarded by the Institute of Acoustics, UK, for an Outstanding Contribution to Acoustics. This medal is awarded once every two years to a "foreign national" and Colin is the first Aussie to have won this Award! Well done, Colin! (See the news item later in this issue.)

We are now looking forward to next year's conference, which will be very exciting as we host the 20th ICA in Sydney at Darling Harbour. Marion Burgess, the conference chair, and her team, are very busy preparing what should turn out to be a rewarding and exciting conference. I encourage all of you to attend and present a paper (AAS members will get a special rate). We are hoping for 1000 attendees from all over the world and we would like to showcase our Australian creativity in the field. For more details, see www.ica2010sydney.org . And don't forget the associated conferences on Music Acoustics (Katoomba), Room Acoustics (Melbourne), Acoustics and Sustainability (Auckland, NZ) and Non-linear Acoustics and Vibration (Singapore).

I want to take this opportunity to congratulate Pam Gunn (see the April number) and Louis Fouvy who have been awarded fellow status of the AAS. Both have contributed significantly to acoustics in Australia and it is fitting that we should recognise their contributions in this way. We now have some 20 Fellows in our Society. If you believe that any other members should be awarded this recognition, please contact your division.

And talking about contributions to our Society, I would like to take this opportunity to thank Charles Don of our Victorian Division who has single-handedly been scanning past conference proceedings so that our members can make use of them as they are loaded onto our web page (I also want to thank Sheena for allowing Charles to clutter their house while doing this). In addition, Charles negotiated the handover of the Vivian H. Taylor collection of acoustically significant historical artefacts and papers to the Melbourne Museum so that these can be displayed on some future occasion and thus not be lost to the acoustics fraternity. Louis Fouvy contributed a vintage GenRad sound level meter and is helping Charles document Vivian Taylor's contribution to Australian acoustics, which will go with the collection.

This year, unfortunately, there was no CSR award for Excellence in Acoustics. We encourage everyone who has done something innovative and different to apply for this Award. Similarly, the AAS offers an education grant to support various acoustical initiatives and we welcome submissions for this award.

In discussion with various consultants and others at the Conference, it seems that the acoustics fraternity is reasonably busy and holding its own. One thing that seems to be mentioned often is the lack of younger people showing an interest in acoustics. This, I think, is demonstrated primarily by the lack of acoustics-related courses being offered around the country in tertiary level institutions. It seems that we need to do a better job at publicising our interest at school level and some divisions have already begun to engage with education and science authorities to try and spark more interest in school age kids. This would seem to be a worthwhile effort and I encourage all Divisions to try and make some noise about the practice of acoustics.

AAS President Awards Fellow Certificate to Louis Fouvy

I take this opportunity to wish everyone a safe and happy holiday and a happy new year. Let's hope that 2010 brings continued prosperity to the acoustics fraternity and success in all our endeavours.

Norm Broner

UNDERWATER NOISE FROM PILE DRIVING IN MORETON BAY, QLD

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This article presents measurements of underwater pile driving noise recorded during the construction of the duplicate Houghton Highway bridge in western Moreton Bay, Queensland. Moreton Bay is a protected marine park, a World Heritage Site and a Ramsar Wetland, providing habitat for turtles, dugong, sharks, dolphins and whales, some species of which are listed as vulnerable to endangered. Pile driving noise was measured for small and large piles at various locations and ranges. Using an acoustic propagation model, a sound map was computed for Bramble Bay. Sound levels were compared to currently available information on impact thresholds. Ranges greater than those corresponding to impact thresholds were scanned for the absence of dolphins before and during pile driving in line with a monitoring and response plan.

INTRODUCTION

In 2008, construction of the duplicate Houghton Highway Bridge began north of Brisbane in western Moreton Bay, Queensland, Australia. Moreton Bay is a marine park, listed as a World Heritage and Ramsar Wetland. Of the marine animals living in Moreton Bay, the Environment Protection and Biodiversity Conservation (EPBC) Act 1999 lists the humpback whale (*Megaptera novaeangliae*), the green turtle (*Chelonia mydas*) and the hawksbill turtle (*Eretmochelys imbricata*) as vulnerable, the loggerhead turtle (*Caretta caretta*) as endangered, and the grey nurse shark (*Carcharias taurus*) as critically endangered. Dugong (*Dugong dugon*) are listed as vulnerable and Indo-Pacific humpback dolphins (*Sousa chinensis*) as rare under the Nature Conservation Act 1992 of Queensland.

With sound travelling much better under water than light, marine animals, in particular marine mammals, rely on acoustics for sensing their environment. Man-made underwater noise has the potential to mask marine mammal communication sounds, environmental sounds (e.g. surf) useful for navigation, predator sounds, prey sounds, and odontocete (toothed whale) echolocation sounds. Noise can alter animal behaviour, animal distribution and habitat usage patterns. At very high levels and under certain circumstances, underwater noise can cause physiological damage to tissues and organs [1].

Pile driving source levels are among the highest of construction activities [2]. In soft substrates vibratory pile drivers are used. They contain a system of counter-rotating eccentric weights, arranged such that horizontal vibrations cancel out while vertical vibrations get transmitted into the pile from above. The sound from vibratory pile driving is continuous and has lower sound levels compared to impact pile driving. A diesel impact pile driver drops a weight through a cylindrical tube onto the pile compressing and heating the air above the pile to the ignition point of diesel fuel injected into the cylinder. The detonation drives the weight back up. Alternatively, the weight can be lifted by means of hydraulics or steam.

When the hammer strikes the pile, sound is created in air at the top of the pile. Acoustic energy spreads as a spherical pressure wave through the air. The impact also gives rise to a stress wave travelling down the length of the pile. This wave couples with the surrounding medium (first air, further down water), radiating acoustic energy into the air and water. The stress wave in the pile also couples with the substrate below the water, creating an acoustic wave travelling through the seafloor. Sound travels as compressional pressure (P) waves and transversal shear (S) waves through the elastic seafloor. Sound can travel very fast and with low attenuation through certain types of seafloor. At some distance away from the pile, acoustic energy can radiate back into the water column from the seafloor. The sound from impact pile driving is transient and discontinuous, called pulsed. Within the water column, the arrival of acoustic pulses from different media and directions and with different phases and time delays tends to result in a complex pattern of higher and lower noise level regions, in particular close to the source.

The level of noise received in the water column at some distance from the pile depends on a multitude of factors, including the size, shape, length and material of the pile, the size and energy of the hammer, the type of sediment and the thickness of the sediment, the type and depth of the underlying bedrock, the water depth, bathymetry, salinity and temperature.

MATERIALS AND METHODS

Two types of hollow steel piles were driven during bridge construction: 624 temporary piles (75cm outer diameter, 12.7mm wall thickness, 28m length) and 156 permanent piles (150cm outer diameter, 25mm wall thickness, 30m length). Underwater noise was recorded from two temporary piles at Pier 2 (on the Brighton shore; 14.5.2008), one permanent pile at Pier 2 (26.7.2008), one permanent pile at Pier 26 and four temporary piles at Pier 29 (19 & 20.3.2009). Temporary and permanent piles were driven with hydraulic piston hammers (Figure 1), model BSP-CG180 (12t weight, 180kJ maximum energy) and model IHC-S280 (14t weight, 280kJ maximum energy) respectively. Temporary piles were driven to about 23m below the seafloor, permanent piles to 26m.

Figure 1: Photo of a temporary pile being driven at Pier 2. The existing Houghton Highway bridge can be seen in the background.

Four different systems were utilized to simultaneously record underwater sound at four different locations and ranges from the piles. The first two systems were autonomous and deployed for 2h at ranges > 200m (Sites 1 & 2). The last two systems were operated manually from a barge drifting close to the piles being driven.

- Geospectrum Technology Inc. hydrophone model M15C (sensitivity -202 dB re 1V/μPa) connected to a Multi Electronique data logger model AURAL-M2. Bandwidth 16384 Hz.
- High-Tech Inc. hydrophone model HTI 96 (sensitivity -164 dB re 1V/μPa) connected to a Multi Electronique data logger model AURAL-M2. Bandwidth 16384 Hz.
- Reson hydrophone model TC4034 (sensitivity -218 dB re 1V/μPa) with external amplifier from Reson model EC6067 connected to a SoundDevices data logger model SDD 722. Bandwidth 48000 Hz.
- Reson hydrophone model TC4043 (sensitivity -201dB re 1V/μPa) connected to a SoundDevices data logger model SDD 722. Bandwidth 48000 Hz.

Frequency responses of the hydrophones and recording systems are measured in the lab every 2 years. In the field, before deployment and after recovery, each recording system was calibrated using a G.R.A.S. pistonphone calibrator model 42AC. The system gain was computed from the recorded calibration signal and applied to the digital recording data to yield sound pressure in units of μPa.

Pile driving and recording locations were measured with a GPS. Distances close to the piles were measured with a Bushnell laser range finder. Water temperature and conductivity were measured with a CTD from AquiStar, model CT2X. The water depth was 1m at Piers 2 & 26 at the time of recording and 1.5m at Pier 29.

Sound metrics

Peak sound pressure level [dB re 1μPa]:

$$SPL_{Pk} = 20\log_{10} (\max(|p(t)|)$$

where $p(t)$ is the time series of pressure measured in the water column.

Peak-to-peak sound pressure level [dB re 1μPa]:

$$SPL_{Pk\text{-}Pk} = 20\log_{10} (\max(p(t)) - \min(p(t)))$$

Root-mean-square (rms) sound pressure level [dB re 1μPa]:

$$SPL_{rms} = 20\log_{10}\left(\sqrt{\frac{1}{T_{90}}\int_{T_{90}} p(t)^2\,dt}\right)$$

where the integral runs over the duration of the pulse, defined as the time over which 90% of the total energy is received. On a cumulative energy curve, the start-time of a pulse is taken at the 5% cumulative energy mark, and the end-time of a pulse is taken at the 95% cumulative energy mark.

Sound exposure level [dB re 1 μPa^2·s]:

$$SEL = 10\log_{10} (\int_T p(t)^2\,dt)$$

which is proportional to the total energy of a plane wave.

1/3 octave band levels: SPL and *SEL* can be computed in a series of adjacent bands, each 1/3 of an octave wide. The following centre frequencies (*fc*) were used: 10, 13, 16, 20, 25, 32, 40, 50, 63, 80, 100, 126, 160, 200, 251, 320, 400, 500, 640, 800, 1000, 1280, 1585, 2000, 2560, 3162, 4000, 5120, 6310, 8000, 10000, 12589, 15849, 20000, 31623, 40000 Hz. Bandwidth (Δf) increases with increasing fc: $\Delta f = (2^{1/6} - 2^{-1/6}) \times fc$.

Power spectrum density levels [dB re 1μPa^2/Hz] give the mean squared sound pressure in a series of adjacent bands of a constant 1 Hz width.

Percentiles: The xth percentile is the level below which the signal falls x% of the time. The 50th percentile is equal to the median.

Sound propagation model

Sound propagation was modelled by the Range-dependent Acoustic Model (RAM) [3]. It is based on the parabolic equation method, assuming that outgoing energy dominates over scattered energy and computing the solution for the outgoing wave equation. As an extension to RAM, shear wave conversion at the sea floor was approximated by the equivalent fluid complex density approach [4]. RAM yields transmission loss data in 2D as a function of range and depth. To achieve a 3D sound level map, RAM was run for a fan of radials from the source, and a tessellation algorithm utilized to seed new radials as the distance between radial end points exceeded a preset resolution parameter.

Table 1: Sound propagation parameters. Depths are below the seafloor.

Depth [m]	Density [g/cm^3]	P-wave speed [m/s]	P-attenuation [dB/λ]
0	1.80	1650	0.165
10	1.60	1700	0.170
20	1.62	1750	0.175
30	1.65	1800	0.180
35	2.40	2900	0.348
40	2.40	3000	0.360
50	2.50	3500	0.420
200	2.58	3800	0.456
2000	2.60	4000	10

The sound propagation model required the geoacoustic input parameters listed in Table 1. A seismic reflection survey and a series of bore hole drillings to 30-35m depth were done by Mapping and Hydrographic Surveys Pty Ltd and the Geotechnical Branch of the Dept. of Main Roads, providing P-wave sound speed profiles. Data for attenuation, density and shear waves were taken from standard reference works [5]. Shear speed was modeled at 418 m/s, shear attenuation at 0.5 dB/ λ. Data below 40m were extrapolated from tables [5]. The sound propagation model assumed an absorbing layer at 2000m depth. The speed of sound in the water column was computed from our temperature and conductivity measurements [6] and was about 1530 m/s in this shallow water.

RESULTS

Pressure waveform

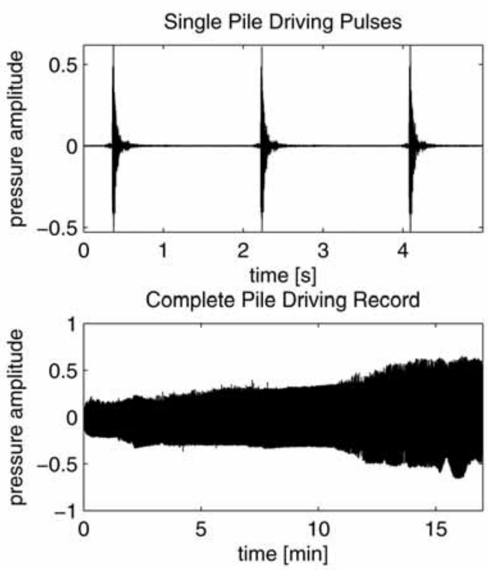

Figure 2: Pile driving waveforms. Single pulses cannot be resolved in the 17min recording.

The acoustic pressure time series recorded underwater exhibited a series of pulses, each pulse corresponding to a single strike of the pile. The pressure rose sharply and then in a dampened oscillation reduced to ambient levels. Figure 2 shows three single pulses recorded at Site 2 from a temporary pile at Pier 29. The interpulse (hammering) interval was 1.8s. Also shown is the complete (17 min) acoustic trace recorded from this pile. The pressure amplitude (not calibrated in this plot) rose by a factor 4 (=12 dB) from the beginning of the trace to the end, because of increasing pile driving energy and increasing resistance. We observed the same increase at Pier 2.

Sound levels

Figure 3 shows the received *SEL* of a temporary pile (driven at Pier 29) and a permanent pile (from Pier 26) recorded at various ranges. *SEL* decreased with range due to propagation losses. The larger permanent piles had higher received levels than temporary piles at similar range. Pile driving noise was very broadband (40 Hz to > 40 kHz) near the source. Absorption as a function of distance increases with frequency, so that at long ranges only energy at frequencies < 400 Hz remained. All levels were computed over two minutes of recording. The statistical variation of *SEL* was largest at low frequencies and long ranges where ambient noise was dominant.

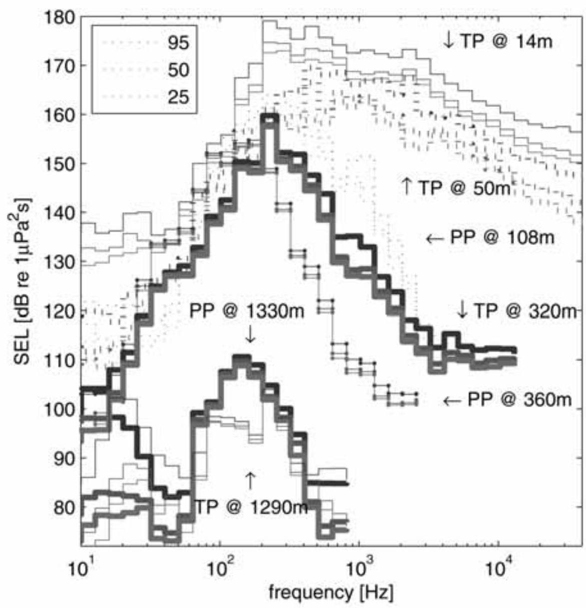

Figure 3: Measured *SEL* (in 1/3 octaves) of a temporary (TP) and a permanent pile (PP) being driven at Pier 29 and 26 respectively. Three lines are shown for each recording: the 90th, 50th and 25th percentiles from top to bottom.

Table 2 summarizes the acoustic properties of the pile driving signals that originated at Piers 26 and 29, and that were measured at different ranges. Peak-peak levels were, of course, highest, followed by peak levels. Root-mean-square levels were lower, because they represent an average pressure over the duration of a pulse. Sound exposure levels were lower still. Broadband *SEL* were computed by summing up the previously plotted 1/3 octave *SEL* on a linear (not dB) scale. The last column gives the length of the pulse. This is the duration over which the pulse energy rose from 5 to 95%. Determining the duration of a pulse is difficult, in particular if the signal is weak (at long ranges) and the background noise loud, or if the pulse is spread out due to dispersion and due to time-lagged arrivals of energy via different propagation paths.

Table 2: Summary of noise level measurements. Means ± standard deviations are given. *: The permanent pile value @ 14m came from Pier 2.

Range [m]	SPL_{PkPk} [dB re 1μPa]	SPL_{Pk} [dB re 1μPa]	SPL_{rms} [dB re 1μPa]	SEL [dB re 1μPa^{2}s]	Pulse Length [ms]
Temporary Pile @ Pier 29 water depth 1.5m					
14	213 ± 2	207 ± 2	194 ± 2	183 ± 2	69 ± 13
51	205 ± 2	199 ± 2	185 ± 2	173 ± 2	59 ± 16
(Site 2) 320	187 ± 1	181 ± 1	173 ± 1	160 ± 1	50 ± 4
(Site 1) 1290	132 ± 1	126 ± 1	115 ± 1	107 ± 1	142 ± 74
Permanent Pile @ Pier 26 water depth 1m					
14*	211 ± 2	205 ± 2	189 ± 1	179 ± 1	94 ± 28
108	198 ± 2	192 ± 2	183 ± 1	168 ± 1	28 ± 3
(Site 2) 360	186 ± 1	180 ± 1	172 ± 1	158 ± 1	36 ± 3
(Site 1) 1330	138 ± 1	133 ± 1	124 ± 1	114 ± 1	101 ± 64

For the temporary pile, the data summarized were all taken from the end of the pile driving, yielding the highest level (worst-case). For the permanent pile, levels at 360m and 1330m range were computed from the end also. At 108m, only the middle section of pile driving was recorded. No close-range data were obtained at Pier 26; the 14m value listed was measured at Pier 2 at the beginning of the pile being driven.

Ambient noise

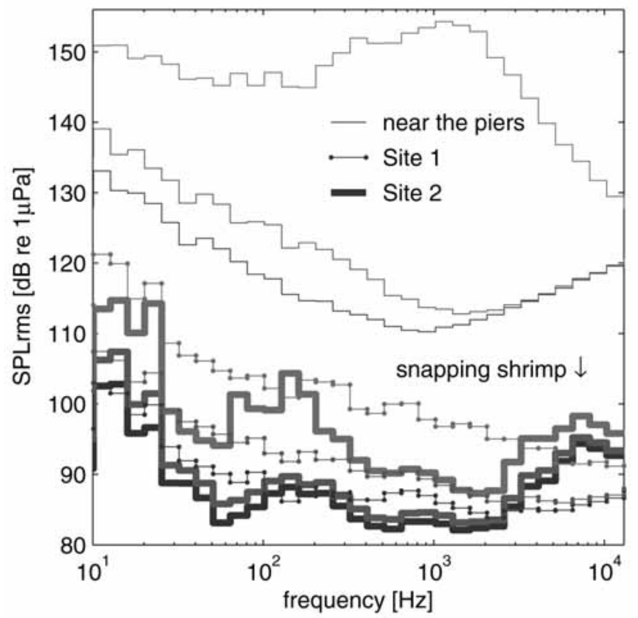

Figure 4: 1/3 octave levels of SPL_{rms} of ambient noise, computed in 1s windows, showing statistics over 15 minutes of recording. Three lines are given for each site corresponding to the 95th, 50th and 25th percentiles.

SPL_{rms} of ambient noise was computed in 1s windows and statistics were calculated for a 15 minute recording at each site. Percentiles of SPL_{rms} in 1/3 octave bands are shown in Figure 4. Close to the piers, right at the construction site, ambient noise was highest and most variable. Construction activities generated noise at low-to-mid frequencies. There were occasional peaks sounding like metal banging (e.g. banging of piles or dropping of chains and metal) to the ear. Furthermore, this site was about 10-15m from the existing Houghton Highway bridge which runs parallel to the duplicate bridge under construction. Cars driving over the existing bridge generated low-frequency noise in the water. There also was splashing and flow noise around the hydrophones, the barge and the piles. The reason why the 25th and 50th percentiles of the noise near the piers increased above 2kHz, was that the rather insensitive recording system used here ran into its noise floor, which shows up as an upslope on 1/3 octave plots (because bands get wider with increasing frequency, adding increasingly more self-noise).

The systems deployed at Sites 1 and 2 were much more sensitive. There was a bit of water flow noise in the recordings. At Site 2, the sound of snapping shrimp was clearly audible in the data. These animals are common in tropical and subtropical regions throughout the world's oceans. The snapping sound occurs when the animal snaps its claw. Near a colony of shrimp, the ambient noise sounds like continuous crackling ('frying' or 'wood burning') with energy between 2 and 24 kHz [7]. Spectra measured here matched published spectra very well [8,9]. There was quite a bit of sand swishing noise at Site 1 (sand swishing over the base of the instrument as evidenced by the presence of sand in nuts and bolts upon retrieval). Weather conditions on March 19 and 20 were cloudy with moderate wind adding to low-frequency noise. We did not see or hear any boats in the vicinity at the times of recording.

Sound propagation

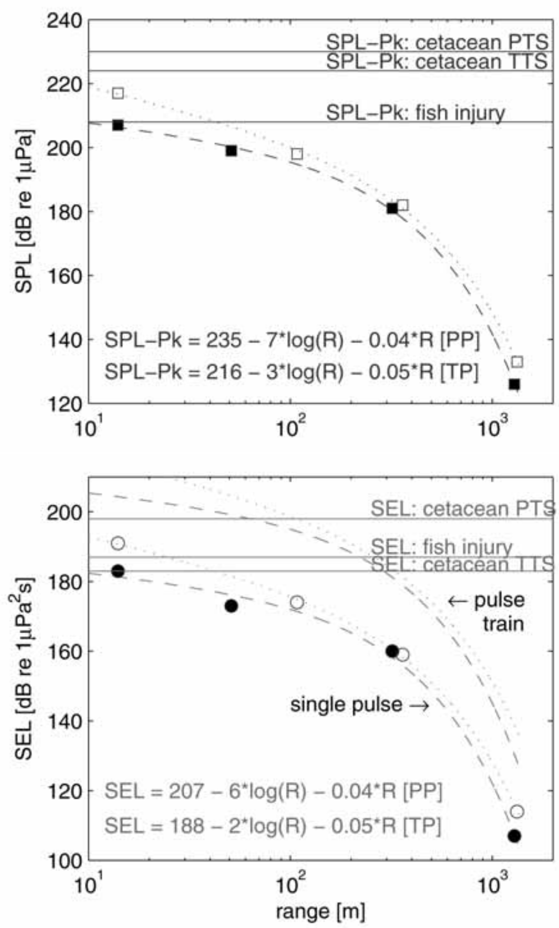

Figure 5: Received SPL_{Pk} (top) and SEL (bottom) from a permanent pile at Pier 26 (hollow symbols) and a temporary pile at Pier 29 (solid symbols). Regression for permanent pile (..), temporary pile (--). SEL as a function of range are shown for a

single pulse and a 17-minute pulse train. Current suggestions for fish and dolphin impact thresholds (solid lines). *SEL* thresholds for fish injury and cetacean TTS correspond to single pulses only.

SPL_{Pk} and *SEL* values from Table 2 were plotted in Figure 5. The permanent pile levels at 14m range were measured at the beginning of driving this pile; 12 dB (the increase from beginning to end measured at Piers 2 & 29) was added to estimate the levels at the end of driving this pile. The permanent pile levels at 108m range were recorded when the middle section was driven, and therefore enlarged by 6 dB to estimate the end levels. Regression analysis was performed using a spreading term proportional to the logarithm of range (R) and an absorption term proportional to range.

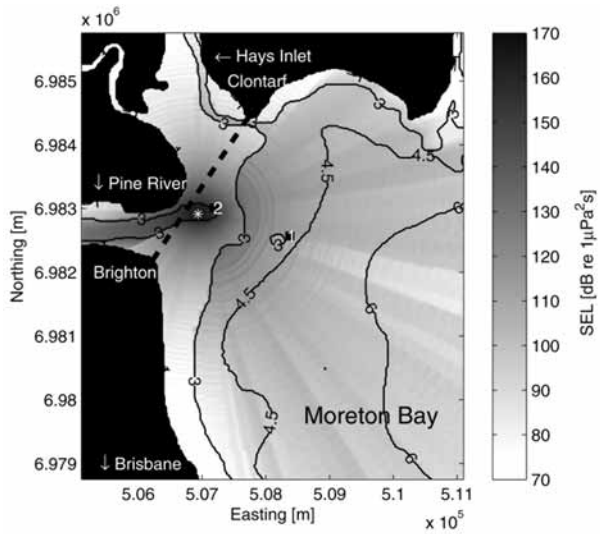

Figure 6: Modelled *SEL* from a temporary pile being driven at Pier 29 (white asterisk). Recording sites 1 & 2 marked in white. Dashed line: bridge under construction. Coordinates refer to UTM Zone 56.

Results of the 3D sound propagation model RAM are shown in Figure 6 for a temporary pile at Pier 29. Plotted are the maximum *SEL* over all depths. Sound energy got channelled into the deeper waters of Pine River and Hays Inlet. Shadowing occurred behind the circular rise near Site 1. At Site 1, the modelled level was 110 dB; compared to 107 dB measured. At Site 2, the modelled level was 151 dB compared to 160 dB measured.

DISCUSSION

Comparison with other pile driving measurements

A number of studies have recorded impact pile driving, only few data are reported in the peer-reviewed literature. Comparison is difficult, because important information is often missing (such as geoacoustics of the location, water depth, pile and hammer parameters). Having said that, some general trends can be observed. For example, noise increases with pipe diameter and blow energy [10].

The temporary pile driven at Pier 29 had an outer diameter of 75cm. Steel piles of 76cm diameter, diesel-driven, produced SPL_{Pk} = 208 dB re 1 µPa, *SPLrms* = 192 dB re 1 µPa and *SEL* = 180 dB re 1 µPa^{2}s at 10m range [11]. We measured SPL_{Pk} = 207 dB re 1 µPa, *SPLrms* = 194 dB re 1 µPa and *SEL* = 183 dB re 1 µPa^{2}s at 14m range--comparable.

The permanent pile driven at Pier 26 had an outer diameter of 150cm. A hollow steel pile of 168cm diameter, diesel-driven, was recorded at 4, 10 and 20m range; SPL_{Pk} was 219, 210 and 204 dB re 1 µPa respectively; *SPLrms* was 202, 195 and 189 dB re 1µPa [11]. We estimated a maximum SPL_{Pk} of 217 dB re 1 µPa and a maximum *SPLrms* of 201 dB re 1 µPa at 14m range for the end of the pile driving, which is comparable. A 160cm-diameter pile, driven with an 80-200kJ hammer and measured at 750m range, had *SEL* = 162 dB re 1 µPa^{2}s [10]. Our measurement at 750m was less possibly due to the high-loss environment in this shallow part of Moreton Bay.

Bioacoustic impact on marine fauna

There is limited knowledge of the effects of pile driving noise on marine life. The only damage that has ever been observed in any marine species as a result of near-by pile driving is lethal injury in fish [12,13]. The latter study was initially carried out by this author. The piles around which dead fish were seen were closed-end piles that were driven into rocky ground; a very different scenario. Post-mortems revealed haemorrhaging and burst swim bladders. Unfortunately, neither of these projects was able to produce any information on the sound metrics and thresholds responsible for the observed damage. A review of the available scientific information on bioacoustic impact on fish [11,14] led to the derivation of interim criteria for injury of fish exposed to pile driving [15]. For any single strike, a sound exposure level of *SEL* > 187 dB re 1µPa^{2}s and a peak sound pressure levels of SPL_{Pk} > 208 dB re 1µPa could cause injury.

No study to-date has shown injury in marine mammals from pile driving. However, data on temporary hearing loss (TTS: temporary threshold shift) after exposure to intense sound, including pulsive sound resembling seismic airguns, has been measured. After a major review effort, marine mammal noise exposure criteria were released [16]. Injury was understood as the onset of PTS (a permanent threshold shift) and extrapolated from TTS data. For PTS in cetaceans (whales, dolphins and porpoises) from single or multiple pulses, the threshold was SPL_{Pk} > 230 dB re 1µPa and *SEL* > 198 dB re 1µPa^{2}s (M-weighted [16]). A TTS in a mid-frequency cetacean was observed at SPL_{Pk} > 224 dB re 1 µPa and *SEL* > 183 dB re 1µPa^{2}s after exposure to a single pulse [17].

These levels were shown as horizontal straight lines in Figure 5. Looking at the SPL_{Pk} criteria, levels for the temporary pile were below all thresholds. The permanent pile was above the fish injury threshold over 40m range, but below the cetacean thresholds. *SEL* in Figure 5 were not M-weighted and are thus higher than if they had been weighted for mid-frequency cetaceans (M-weighting reduces the energy at frequencies below a few hundred Hz, where most of the pile driving energy is distributed.). The comparison with SEL criteria is thus “conservative”. A single pulse from the temporary pile was below all thresholds; a single pulse from a permanent pile exceeded the fish injury threshold for ranges <25m and the cetacean TTS threshold for ranges <35m. Summing *SEL* over an entire pulse train (Figure 2b), yielded cumulative *SEL* that were 23 dB higher than single-pulse *SEL*, surpassing the cetacean PTS threshold over ranges <60m for the temporary and <100m for the permanent pile. It is unlikely that an animal would remain within close range for the duration of driving an entire pile. If an animal starts moving away after receiving one pulse, and given that received levels drop quickly with increasing range, the single pulse curves might be more applicable than the cumulative *SEL* integrated over an entire pulse train. There are no data on cumulative *SEL* thresholds for fish injury or cetacean TTS; the thresholds plotted are for single pulses only. It took less than 20 minutes to drive the recorded piles. If all went well, a maximum number of two temporary piles and one permanent pile were driven in one day, yielding a duty cycle (the ratio of pile driving noise being 'on' and 'off') of 60 min / 24 h = 0.04. It took a minimum of 24-28h before the next set of temporary piles could be driven, and at least a week in between permanent piles.

Moreton Bay has two resident dolphin species, both mid-frequency cetaceans, the bottlenose dolphin (*Tursiops truncatus*) and the indo-Pacific humpback dolphin (*Sousa chinensis*). Local residents and construction workers have reported sighting dolphins in the area throughout the year. However, we did not observe any animals within the vicinity of the piles during the recording, and a shut-down zone of 200m radius was scanned for cetacean presence as part of a monitoring and management plan, which J.F. Hull, Albem, the EPA and the Department of Main Roads had worked out. Elsewhere, harbor porpoises (*Phocoena phocoena*) avoided close ranges and responded to pile driving sound at ranges > 21km [18] from larger piles; however, received levels at these ranges were not reported.

We did not observe any dead fish close to the pile, nor did we see fish in abundant numbers here. It is possible that fish temporarily avoided close ranges to the pile; behavioural thresholds for fish require more research.

ACKNOWLEDGEMENTS

The duplicate Houghton Highway bridge was built by J.F. Hull Holdings Pty. Ltd. and Albem Pty. Ltd. in a joint venture. Their teams were extremely helpful and supportive. Special thanks go to Gary Christian of Albem and Clayton Smith of J.F. Hull. It was a pleasure working with them.

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DUCT DIRECTIVITY INDEX APPLICATIONS

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This paper concerns the application of a recently developed chart for determining the directional properties of sound emitted from the open end of a ventilation duct. When designing a duct silencer to reduce noise from a large vertical discharge duct, it is useful to note that the first 5 to 10 dBA noise reduction may result from directivity losses at

90 degrees and can be accurately predicted. In 1971 the first author conducted sound directivity tests with 300 and 600 mm diameter ducts and the results were made into a rough chart of Duct Directivity Losses that ultimately found its way into the NSW EPA Environmental Noise Control Manual (5 June 1985, page 207.1). It is wrong in principle and rather inaccurate, but some users are unaware of its failings.

Over the last 13 years further duct directivity testing has been conducted and a new duct directivity chart drawn. It is based on sound directivity testing on ducts of 305, 400, 610, 915 and 1220 mm diameter. The directivity data has been related to the sound power level of noise emitted from the duct and the spherical dispersion of sound energy. The new Duct Directivity Chart allows the directivity gain or loss to be obtained for any diameter from 100 mm to 10 metres, at angles from zero to 135 degrees without the need for complex calculations.

INTRODUCTION

In this paper the basic principles of duct directivity are explained in order to reveal the basic errors of earlier directivity charts and to provide practical engineering applications for the improved Duct Directivity Chart shown in Figure 2.

A rough chart for calculating the Directivity Loss of an open-ended duct was prepared by the first author for Vokes Australia Pty Ltd in 1971 (see Figure 3). This chart was a poor attempt to correlate the levels in line with the end of the duct with those at various angles around the duct up to 135 degrees. These were not Directivity Indices that would accurately relate to the sound power of noise emitted from the open end of a duct. The first author modified it for the Department of Public Works in 1984 and this has found its way into the NSW EPA Environmental Noise Control Manual – page 207.1 dated 5 June 1985 (see Figure 4).

DUCT DIRECTIVITY INDEX (DI)

If the noise emission from a ventilation duct were equal in all directions, it would be "omnidirectional". If the duct diameter were small compared to the measurement distance, it could be considered to be a point source and the sound pressure level could be calculated from the known sound power level using:

$$L_p = L_w - 10 \text{ Log } S - A_E$$

where:

L_p is the sound pressure level - dB re: 20 μPa

L_w is the sound power level - dB re: 1 pW

S is the area of the measurement surface in m^2

A_E is the excess attenuation (dB) due to the ground effect, other reflecting surfaces, air absorption, obstacles between the source and receiver and meteorological effects.

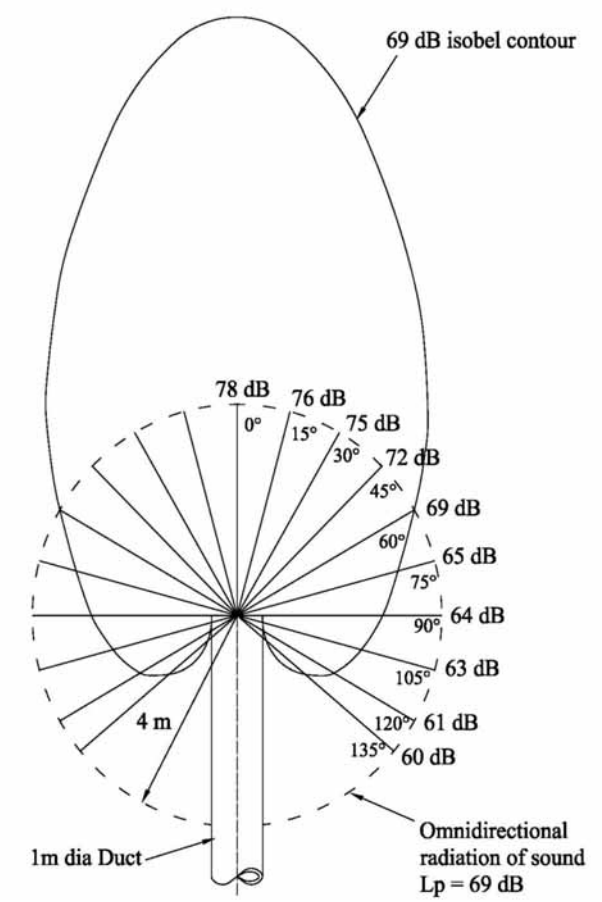

Figure 1. Typical Duct Directivity Isobel Plot

If the sound emitted from a duct termination with a sound power level of 92 dB at 500 Hz were omni-directional and if the excess attenuation were zero, the sound pressure level at 4 metres would be 69 dBA in a spherical pattern as shown in Figure 1. However, it has been established by rigorous testing that noise emission from the end of a duct is quite directional. Using the Directivity Indices from Figure 2, the predicted sound pressure levels at a distance of 4 metres have been plotted. The 69 dB sound level Isobel curve is also shown.

If the noise emitted from the open the end of a duct is "directive", more of the sound energy goes in one direction than another. However, "directivity" does not affect the overall sound power level. If more noise is emitted along the axis of the duct, then less will be emitted in other directions. It can be seen in Figure 2 that for directions characterised by an angle from the duct axis of less than 60 degrees, the sound pressure levels are higher than the mean omni-directional sound pressure level and for angles greater than 60 degrees, the sound pressure levels are lower.

The Vokes chart shows only losses; hence it may not be considered to be an acceptable Directivity Chart. The Directivity Charts published by Bies and Hansen (2009), (Figures 9.29, 9.30 and 9.31) are all valid in principle since they all show both gains and losses. Figure 9.29 of the referenced textbook is based on research with small tubes and employed acoustic modelling techniques to extrapolate data for larger duct sizes. It may or may not be accurate. Also, Figure 9.30 in the text book is slightly different to Figure 2 below due to a small plotting error in the textbook figure. Figure 9.30 is based on directivity testing with a range of typical industry duct sizes by professional engineers and provides comparable but slightly more conservative results than the two other graphs in the textbook.

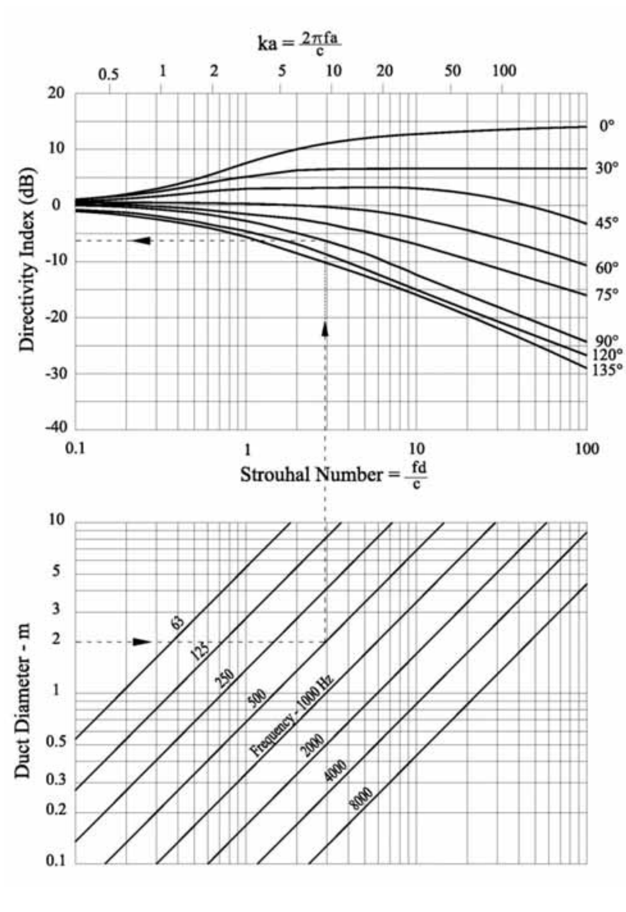

Figure 2 Duct Directivity Chart

Example: By following the dotted lines in the Figure 2 chart, it can be seen that a 2 metre diameter duct has a 6 dB directivity loss in a direction at 90 degrees to the duct axis at a frequency of 500 Hz. At 4000 Hz the DI is -16 dB.

VOKES AUSTRALIA AND NSW EPA DUCT DIRECTIVITY CHARTS

The first author feels obliged to expose the errors of the older charts that he derived many years ago. These are presented below for reference purposes.

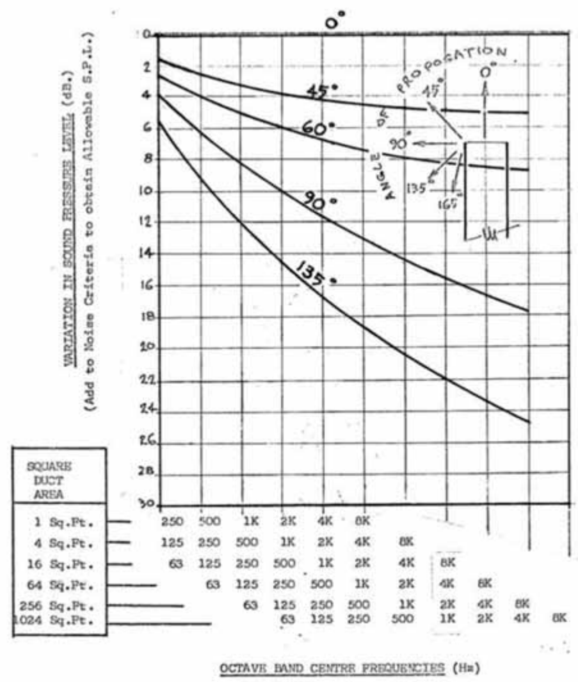

Figure 3. Vokes Australia – Duct Directivity Chart

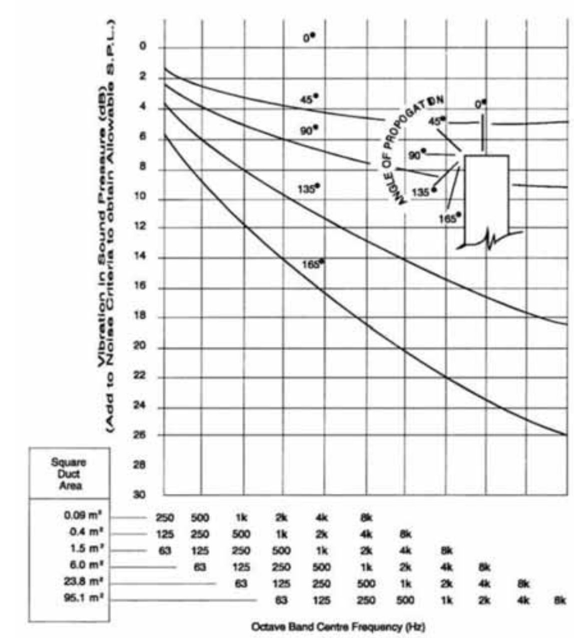

Figure 4. NSW EPA Duct Directivity Chart 1985

Figure 4 indicates that a 2 metre diameter duct with a cross sectional area of 3.14 m^2 has a directivity factor of 6.5 dB at 500 Hz and 90 degrees. At 4000 Hz the directivity loss is 8 dB. These differences are fortunately small. However at 0 degrees, Figure 2 shows an 11 dB gain while Figure 4 shows a 0 dB gain, which is a gross error.

DUCT DIRECTIVITY MEASUREMENTS

In 1995, sound directivity tests were undertaken for 400 and 1220 mm diameter ducts by Neish [5]. Flanking sound proved to be a significant problem at angles in excess of 90 degrees, so the 1220 diameter duct measurements were made with greater care, including a blocked end test to quantify worst case flanking transmission.

Source: Murray Neish
Figure 5 Murray Neish Testing a 400 mm Duct

In 2006, further duct directivity tests were undertaken using 305 mm, 610 mm and 915 mm diameter ducts [4].

Source: Daniel Potente
Figure 6. Loaded vinyl wrapped 915 mm duct

DIRECTIVITY INDEX CALCULATIONS

The Duct DI of an open-ended duct is the difference in decibels between the sound pressure of an omni-directional noise source and that from the duct termination.

Most ventilation ducts are small compared to the distances at which their noise emission is of interest, so they are often assumed to be point sources. However, noise from a true point source is omni-directional and radiates sound energy in a spherical manner. It has been observed that noise emission from the open end of a duct radiates in a directional manner, tending to be higher along the duct axis. Measurements show that the larger the duct diameter and/or the higher the sound frequency, the greater is the directivity effect.

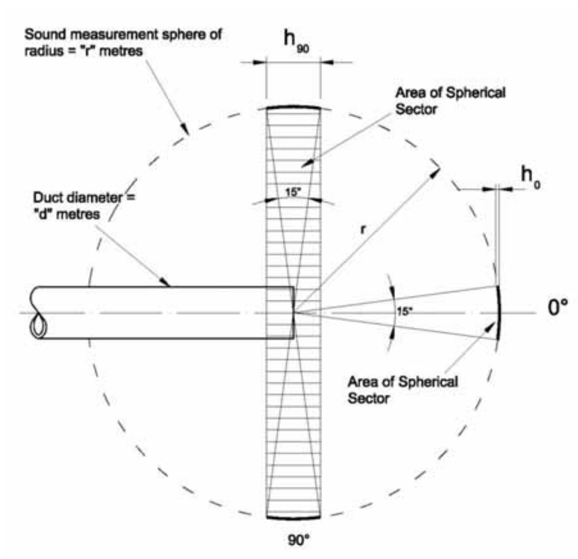

Figure 7. Diagram of Sound Radiation from an Open Ended Duct

In Figure 7 it can be seen that the area of the sound measurement sphere subtended by the 0 degree angle is significantly less than the area of sphere subtended by the 90 degree angle. The surface area of a sphere is $4\pi r^2$, where r is the radius of the sphere. The annular surface area of a sector of a sphere is $2\pi rh$, where h is the height of the sector and where the centre of the sector corresponds to the particular directivity angle. The sound power emitted from a duct for each angular sector is equal to the average sound intensity for that sector multiplied by its radiation area and may be quantified as follows.

$$L_w = L_p + 10 \log (2\pi rh)$$

The total sound power level of the source is the sum of the individual measured sound power levels of all the sectors from 0 to 180 degrees.

For an omni-directional sound source, the Lp at any specific distance will be the same all around the sphere. The mean omni-directional $L_p = L_w - 10 \text{ Log } S$

$$= L_w - 10 \log (4\pi r2)$$

The DI corresponding to any particular angle is the difference between the measured sound pressure level at that angle and the mean L_p that would be produced at the same location by an omni-directional sound source of the same total sound power level.

Directivity Indices when plotted against the Strouhal Number were found to follow the curves shown in Figure 2. The Strouhal Number, *Str* is a dimensionless number that relates the frequency, f duct diameter, d, and the speed of sound, c, as follows:

$$Str = f\, d \,/\, c$$

We have added dimensionless "ka" values to Figure 2 for comparison with values found in Bies and Hansen [1].

APPLICATION OF DI DATA

To calculate the Sound Pressure Level at a receptor location at a given distance from an open-ended duct, one should calculate the level based on omnidirectional radiation then apply the Directivity Indices (DIs) in octave bands and sum the resulting levels to predict the overall level at the receptor. From Figure 2 we would determine the DI results at 90 degrees from a 2 metre diameter duct as follows in Table 1.

Table 1. Typical Predicted DI

Frequency Hz	63	125	250	500	1k	2k	4k	8k
Strouhal	.37	.73	1.5	2.9	5.8	12	23	47
DI	-1	-2	-4	-6	-9	-13	-17	-21

The scatter of data shown in Figure 9.33 on page 512 of Bies and Hansen [1] is both above and below the curve of best fit. At 90 degrees, the scatter of data is mainly within plus or minus 3 to 4 dB. We consider that for broadband noise the DI should have an accuracy of ± 3 dB. For noise characterised by a narrow band of frequencies, the accuracy of the DI prediction may be ± 4 dB.

When calculating the Sound Power Level based on field measurements of Sound Pressure Level near the open end of a duct, it is necessary to record the distance from the centre of the duct termination and the angle of measurement from the duct axis. Care should be exercised to ensure that ambient noise from other sources is excluded. Measurement on the duct axis is often impossible because of noise generated by airflow over the microphone. Measurements at 45, 60, 75 and 90 degrees to the axis are recommended. These can be corrected for directivity to determine the omnidirectional equivalent Sound Pressure Level from which the Sound Power Level is calculated. The omnidirectional Sound Pressure Level can be approximated by measurement at 60 degrees, because the DI is close to zero at this angle.

When calculating the Sound Pressure Level from a duct discharge of known Sound Power Level we use:

$$L_p = L_w - 10 \text{ Log } (4\pi r^2) - A_E + \text{DI} \quad \text{at the angle of interest}$$

The DI must be determined at each octave band for a given duct diameter and angle from the duct axis. If the duct is above and adjacent to hard ground and there are no other excess attenuation effects, A_E = -3 dB.

Noise from large lantern vents on factory rooftops has been found to be very directive. Davy's theory on directivity from roof openings indicates that it may be quantified using Figure 2 assuming a diameter equal to the width for directivity normal to the long side and diameter equal to the length for directivity normal to the short side [2].

If the sound pressure from a duct is measured at a specific angle to the duct axis and at a distance r_b, it is not necessary to calculate the sound power level in order to determine the sound pressure level at any other angular location or at any other distance, r_a from the duct outlet. The sound pressure level at location *a* can be calculated from that measured at location *b* using:

$$L_{pa} = L_{pb} - 20 \text{ Log } (r_a/r_b) + \text{DI}_\text{a} - \text{DI}_\text{b}$$

where DI_a is the directivity index corresponding to the angular location of point *a* and DI_b is the directivity index corresponding to the angular location of point *b*. The equation holds provided that:

- all locations are sufficiently far from the duct outlet to be in the far field;
- the difference in excess attenuation effects from one location to another is taken into account; and
- breakout from the walls of the duct is contributing negligibly to the sound levels at locations a and b.

DI OF LINED DUCTS

Bies and Hansen [1] point out in Section 9.15 of the referenced textbook that: "Ducts lined with sound absorbing material radiate more directionally so that higher on-axis sound levels are produced."

CONCLUSION

The Directivity Index chart presented in Figure 2 of this paper is based on comprehensive testing by acoustical engineers using calibrated precision instrumentation [3]. Figure 2 is considered suitable for general use by acoustical engineers for the prediction of duct directivity gain or loss.

The Directivity Index results published in Section 9.15 of the textbook by Bies and Hansen [1] use the dimensionless parameter *ka*, which can be found by multiplying the Strouhal Number by π. Values for ka have been added to Figure 2 to allow comparison with the charts in the Bies and Hansen textbook. Please note the close agreement.

ACKNOWLEDGEMENTS

The duct sound-directivity data measured by both Murray Neish and Daniel Potente for a range of typical duct sizes and presented in previous papers, has been re-analysed by the authors and its use is gratefully acknowledged.

During the course of preparing this paper, the authors had extensive discussions with Associate Professor John Davy of the Royal Melbourne Institute of Technology. We cannot thank him enough for his insight and guidance on the analysis of our measured results and Directivity Index calculations.

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BINAURAL MEASUREMENT AND SIMULATION OF THE ROOM ACOUSTICAL RESPONSE FROM A PERSON'S MOUTH TO THEIR EARS

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This paper outlines methods to simulate the sound of one's own voice as it is affected by room acoustics, using binaural technology. An oral-binaural room impulse response (OBRIR) measurement can be made of a real room environment from the mouth to the ears of the same head. For simulation, a talker's voice is convolved in real-time with the OBRIR, so that they can hear the sound of their own voice in the simulated room environment. We show by example how OBRIR measurements can be made using human subjects (by measuring the transfer function of speech) or by a head and torso simulator (HATS), and we illustrate the differences between individualised measurements and HATS measurements. We extend the HATS measurement method through binaural room scanning, which allows the simulation system to produce natural changes in the OBRIR as subjects rotate their heads while listening to their own voice.

INTRODUCTION

There are many situations in which the sound of one's own voice produces a striking aural effect, for example an unfurnished room, a very small or very large room, an anechoic room, a reverberant room, or rooms with various echo phenomena. In less extreme everyday situations, we may analyse aspects of our environment through such acoustic feedback [1], and the feedback plays a significant role in speaking [2-4] and in playing music [5]. This paper is concerned with techniques that can be used to measure the room acoustical feedback in real rooms from real or artificial speech, for the purpose of simulation. By simulation, we mean a system that allows a speaking person to hear the sound of their voice in real-time in the simulated rooms. The purpose of this could be a tool for the scientific study of self-sound in relevant architectural acoustical contexts (for example, stage acoustics, classroom or lecture theatre acoustics, meeting room acoustics, etc), and could also contribute to virtual reality applications such as teleconferencing and games.

The sound of one's own voice has three components: corporeal transmission (usually referred to as bone conduction); 'direct' airborne transmission from mouth to ear (including body-related acoustic effects, such as shoulder reflections); and reflections from the environment. Nukina and Kawahara [6], Pörschmann [7], and predecessors such as Békésy [8], have studied the first two of these, showing that air-conducted and bone-conducted sound are of similar magnitudes (most similar between 500 Hz and 3 kHz, wherein the acoustic power of the voice is greatest). Outside this range, air-conducted sound is greater than bone-conducted sound. Almost all acoustic radiation is from the mouth. However, in the present paper we are mainly concerned with the third component of the sound of one's voice: reflection from the environment. Pörschmann [9] has approached this problem using computer modelled virtual environments, but the present paper is concerned with measuring and simulating real environments.

A closely related approach to this problem was taken by Sato et al. [2] in a study of listening difficulty, talking difficulty and conversational speech difficulty. They implemented a system with a microphone 0.1 m from the subject's mouth, which fed the signal through a two-channel real-time convolver to simulate the room reflections at the subject's ears (using ear loudspeakers, AKG K1000). A convolver performs an operation equivalent to time-domain convolution of the almost anechoic speech input signal with the room's binaural impulse response, although the operation is usually implemented (at least partly) by multiplication in the frequency domain. They found that talking and conversing difficulty were much more sensitive to clarity index (C50) than was listening difficulty. Again, the difference between the present study and that study is that we wish to accurately simulate real reverberant environments, whereas Sato et al. did parametric control of the simulation's reverberation.

This paper restricts its attention to binaural spatial analysis and synthesis. It is also feasible to approach the problem using high order microphone systems for measurement, and high order loudspeaker systems in the sound-field simulation, although that is much more complex to implement (Ueno and Tachibana's [5] system for stage acoustics simulation for musicians is a simple version of that approach, with six measurement and convolution/reproduction channels, and Favrot and Buchholz [10] have devised a system for real-time auralization of computer-modelled room reflections from a person's speech using many loudspeakers via high order ambisonics). In room acoustics, the term 'binaural room impulse response' (BRIR) is frequently used to denote the impulse response from a source to the two ears of a binaural receiver. In this paper, we use the term 'oral binaural room impulse response' (OBRIR) to make clear that we are discussing the room impulse response from a mouth (or mouth simulation) to the ears of the same head.

MEASUREMENT

2.1 Method

In this section, we compare two approaches to measuring the OBRIR: using a head and torso simulator, and using a real person. A head and torso simulator equipped with mouth and ear simulators provides an obvious approach to the measurement of OBRIRs. It is a simple matter to measure the transfer function or impulse response from an input signal (fed to the loudspeaker of the mouth simulator) to the output signals (from the ear microphones). Alternatively, a transfer function can be measured from a microphone near the mouth to the ear microphones. We take the second approach, which has the advantage of removing the response of the mouth simulator from the measurement, and is also well-suited to simulation – as a talking subject can have a microphone positioned similarly near their mouth as part of the simulation system.

We tested this approach to measurement using a Bruel & Kjaer 4128C head and torso simulator (HATS). As shown in Figure 1, the mouth simulator directivity of the HATS is similar to the mean long term directivity of conversational speech from humans, except in the high frequency range [11, *c.f.* 12]. The HATS' standard mouth microphone position (known as the 'mouth reference point') is 25 mm away from the 'centre of lip' (which in turn is 6 mm in front of the face surface) [13, 14]. We used a Bruel & Kjaer Type 4939 (1/4″) microphone at the mouth reference point. Rather than using the inbuilt microphones of the HATS (which are at the acoustic equivalent to eardrum position), we used some microphones that are positioned near the entrance of the ear canals (Bruel & Kjaer type 4101). One reason for this is that we could use the same microphones on a real person at equivalent positions. Another reason is that it is desirable to avoid measuring with ear canal resonance, as the strong resonant peaks would need to be inverted in the simulation, which would introduce noise and perhaps latency.

The measurement was made by sending a swept sinusoid test signal to the mouth loudspeaker, the sound of which was recorded at the mouth and ear microphones (Fig 2). The sweep ranged between 50 Hz – 15 kHz, with a constant sweep rate on the logarithmic frequency scale over a period of 15 s. A signal suitable for deconvolving the impulse response from the sweep was sent directly to the recording device, along with the three microphone signals. This yielded the impulse response (IR) from the signal generator to each of the three microphones, and we obtained the transfer function from mouth microphone to ear microphones by dividing the latter by the former in the frequency domain. The procedure for this is, first, to take the Fourier transform of the direct sound from the mouth microphone impulse response, zero-padded to be twice the length of the desired impulse response. The direct sound is identified by the maximum absolute value peak of the mouth microphone IR, and data from -2 to +2 ms around this is used, with a Tukey window function applied (50% of the window is fade-in and fade-out using half periods of a raised cosine, and the central 50% has a constant coefficient of 1).

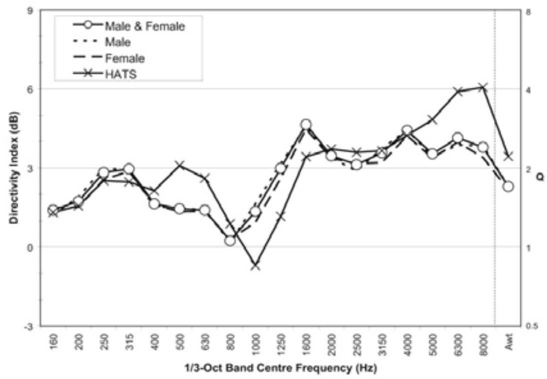

Figure 1. Directivity of a Bruel & Kjaer 4128C head and torso simulator (HATS) compared to the long term directivity of conversational speech (derived from Chu and Warnock's data [11]).

The same Fourier transform window length is used for each of the ear microphone impulse responses, with the second half of the window zero-padded. The transfer function is obtained by dividing the cross-spectrum (conjugate of mouth IR multiplied by the ear IR) by the auto-spectrum of the mouth microphone's direct sound. Before returning to the time domain, we bandpass-filter the transfer function to be within 100 Hz – 10 kHz to avoid signal-to-noise ratio problems at the extremes of the spectrum (this is done by multiplying the spectrum components outside this range by coefficients approaching zero). After applying an inverse Fourier transform, we truncate the impulse response (discarding the latter half). The resulting IR for each ear is multiplied by the respective ratio of mouth-to-ear rms values

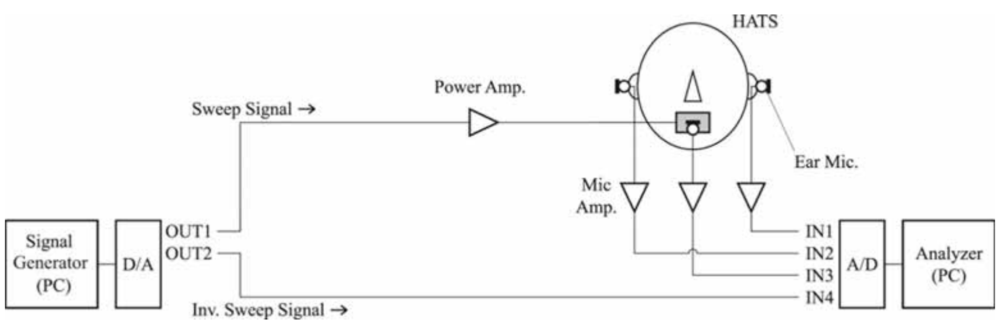

Figure 2. System for measuring OBRIRs using a head and torso simulator (HATS)

of microphone calibration signals (sound pressure level of 94 dB) to compensate for differences in gain between channels of the recording system. To test the process, we made these measurements in an anechoic room and a reverberant room (130 m^3, with a mid-frequency reverberation time of 2.5 s).

Measuring OBRIRs using a real person can be done using a similar microphone arrangement. The sound source could simply be speech, although other possibilities exist. The transfer function is calculated between a microphone near the mouth to each of the ear microphones. This approach was taken by Pörschmann and Nukina and Kawahara in measuring the transfer function from mouth to ear (without room reflections), but it can be used for measuring room reflections too. The advantages of using such a technique (compared to using the HATS) could include matching the individual long term speech directivity of the person; matching the head related transfer functions of the person's ears; and that the measurement system only requires minimal equipment (three microphones). Disadvantages may include effects of time-variance, a poorer signal-to-noise ratio in the measurement, and that some reverberation will be mixed with the direct sound at the mouth microphone (because we cannot isolate the direct sound as we could with an impulse response measurement).

We tested the real person method using the first and second authors. A B&K Type 4939 microphone was positioned near the middle of the mouth (taped to the nose, with a windshield), and the ear microphones were the same as those used in the HATS measurements. The mouth microphone was about 40 mm from the mouth (i.e. further than the HATS microphone). A laser-pointer was attached to the top of the head so that we could maintain an approximately constant head position during prolonged speech utterance (no physical head restraint was used). About ten minutes of continuous speech was recorded, and measurements were made in the anechoic and reverberant rooms as for the HATS. As the authors had different standing head heights, the HATS measurements in the reverberant room had been made to match both heights.

The transfer functions from mouth to ears were derived using the cross-spectrum method [15] with window lengths of 2^{16} samples for the anechoic room, and 2^{18} samples for the reverberant room (sampling rate of 48 kHz), with a Hann window function and a window overlap of 90%. However, we only used the 50% of the windows that had the highest signal level in the mouth microphone, so as to increase the signal-to-noise ratio of the process. The transfer function is estimated from the average cross-spectrum divided by the average auto-spectrum of the mouth microphone signal. The extremes of the spectrum (below the lowest speech fundamental, and above 10 kHz) could not be reliably processed, but indeed are not important for a reflected speech simulation system. There are two limitations to processing in the very high frequency range (above 10 kHz): the signal-to-noise ratio is poor because the voice produces little very high frequency energy at the ears; and the effect of time variance (variable directivity due to varying mouth shape and incidental head movement) is greatest for short wavelengths. We use a 100 Hz – 10 kHz bandpass filter in the same way as for the HATS measurements. The impulse response is obtained by inverse Fourier transform of the transfer function, and the latter half of the impulse response is discarded. An estimate of the reliability of the transfer function estimate is given by the associated coherence function (a value between 0 and 1, which is the squared absolute value average cross-spectrum divided by the product of the two average auto-spectra).

2.2 Results

The anechoic measurements (shown in Figure 3) have similar magnitude spectra to those of previous studies. The magnitude of the HATS transfer function is less than that of the human measurements because its microphone is closer to the mouth. The main notches in the spectra are due to the shoulder reflection, and so the tuning of the notches is affected by the mouth microphone position. However, it should be remembered that we are not aiming to simulate direct sound in the present study, so the mouth microphone position should not be critical (so long as it is near the mouth on the median saggital plane).

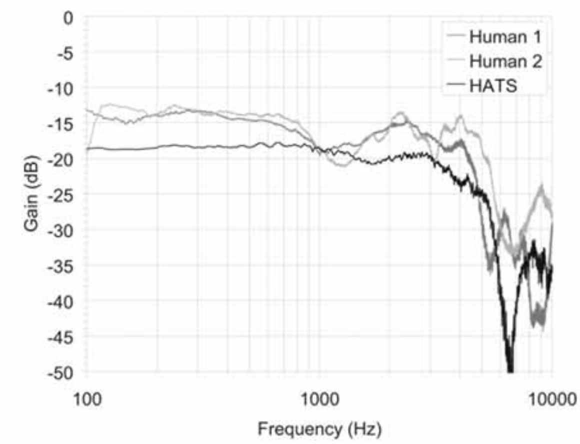

Figure 3. Magnitude of the transfer function from mouth microphone to ear microphone measured from speech (human measurements) or from a swept sinusoid test signal (HATS measurement).

Figure 4 compares the magnitude of the transfer functions (right ear only) for HATS and human measurements in the reverberant room. Differences are greatest in the high frequency range, and it can be observed that the general forms of the curves are similar to the respective anechoic measurements. While the HATS has greater high frequency gain in the reverberant room, this is not seen to the same extent in the human measurements – which may be due to the human mouth's time-varying radiation pattern within the measurement period (indicated by lower associated coherence values).

Comparison of the fine temporal structure of the reverberant room OBRIRs between the HATS and human measurements shows some similarity in peak times and levels up to about 50 ms (Figure 5). Beyond 50 ms, it is difficult to see any relationship between the fine structures. In Figure 5, the normalised values for humans after the direct sound are a little lower than those for the HATS because of the different mouth microphone position.

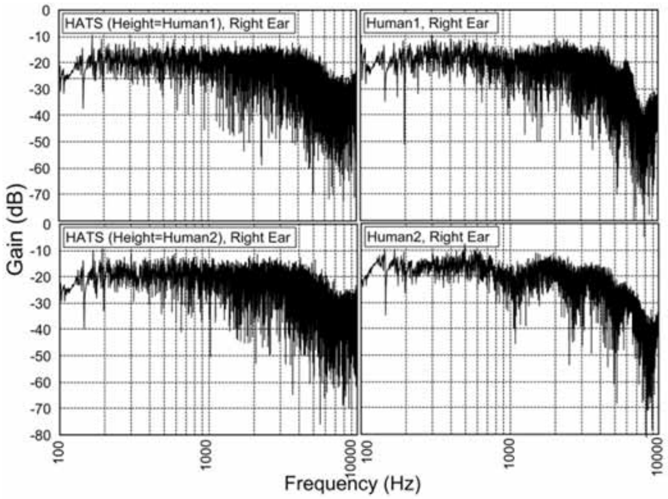

Figure 4. Magnitude of the transfer function of OBRIRs (right ear only) measured in the reverberant room.

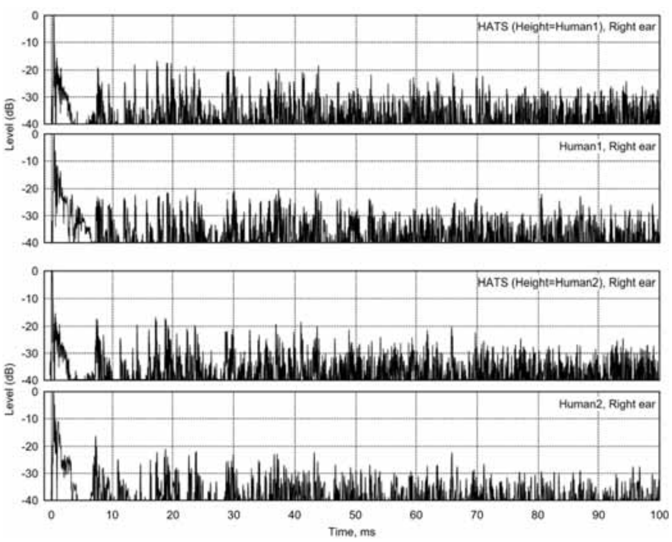

Figure 5. Normalised squared OBRIRs (right ear only) in decibels measured in the reverberant room, showing a comparison between the HATS and human measurements for the first 100 ms of each impulse response.

Comparison of the coarse temporal structure of the reverberant room OBRIRs between the HATS and human measurements can be done using reverberation time. However, since the source and receiver are very close, reverberation time was evaluated between -15 dB and -30 dB in the octave band reverse integration curves, rather than from the standard -5 dB point, because otherwise the reverberation time is artificially reduced due to the large drop in sound level after the direct sound. Results (Table 1) show a similar reverberation time spectrum shape, but with the human measurements reduced by a factor of about 0.82 relative to the HATS. The likely cause of this reduced reverberation time in the human measurements the time-varying directivity of human speech due to changes in the mouth shape and size, as well as minor head movements. Another possible contribution, at least in the low frequency range (where the wavelength is much larger than the distance between mouth and ear), is de-reverberation that could occur from not removing the reverberation from the mouth microphone in the human measurements – although the results do not show greater proportional reduction in reverberation in the low frequency range.

Table 1. Octave band reverberation times measured from the OBRIRs from the HATS and the humans in the reverberant room. Each value is the mean of two head heights and left and right ears. The final row gives the ratio of human to HATS reverberation time values. Values could not be derived reliably for humans in the 8 kHz octave band.

	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8k
HATS	4.03 s	3.44 s	2.70 s	2.30 s	1.94 s	1.43 s	1.05 s
Humans	3.42 s	2.76 s	2.24 s	1.89 s	1.53 s	1.22 s	
Humans/Hats	0.848	0.803	0.83	0.821	0.789	0.848	

This comparison between HATS and human measurements suggests that, while there might be some advantage in individualising measurements through measurements from real humans, further refinement of the measurement and analysis method would be required to yield results close to measurements from a HATS. Rather than using long duration speech, particular phonemes, including individual vowels over a range of fundamental frequencies, could be used [6]. Restricting the derivation of transfer function to one phoneme would reduce time variance due to the changing mouth shape. It might then be possible to switch impulse responses in a simulation system depending on the phoneme of the talker (although there are considerable practical obstacles to achieving this). However, we tested the concept of single phoneme measurement with the unvoiced phonemes ‘sss’ and ‘shh’ in an effort to obtain increased high frequency coherence, but the results were not substantially better than normal speech.

SIMULATION

The binaural simulation system is illustrated in Figure 6. A microphone near the mouth is used to obtain the voice signal, which is sent to a real-time convolver. The convolver uses a measured OBRIR, and the resulting convolved speech is presented to the subject via near-ear loudspeakers. The ear loudspeakers that we used are AKG K1000. These are more appropriate than conventional headphones because they provide little occlusion of the ears, thereby allowing the direct airborne sound to arrive from the mouth relatively undistorted. Binaural simulations can usually be improved by implementing a headphone correction filter, which is an inversion of the transfer function from the headphones to the in-ear measurement microphones. We used a 256-sample (sampling rate of 48 kHz) inverse filter (finite impulse response), which was combined with the OBRIR in the real-time convolver. The convolver had a latency of 66 samples (1.375 ms) between input and output, and combining this with the inverse filter yielded a latency of 3.7 ms. The start of the OBRIR was truncated by this system latency so that the simulated room reflections would arrive at the ears correctly delayed. The gain of the simulation system was adjusted to match the relationship between direct and reflected sound that existed in the reverberant room measurement.

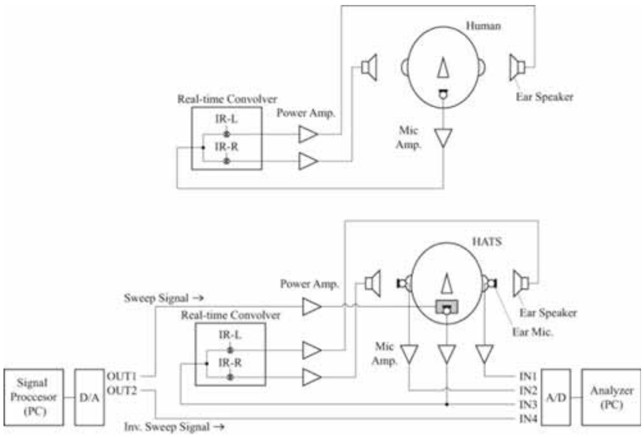

Figure 6. Block diagram of the simulation system (above), and of the test of the simulation system (below).

Measurements of the simulation system were made with the HATS in an anechoic room, as if it were a subject using the simulation system (Fig 7). A swept sinusoid was emitted from the mouth simulator so as to measure an impulse response at the microphone positions. The reverberant room measurement was used for the comparison.

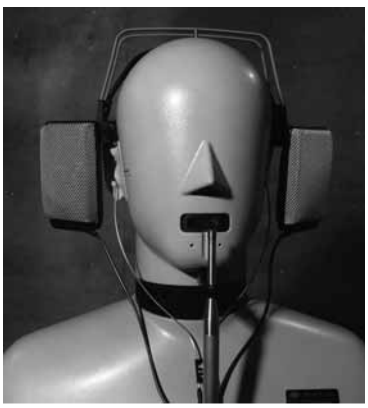

Figure 7. The AKG K1000 ear loudspeakers and the mouth microphone used for the simulation system, along with the head and torso simulator and ear microphones, which were used to test the simulation.

The results of the test show agreement between the simulation and the measured OBRIR, with some minor deviation immediately following the direct sound (which is at least partly due to the acoustic influence of the ear loudspeakers). The deviation at the start is likely to be masked by the direct sound (as it is -25 dB from the peak, and within 20 ms). The impulse response pattern that follows is a close match. Figure 8 shows this comparison for the first 100 ms of the left ear. We have not conducted a formal listening test comparing the original with the simulation, but informal listening has produced very positive responses.

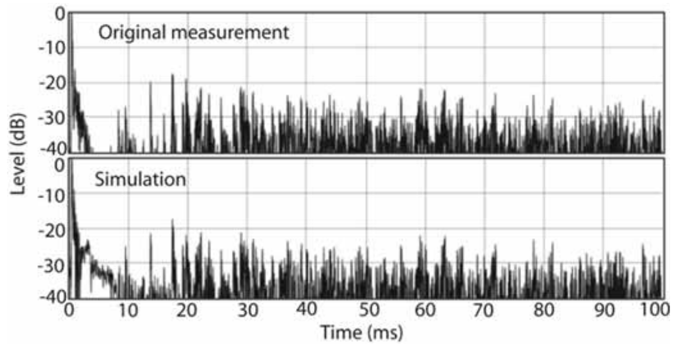

Figure 8. Comparison between the measured and simulated OBRIR (left ear only).

BINAURAL ROOM SCANNING

Binaural room scanning refers to a process of collecting and reproducing room impulse responses for a range of head orientations in a room [16]. Assuming that a subject will only make relatively constrained movements, measurements are made for horizontal rotations of a binaural recording device, at 2º intervals between -60º and +60º from the direction that is nominally straight ahead. The resulting sixty-one OBRIRs are switched in the real time convolver, with the selected OBRIR determined by the horizontal rotation of the subject's head. A head-tracking device is used to provide real time data to the computer so that the OBRIR is continually updated for the convolver.

Binaural room scanning has been used previously to simulate sound sources (such as loudspeakers) in rooms, with the listener at some distance from the source. In that application, maintaining the exocentric position of the sound source (independent of the head position) provides a great advantage in realism compared to simple head-locked binaural reproduction (where not using head tracking means that the auditory space moves with the head). The purpose of binaural room scanning is not to encourage large head movements, but rather to account for incidental head movements – and in this way it subtly provides a dramatic improvement in externalisation and realism. Only accounting for horizontal rotations is an approximation which is nevertheless effective because the predominant incidental head movements that strongly affect binaural hearing are horizontal rotations, which are also larger than head rotations typically observed around other axes. Measurements of the typical extent of incidental head rotations were collected for five human subjects engaged in talking, and the values sampled over a 3-second sampling period using a Polhemus FASTRAK system showed a standard deviation of 6.7º for the concatenated horizontal rotation data. Of course, during talking the human head is continuously shifting in orientation along its other two degrees of freedom, most often termed roll and pitch. Compared to the standard deviation of the measured horizontal rotation values, it was found that there was less than half that variation in head roll over the same 3-second sampling periods (the roll standard deviation was 2.3º). Although others have included a coupling of convolution with both head rotation and head pitch [17], coupling only horizontal rotations was included in our current implementation of head-tracked OBRIR reproduction, so that the collected room responses could be limited to a single sequence of only sixty-one OBRIRs.

Using binaural room scanning for OBRIRs is a little different to its conventional implementation, because the direct sound is not simulated (i.e., all that is being simulated is room reflections from the mouth source to the ears). Another difference is that the mouth and ears are all being rotated in the room, so that effects of moving the directional voice can be simulated. While changes in voice direction are probably only clearly discerned over large rotations (in environments with an uneven reflected soundfield, such as an auditorium stage) binaural room scanning provides a compelling reinforcement of externalization in the perception of the soundfield, and so is a cost-effective solution to rendering OBRIRs. At the time of writing, we have made binaural room scanned measurements of ten rooms from small to large for the purpose of experimental study of room acoustical features. Figure 9 gives an example of OBRIRs measured using binaural room scanning in a reverberant room, illustrating how the timing and strength of early reflections at each ear vary.

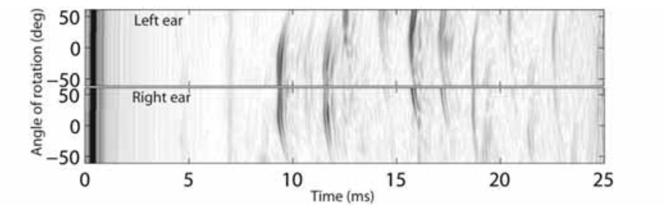

Figure 9. The first 25 ms of OBRIRs measured in a reverberant room using binaural room scanning from -60 to +60 degrees. The values shown are the absolute value of Hilbert-transformed waveforms, where black indicates high magnitude. The direct sound is seen just after 0 ms, and the floor reflection is a faint trace in the vicinity of 7 ms (ear height of 1.2 m).

CONCLUSIONS

This paper has outlined methods for measuring and simulating room acoustics using oral-binaural room impulse responses. Measurements can be made with minimal equipment from a speaking person, but using a head and torso simulator provides greater repeatability, a greatly shortened measurement time, removal of reverberation from the mouth microphone, and the possibility of implementing binaural room scanning. A simulation system using a mouth microphone, real-time convolver and ear loudspeakers produces signals at the ears that are close to those recorded in a real room. For such a system, the direct sound in the OBRIR is removed, which allows for a few milliseconds of latency in the simulation system.

Measurement and realistic simulation of OBRIRs can be used for the scientific study of the perception of room acoustics (for example, of loudness, clarity, stage support, speaking difficulty or room size). While binaural room scanning of horizontal head rotation provides some support for dynamic binaural perception, this is probably inadequate for some room acoustical studies, such as of human echo-location. As suggested by Pörschmann, there are also applications of such simulation systems beyond scientific studies, for example, in teleconferencing.

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Technical note

CONTROL METHODS FOR QUIET OPERATION OF PERMANENT MAGNET SYNCHRONOUS MOTORS

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INTRODUCTION

Noise and vibration production in electric motors is predominantly driven by pulsating torque. Though methods are available to isolate the pulsating torque from creating unwanted noise or vibrations, this research considers methods that act at the source and minimise pulsating torque.

Smooth torque production in electric motors usually requires careful motor design and manufacture. High precision manufacturing increases costs and the challenge is to make motors cheaply that have the smoothness and quietness of more expensive items.

One method to minimise design and manufacture restrictions in Permanent Magnet Synchronous Motors (PMSM) is to measure each motor's imperfections as it leaves the production line. This information can then be used to calculate specific current waveforms to compensate for these imperfections and allow quiet operation from an otherwise noisy motor. Current feedback control is used to ensure that these current waveforms are followed in operation.

The success of this method relies heavily on accurate measurement of the motor imperfections responsible for the pulsating torque. Without this accurate measurement, compensation and quiet operation is not possible.

The key objectives of this research were to:

- identify critical motor imperfections responsible for creating pulsating torque.
- develop a new method to accurately measure these imperfections.
- implement previously published current control methods using the new imperfection information to minimise pulsating torque.

MOTOR IMPERFECTIONS

Motor noise originates from pulsating torque. The main motor imperfections responsible for creating pulsating torque are cogging torque and current measurement error (offset and gain) [1].

Cogging Torque

Cogging torque in a PMSM is caused by the magnets on the rotor tending to align with the steel teeth on the stator rather than with the copper windings. Careful motor design (magnet arc adjustment, skewed magnets) can eliminate cogging torque. However, in practice, manufacturing errors (magnet placement, eccentricity and material property variation) will always lead to some cogging torque.

For one motor studied, Islam, Mir and Sebastian [2] suggested that if a magnet is misplaced from its "perfect" position by 1°, then the magnitude of the cogging torque can be increased by over three times.

Current

Current measurement error is normally considered as a combination of an offset and a gain error (i.e. the assumption is made that the output remains linear).

Chen, Namuduri and Mir [3] calculated that in the worst case, a 1% error in offset could lead to a 4% error in torque ripple and a 1% scaling error between sensors in different phases could lead to a 2.3% torque ripple.

EXPERIMENTAL SETUP

To ensure accurate torque measurement, a test setup was designed using finite element modal analysis to avoid any resonant frequencies in the measurement range. To verify the analysis, the system was analysed using an impact hammer and accelerometer. This testing confirmed that the first resonant frequency was at 700Hz, well above the upper measurement limit of 150Hz. The final test setup is shown in figure 1.

ACCURATE MEASUREMENT

Traditionally, motor imperfections (cogging and current errors) have been measured independently. Our research showed that a more effective method of measuring motor imperfections was to split up the total motor noise into individual components. This pulsating torque decoupling (PTD) approach was done by applying a least squares minimisation between the electromagnetic torque generated by the current and the measured torque. The cogging torque, which should be independent of current was then the residual, or remaining torque not explained by the electromagnetic torque.

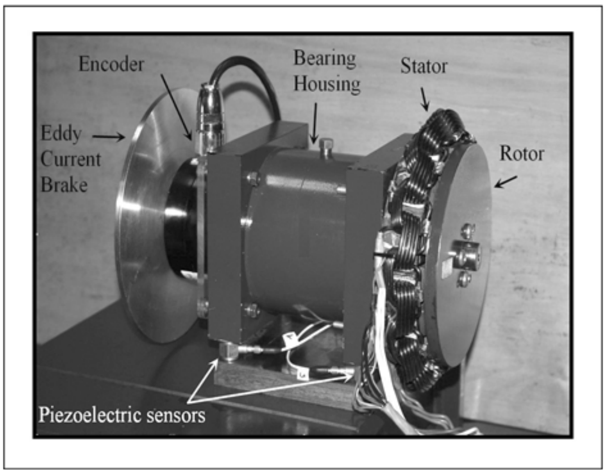

Figure 1. Charles Darwin University Motor Test Setup

The first pane in figure 2 shows the original pulsating torque. The second pane demonstrates how the torque can be split up into the "Xy" component that is due to the current errors and the "z" component that is predominantly the cogging torque. Some of the "z" component cannot be explained by cogging torque and this is shown in the third pane.

More detailed analysis is presented in [4].

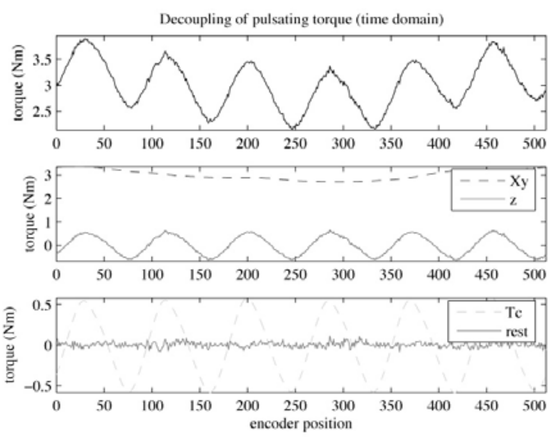

Figure 2. Torque decoupling (time domain)

RESULTS

For the test motor considered, the pulsating torque levels were calculated as the root mean squared (RMS) value of pulsating torque divided by the rated torque of the motor. A standard sinusoidal current was used as a baseline and resulted in 8 - 9% pulsating torque.

To see the potential improvement, five published methods for minimizing pulsating torque were compared. For each method, the performance was evaluated using information about motor imperfections determined by (1) traditional methods of measurement and (2) the PTD method.

Figure 3 shows that while there was a small variation between methods, the source of imperfection information was a much larger determinant for the total pulsating torque. The use of traditional methods, resulted in 3-4% pulsating torque while when the PTD method was used, the best method was capable of reducing the pulsating torque to less than 1% of rated torque.

This result demonstrated that if the motor imperfections can be adequately determined, an otherwise noisy motor can be made to run smoothly and quietly.

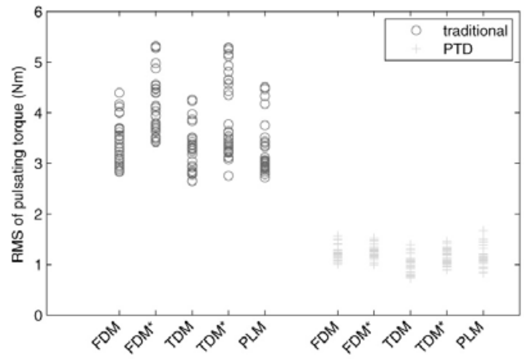

Figure 3. Pulsating torque method comparision

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Vacancy – General Secretary of Australian Acoustics Society

The society is currently seeking applications for the role of General Secretary. The appointee may be located in anywhere in Australia, but will require phone, fax, email and broadband connectivity. Please address applications to the President.

From the Editors

It's that time of year characterised by the impulsive sound of leather upon willow and the slowly modulated broadband of the surf.

And cicadas, we were reminded in a national park, once again surprised by the high level sound field that at first seems homogeneous. It isn't homogeneous on and below the metre scale, of course: female crickets have very good directional hearing at the frequencies of their species' song, which is what the whole energetically expensive business is all about. For large observers such as us, however, the local sources cease on close approach.

Cicada, you and I use a range of techniques for directional hearing but we hear only longitudinal waves. For infrasound, elephants 'hear' with their feet. In principle, the feet could detect the polarisation, whose plane gives directional information. Where soils are acoustically birefringent (and, to continue the analogy, dichroic) the delay and other differences between ordinary and extraordinary rays might give the pachyderms information about the distance, vegetation etc.

Polaroid sunglasses give us insight into the experience of bees, which see (and navigate by) the polarisation of light. We can think of no analogous simple tool to allow us to eavesdrop on polarised hearing. Still, as we distribute our mechanosensors over the beach sand, listening to the message from the surf, perhaps we'll have an inspiration.

Happy new year from

Marion Burgess, Emery Schubert,
John Smith and Joe Wolfe.

News

Prof Colin Hansen awarded Rayleigh Medal

Professor Colin Hansen, from Adelaide University has received the Rayleigh Medal from the UK Institute of Acoustics. This is a premier award from the Institute and was presented by John Hinton OBE, President of the Institute of Acoustics at the Euronoise 2009. The Rayleigh Medal is awarded without regard to age to persons of undoubted renown for outstanding contributions to acoustics. It is normally presented to a UK acoustician in even numbered years and an overseas acoustician in odd numbered years.

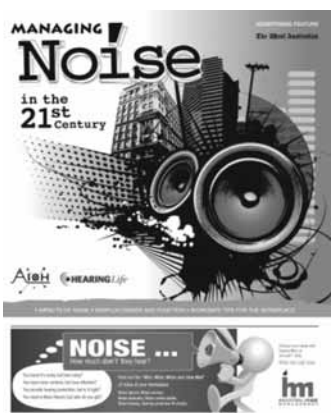
Colin was awarded on the basis of his pioneering work in the areas of active and passive noise and vibration control, in which he is a world leader as well as being one of the most well known and prolific contributors to research and development in the field of engineering acoustics.

Airmet and Allara combined

Airmet Scientific and Allara Instrument Hire combined on the 1st of November. The combined company will offer solutions for equipment and consumable sales, equipment life cycle management, technical support, service contract management, on site equipment service, fixed systems and short and long term rental. It will operate under the name Air-Met Scientific Pty Ltd with headquarters in Nunawading Victoria. Further information from www.airmet.com.au

WA Managing Noise in the 21st Century

During Noise Awareness Week in August 2009 a special liftout was featured in the West Australian newspaper. "Managing Noise in the 21st Century" comprised 20 pages and spanned Noise and Hearing Loss, Aids and Hearing Protection, and Industry and Workplace Issues. Sixteen contributors provided engaging and informative articles in an appealing format, guaranteed to raise awareness. The liftout can be viewed at: www.commerce.wa.gov.au/WorkSafe/PDF/Misc/WANoiseSupplement.pdf



NSW design guidelines

Following considerable discussion on a draft the NSW Department of Planning 'Development in Rail Corridors and Busy Roads – Interim Guideline' has been released. This document was developed with significant input from acoustic consultants and government agencies, and provides guidance on building design, internal layout and architectural principles to achieve an acceptable internal acoustic environment. This document is available from: www.planning.nsw.gov.au/rdaguidelines/documents/DevelopmentNearBusyRoadsandRailCorridors.pdf.

WA planning policy for road and rail noise

The WA Planning Commission has released "State Planning Policy 5.4: Road and Rail Transport Noise and Freight Considerations in Land Use Planning". The policy and accompanying guidelines apply to proposals for (a) new noise-sensitive developments near major transport routes, (b) new/upgraded railways or major roads, and (c) new freight handling facilities. The policy doesn't apply to noise from existing transport infrastructure in the absence of any major redevelopment. Documents are available at: www.planning.wa.gov.au/Plans+and+policies/Publications/State+planning+policies/default.aspx

NSW construction noise

The NSW Department of Environment Climate Change and Water released the Interim Construction Noise Guideline deals with the assessment of noise from construction activities and advises on best practice approaches to minimise noise impacts. An information sheet explains the key features of the guideline. The noise from reversing alarms is the source of the majority of complaints about noise from construction sites and the Department has now made available a review of alternatives to reversing 'beeper alarms'. All these documents are available from www.environment.nsw.gov.au/noise/constructnoise.htm

ACT noise measurement manual

The legislation in ACT has previously referred to the NSW noise measurement manual but has now released a Noise Management Manual that sets out the requirements for noise measurements. Its primarily aimed for the guidance of Authorised Officers but it is expected that all environmental noise measurements in ACT should be in accord with the manual and available from www.legislation.act.gov.au/di/2009-234/default.asp

Your noise is not the neighbours' choice

Simon Corbell, the ACT Minister for the Environment, Climate Change and Water, today urged all Canberra residents to consider their noise outputs, both at home and around town, which may affect other members of the community. The EPA information campaign with the theme 'Your noise is not the neighbours' choice' includes a dedicated website, cinema and radio advertising and a mail-out to ACT households and will last for a month leading into the warmer months. For more information on the program including viewing the short video visit http://www.noise.act.gov.au/

Australian Acoustical Society (NSW Division) Acoustics Prize

The Australian Acoustical Society (NSW Division) Acoustics Prize has been established this year to increase the exposure of acoustics to undergraduate students potentially interested in pursuing a career in acoustics. The prize recognises the best undergraduate student in an acoustics course at a university in NSW. The Prize consists of $250.00 and one year's subscription to the Australian Acoustical Society and will be awarded annually.

All universities with courses in acoustics were approached regarding the prize. In 2009 we established a Memorandum of Understanding regarding the prize with The University of New South Wales, The University of Sydney and Macquarie University. Congratulation to the following students who were awarded prizes:

- David Barker - MECH9325 Fundamentals of Noise course in the School of Mechanical and Manufacturing Engineering, The University of New South Wales;
- Anjali Chandhok - DESC9133 Architectural Acoustics Practice in the Faculty of Architecture, Design and Planning, The University of Sydney.

New Products

Better noise mapping with Dynamic Search

Braunstein + Berndt GmbH are pleased to announce the release of their newly developed Dynamic Search calculation method. Dynamic Search estimates the contribution for each receiver and ranks the influence of all sources. Only the sources important to the final result are calculated; the rest are estimated. This allows faster calculations on more data than hitherto thought possible.

The Dynamic Search method makes it possible to calculate huge noise maps with complex geometry, and to simulate details previously not possible in noise control programs. This new method was used to successfully complete the world's largest noise map, the European Noise Directive (END) noise mapping of 12,000 kilometers of railway throughout Germany. It included 11 GB of terrain information, eight million buildings and 36 million receiver points. All of this was calculated on four personal computers in less than 30 days run time using 32-bit WindowsXP.

For further information regarding the Dynamic Search Method and SoundPLAN please phone Marshall Day Acoustics Sydney on (02) 9282 9422.

Sensear short range communication ear muffs

One of the major draw backs of conventional hearing protection is that it blocks all sound, including speech, which means users are inclined to remove protection to communicate with each other, therefore exposing themselves to the risk of noise induced hearing loss (NIHL). Australian hearing protection company Sensear has a new addition to the innovative range of Sensear Smart Ear Muffs, the SM1xSR. Designed for clear, safe communication within short distances (50 metres) the SM1xSR utilises the power of SENS (Speech Enhancement, Noise Suppression) to provide a device-to-device communication solution in high noise environments. It allows users to converse easily with each other while wearing hearing protection without relying on two–way radio or cell phone. Incorporating Bluetooth technology, the new SR device also supports two-way or mobile phone communications, so the user can have the full communication capability beyond 50 m plus the added benefit of the face to face communication for which Sensear is renowned. Sensear thus offer the most advanced hearing protection and speech clarity solution available on the market today.

Further information from www.sensear.com, tel 1300 859 120 or info@sensear.com

Bridge-Alert-M and Bridge-Alert-F systems

Heggies has developed real time remote monitoring systems for Modular and Finger Plate Expansion Joints; critical bridge elements that are subject to potential failure. Expansion joint elements can become loose in service, dislodge and sometimes fail mechanically. The failure, loss and/or displacement of such bearings and other components may result in further damage to the bridge expansion joints. Detached joint segments can become a collision hazard for motorists and pose a real threat to human life. Such events have the potential to close major transport arteries or cause significant and costly traffic disruptions. Until the development of Heggies' Bridge-Alert-M and Bridge-Alert-F systems, visual inspections were the only method of checking the integrity of bridge expansion joints. These required lane closures and night inspections to increase safety and reduce traffic disruptions. Heggies innovative bridge monitoring systems detect loose and/or damaged joint segments by continuously analysing the sounds produced when the wheels of the passing traffic cross the joint. The detection algorithms have been fine-tuned to cope with the widely varying traffic densities, vehicle types and traffic speeds.

Further information, Jerome Rivory at Heggies, on 07 3858 4800 or jerome.rivory@heggies.com.

Meeting Reports

2009 AAS Annual Conference

The annual conference for the AAS was held in Adelaide in November with more than 120 participants. The program included the following keynote presentations;

- "Acoustical properties of ancient Chinese musical bells" by Jie Pan,
- "Sense from sensing sound" by Brian Ferguson,
- 'Industry and university partnerships in acoustic research - factors for success" by David Rennison,
- "Advanced passive treatment of low frequency sound and vibration" by Chris Fuller.

Two to three streams of contributed papers of high quality comprised the main program. Additionally there was a series of workshops on SoundPlan and VA-One, and also general discussion meetings on Wind Farms and Transit Oriented Development.

To emphasise the importance of the technical

program the AAS has instituted a number of prizes for the best papers in different categories and these were awarded as follows:

- Environmental Acoustics: Benjamin Hall for his paper on “CoRTN maximum noise emissions”
- Underwater Acoustics: D. Hanson, J. Antoni, G.P.Brown and R. Emslie for their paper on “Cyclostationarity for ship detection using passive sonar: progress towards a detection and identification framework”
- Industrial Acoustics: Gareth Forbes and Robert Randall for their paper on “Simulation of gas turbine blade vibration measurement from unsteady casing wall pressure”
- Research: Con Doolan for his paper on “Aeroacoustic simulation of bluff body noise using a hybrid statistical method”

It was a busy time with the large technical exhibition and social events on two evenings and the concluding BBQ lunch. The Welcome Cocktail Function was held in the great Hickinbotham Hall of the Wine Centre, featuring superb SA wines from 6 wine regions along with selections of gourmet canapes. The Young Adelaide Voices (‘Aurora’) provided an atmospheric choir performance in the tremendous acoustic space of the Hall and entertained the 100 people attending. The conference dinner commenced with champagne during the tram ride to Glenelg followed by a superb dinner. The after dinner speaker, Prof Barry Brook from University of Adelaide Institute for Climate Change & Sustainability, presented his views on the sustainable energy source for the future – modern nuclear power stations.

At the closing of the conference, the AAS president, Norm Broner, thanked the SA division for their efforts on this successful conference.

AAS Council Meeting This was held prior to the AAS annual conference and among many of the work items it was agreed that subscriptions will remain at their current rate for the coming year. The Excellence in Acoustics and Education Awards will be offered during 2010 so look out for the promotion of these.

AAS AGM This was unable to be held during the AAS annual conference and has been postponed to February 2010 when it will be held in conjunction with the first technical meeting for the year for the Victorian Division.

WA State Seminar 2009

Once again the annual AAS State seminar drew a good crowd. The August event was held at the Perth Zoo function centre, although “zoo noise” was not a focus apart from underwater bio-acoustics.

Presentations commenced with Iain Parnum (CMST) outlining work with DSTO on listening to beaked whales amongst other bio-sounds in the Coral Sea via high frequency noise loggers. John Macpherson (DEC) summarised a laborious process of reviewing the environmental noise regulations, culminating in a package of amendments that are soon to be released. Sandro Ghiotto (L3-Nautronix) outlined the underwater surveillance system for detecting and reporting acoustic events with the aid of submersible radio/satellite buoys, forming part of the Autonomous Undersea Surveillance Sensor Network. Moving back into the realm of non-submersibles, Luke Zoontjens (NDY) presented outcomes from several projects recently completed, including use of acoustic beam forming microphone arrays to assess complex vibro-acoustic issues with a locomotive engine, development of statistical environmental noise compliance tools, and design integration of diffusive panelling in sound studios.

Following the AGM and lunch, John Macpherson (DEC) and Brigitte Sieger (The West Australian) discussed environmental noise in the 21st century, in the context of increasing public awareness and raising the profile of the acoustics profession. Pam Gunn (Worksafe) reported on the progress of national model regulations for occupational noise, the Getting Heard project and concerns with the future of standards development.

Submersing for the final session, Rob McAuley (CMST) described the large sea-noise datasets from passive acoustic observatories along the WA coast, and discussed the vast possibilities for examining little-known animal habits through the IMOS system’s capabilities. Jessica Manea (L3-Nautronix) explained investigations on problems when hydrophones are located on a noisy platform, demonstrating the benefits of adaptive filtering for improved signal-to-noise ratios. The final presentation by Darryl McMahon (DSTO) provided an overview of the Future Submarine project and potential technologies.

WA Technical Meeting, Perth Concert Hall

The 1,700 seat Perth Concert Hall (PCH) is one of the most highly regarded orchestra performance spaces in the southern hemisphere, featuring a 3,000 pipe organ. Prof. A. Harold Marshall, in association with Warwick Mehaffey of the ABC, provided the original acoustic design about 40 years ago.

In October there was a relatively large turnout at the AAS technical meeting to see and hear the upgraded PCH audio systems. The resident AV manager of the facility gave an insightful talk on the design strategies and procurement methods used to ensure best outcomes on investment. This included the demand for greater flexibility and capacity for modern live performances being addressed through design that accounted for the complex effects provided by the original ornate ceiling coffers and wall features.

After an enjoyable dinner across the road, attendees were given full access to the stage and seating areas and also toured the rear amplifier and original sampling room whilst the system played a broad variety of music for comparison.

IINCE General Assembly

I had the privilege of attending the 35th General Assembly Meeting of the International Institute of Noise Control Engineering in Ottawa on the 23rd of August 2009 as representative of the Australian Acoustical Society. Current I-INCE president, Gilles Daigle, addressed the forum and confirmed that the Institute currently has 42 full and 8 sustaining members, that the institute remains in a healthy financial state and that its activities are ongoing. The Institute was pleased to report that the numbers for Internoise 2009 were a significant success, particularly given the financial strains on most North American and Canadian industries. As at 15th August 2009, Internoise 2009 had confirmed 626 registrants and 123 students. Internoise venues for 2010 (Portugal) and 2011 (Osaka) were advised, the theme for 2010 being “Noise and Sustainability”. More information is available at info@internoise 2010.org. The I-INCE committee encouraged acousticians to access the comprehensive database of technical papers going back to 1996. Papers are available via the I-INCE website or via www.noisenewsinternational.net.

Malcolm Pettigrew

NSW Division Student Grants

Two students were awarded grants to assist with their attendance at the AAS Annual Conference in Adelaide. Peter Jones presented a joint paper with Nicole Kessissoglou on “An evaluation of current commercial acoustic FEA software for modelling small complex muffler geometries: prediction vs experiment”. Michael Coats presented a joint paper with Nader Sawalhi and Bob Randall on “Extraction of tacho information from a vibration signal for improved synchronous averaging”

Future Conferences

AAS Annual Conference and ICA 2010

The AAS annual conference will be held as part of the International Congress on Acoustics to be held in Sydney 23 - 27 August 2010 at the Sydney Convention & Exhibition Centre. This congress is held every three years on behalf of the International Commission for Acoustics and provides the opportunity for all acousticians to meet and discuss recent advances in their fields of interest. The congress will feature a high standard technical program that will include plenary, distinguished, contributed and poster papers. Congress topics cover all areas of acoustics including:

Bioacoustics
Communication
Speech
Computational Acoustics
Electro-acoustics and Audio Engineering
Environmental Acoustics
Musical Acoustics
Non-linear Acoustics
Noise: Sources and control
Physical Acoustics
Physiological and Psychological Acoustics Room and Building Acoustics Structural Acoustics and Vibration Ultrasonics Underwater Acoustics

It is the opportunity to showcase the range of acoustics undertaken in Australia as well as to learn the latest findings from overseas. Submission of abstracts and registration information is available from the website along with accommodation, touring options and information on Sydney

For sponsorship opportunities and participation in the technical exposition for this prestigious event see the congress website

We look forward to your participation in ICA 2010 Sydney which will incorporate the AAS Annual Conference and we ask that you encourage colleagues from all around the world to participate in this event. For further details on any aspects of the congress, please check the congress website at www.ica2010sydney.org

ICSV17 Cairo

The Seventeenth International Congress on Sound and Vibration (ICSV17), sponsored by the International Institute of Acoustics and Vibration (IIAV) the Acoustical Society of Egypt, Ain Shams University, and the Nile University in cooperation with the International Union of Theoretical and Applied Mechanics (IUTAM), the American Society of Mechanical Engineers International (ASME International), and the Institution of Mechanical Engineers (I Mech E), will be held 18 - 22 July 2010 in Cairo, the capital of Egypt and the centre of Egyptian life and culture. The Congress venue, the Cairo International Conference Center (CICC), is Cairo's most comprehensive convention centre. Theoretical and experimental research papers in the fields of acoustics, noise, and vibration are invited for presentation. Companies are invited to take part in the ICSV17 Exhibition and sponsoring.

Further information from icsv17@icsv17.org

INTER-NOISE 2010, Lisbon

The 39th International Congress and Exposition on Noise Control Engineering will be held in Lisbon, Portugal, from 13-16 June, 2010. The Congress is sponsored by the International Institute of Noise Control Engineering (I-INCE), and is co-organized by the Portuguese Acoustical Society (SPA) and the Spanish Acoustical Society (SEA). The Congress venue will be the modern Lisbon Congress Centre, located on the north bank of the River Tagus in a new rehabilitated tourist waterfront area, full of amazing gardens and esplanades. The main theme of the Congress is Noise and Sustainability. The INTER-NOISE 2010 Congress will provide a great opportunity to make contacts and exchange ideas on various fields of acoustics and vibration, such as: Building and Environmental Acoustics, Education, Psycho and Physiological Acoustics, Speech, Measurement and Analysis, Equipments, Noise and Vibration Tools, Material and Technologies for Noise and Vibration Solutions. Many distinguished speakers will share their knowledge and experience with the participants.

Information from www.internoise2010.org.

Standards Australia

No change to the funding model

Standards publishing and the income stream from the sale of Australian Standards by SAI Global continue to be a point of discussion. It has been suggested to us that instead of asking stakeholders to help fund Standards development, Standards should be utilising the capital funds in our investment portfolio. This suggestion is based on the premise of Standards Australia being able to command a substantial publishing contract fee to replace these funds upon conclusion of the publishing contract with SAI Global in 2018 or 2023. The Board and Management of Standards Australia consider that given rapid changes in e-publishing and the push for more open standards development models, to proceed on the assumption that a significant contract fee could be realised would be imprudent. At a constructive meeting with the Commonwealth on 25 September 2009, the Commonwealth indicated the need for Standards development pathways to remain available where projects are principally funded by Standards Australia with in-kind support from contributors.

Bridge Street Office

Two years on, Standards Australia's move to Bridge Street has provided a modern facility, better technology and much easier access for Committee members and staff. Once located in the same building, the move has also helped to demonstrate that Standards Australia and SAI Global are completely separate organisations. In selecting the Bridge Street location, some 22 buildings were inspected from South Sydney to the lower North Shore. Importantly, the move was cost neutral and not one cent less has been spent on Standards development as a result of the move from our Sussex Street offices.From Standards Aust Bulletins 2009

Standards Australia Strategy.

The Strategy outlines strategic principles and initiatives to guide the development, coordination and alignment of voluntary standards in Australia. Stakeholders are invited to comment on the Strategy which can be viewed at www.standards.org.au

Draft Standards

Wind farms. A draft standard for public comment DR AS 4959 Acoustics - Measurement, prediction and assessment of noise from wind turbine generators has opened for combined procedure (combined commenting and balloting) and is available on the SAI Global website. While it is not a Standards Australia document those involved with wind farms may be interested in the recently released Environment Protection and Heritage Council Draft "National Wind Farm Development Guidelines" available from www.ephc.gov.au/sites/default/files/Draft_National_Wind_Farm_Development_Guidelines__Oct09.pdf

Construction site noise.

The comments on the draft standard "Guide to noise and vibration control on construction, demolition and maintenance sites" are being considered and the standard should be released soon.

FASTS

Fasts AGM 2009

The FASTS 2009 AGM was held on 24 November in Canberra and Neville Fletcher attended on behalf of the AAS. Following are some key points from his report and other documents presented at the meeting:

- Bradley Smith has resigned as Executive Director of FASTS and has been replaced by Annamaria Arabia. Ken Baldwin completed his term as President of FASTS and his place has been taken by Cathy Foley from CSIRO (immediate Past President of the Australian Institute of Physics). Bob Watts was elected as FASTS Vice President at the meeting.
- Graeme Durant, Director of Questacon, spoke of the decline in promulgation of science in Australia as exemplified by the decline in number of science journalists. He described plans to achieve "A Scientifically Engaged Australia".
- Ken Baldwin examined the performance of FASTS over the past year in terms of its Strategic Plan. Everything has gone well and the objectives have been met, with the Science Meets Parliament day involving many young scientists and proving cost-neutral because of supporting sponsorships. The publication of the report on Women in Science in Australia received considerable media coverage. During the year FASTS held meetings in every state capital except Darwin. A study is proposed on the profiles of young science graduates in relation to the skills desired by industry.

A paper by Ken Baldwyn addressed "When is Science Valid?" and could be summarised by:

- science works by systematically testing ideas against the evidence;
- evidence-based ideas are examined by peer review and published for further scrutiny in the scientific literature so that additional tests can be applied;
- scientific ideas are adopted when they usefully describe the world;
- when scientific ideas are widely accepted they become mainstream, and are applied until replaced by the widespread adoption of an alternative idea that makes better sense of the evidence;
- a scientific idea is validated when it is published in the peer-reviewed literature in the field, has stood up to further tests, and has been positively cited.

Scientist remuneration report

The *2009 APESMA/FASTS Professional Scientist Remuneration Survey Report* is now available. This is the 3rd year that APESMA and FASTS have presented the report. The survey was completed on-line in March/April of this year and finds the mean annual salaries for all respondents increased by 4.7% and median by 4.0%. In comparison CPI increased by 2.5% but average weekly ordinary time earnings index increased by 5.9%.

The report has extensive data by discipline, industry sector, level of employment, state, job function, highest science qualification, employment status and gender. In some of these categories, however, the number of respondents is quite low and thus the data should be viewed with a fair degree of caution. As the survey is also a commercial product for APESMA, electronic copies of the report are not available, but the AAS has a hard copy for the information of members. For further information contact m.burgess@adfa.edu.au.

Onus falls on industry to put a new face on science

This is extracted from an article by John Rice, Executive Director of the Australian Council of Deans of Science (ACDS) and Bradley Smith, Executive Director, Federation of Australian Scientific and Technological Sciences (FASTS). It was originally published in *Australia Financial Review,* Monday 7 September 2009.

A critical challenge for industry, universities and scientific organisations is to develop convincing profiles of scientific work in the 21st century, not least of all as the basis for attracting young people to science. There is a significant disconnect between the image of a research scientist and the working life of the majority of scientifically and technologically trained graduates. In part, this has come about because of a lack of attention to how the 'business models' of professional scientists and technologists have been utterly transformed in the past few decades. Education authorities, Government and science organisations promote the study of science on the basis that science and technology drives the global economy, and a scientifically skilled workforce is necessary for competitive advantage in both generating and adopting new technology. From here there is a tacit leap to the idea that there are lots of 'science' jobs out there. However, 'science' jobs have changed dramatically. Computer systems and new sophisticated instruments do much of the routine technical work that trained technicians used to do, resulting in both astonishing increases in the productivity of techno-science and changes in divisions of labour.

The erosion of funding per student in real terms over the past twenty-five years has reduced the capacity of universities to provide adequate and comprehensive technical, laboratory, field work and other skills in undergraduate science programs. However, it seems to us that the issues are more complex and interesting than simply that of skills shortages. At the heart of the matter seems to be a poorly articulated sense of how the nature of the technical workforce has changed over the last thirty years in the face of the globalisation, generalisation of ICT and technological change. In addition, there is a lack of clarity as to how employers identify and describe the kind of science background that they require of their recruits, and the roles that they have in mind for them.

One problem is how much we take scientific research as the only idea of real science, and any other kind of scientific worker, such as teachers or industrial workers, often consider themselves relegated to the 2nd division. More than 90% of science graduates do not become researchers, yet we know very little about where they go and how their training relates to career pathways. Nevertheless there does seem to be an imperative for convincing professional profiles of science at work. Industry groups have a key role in helping to articulate these roles. This needs to go beyond focusing on cocktails of skills and open up a more sophisticated and generative way to think about attributes of scientifically trained people in the workplace.

Australian Acoustical Society Annual Conference Acoustics 2010

In 2010, the annual conference of the Australian Acoustical Society will be combined with the International Congress on Acoustics in Sydney

23-27 August.

Full details at

www.ica2010sydney.org

(see advert opposite)

Diary

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

06 - 07 May, Paris
2nd International Symposium on Ambisonics and Spherical Acoustics.
ambisonics10.ircam.fr

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

22 – 25 June, Ancona, Italy
9th International Conference on Vibration Measurements by Laser and Noncontact Techniques
www.aivela.org

5 – 9 July, Istanbul
10th European Conference on Underwater Acoustics.
ecua-2010-istanbul.org

18 – 22 July, Cairo
ICSV17
www.icsv17.org

23 –27 August, Seattle
11th International Conference on Music Perception and Cognition
www.icad.org/node/3102

2010

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/

29 – 31 August, Auckland
Sustainability in Acoustics
issa.acoustics.ac.nz

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

23 – 27 August, Seattle
11th International Conference on Music Perception and Cognition
TBA

6 – 10 September, Graz
DAFx–10: 13th International Conference on Digital Audio Effects.
dafx.de

26 – 30 September, Makuhari, Japan
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

2012

20 – 25 March, Kyoto
IEEE International Conference on Acoustics, Speech, and Signal Processing.
www.icssp2012.com

2013

26 - 31 March, Vancouver
IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)
http://www.icassp2013.com

02 - 07 June, Montréal
21st International Congress on Acoustics (ICA 2013)
http://www.ica2013montreal.org

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

Acoustics Australia

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PAPERS

TECHNICAL NOTES

Obituary

Vale Eric Martin TAYLOR M.A.A.S.

29 March1930 to 13 November 2009

Eric grew up on a dairy farm in Tongala, Victoria and then in Melbourne. Eric completed his Diploma of Civil Engineering at the Melbourne Technical College (now the RMIT) and whilst working, completed his Diploma of Architecture at night school at the same institution. Eric worked for Godfery and Spowers Hughes Mewton & Lobb, architects and engineers, and worked on the significant projects the Dallas Brooks Centre. In 1969 the family moved to Canberra where Eric was responsible for the design and supervision of "Scarborough House" project. Godfery & Spowers in Canberra received many commissions for acoustical advice until eventually it became appropriate for Eric to establish his own consultancy.

Eric was admitted to the Australian Acoustical Society in the grade Member at the first Council Meeting in 1971. In 2004 Eric Taylor Acoustics Pty Ltd was taken over by Heggies Australia Pty Ltd.

Progressively Eric reduced his time at the office as he moved to retirement mode; and later as his health deteriorated. Eric died on November 13 at Clare Holland House and is survived by his wife Rosemary, his two sons Ross and Rohan; and daughter-in-law Cheryl and grandchildren Shannon, Damon and Kirsten. The funeral service was held in The Church of the Good Shepherd at Curtin; a lovely church with good acoustics designed by Eric Taylor.

Graeme Harding

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

3M AUSTRALIA
www.3m.com

ACOUSTIC RESEARCH LABORATORIES
www.acousticresearch.com.au

ACRAN
www.acran.com.au

ACU-VIB ELECTRONICS
www.acu-vib.com.au

ADAMSSON ENGINEERING
www.adamsson.com.au

ASSOCIATION OF AUSTRALIAN ACOUSTICAL CONSULTANTS
www.aaac.org.au

BORAL PLASTERBOARD
www.boral.com.au

BRUEL & KJAER AUSTRALIA
www.bksv.com.au

CSR BRADFORD INSULATION
www.csr.com.au/bradford

EMBELTON
www.embelton.com.au

ENERFLEX ENVIRONMENTAL
www.enerflexglobal.com

IAC COLPRO
www.colpro.com.au

NSW DEPT OF ENVIRONMENT & CLIMATE CHANGE
www.environment.nsw.gov.au

PEACE ENGINEERING
www.peaceengineering.com

PYROTEK SOUNDGARD
www.soundguard.com.au

SINCLAIR KNIGHT MERZ
www.skm.com.au

SOUND CONTROL
www.soundcontrol.com.au

SOUND SCIENCE
www.soundscience.com.au

VIPAC ENGINEERS AND SCIENTISTS
www.vipac.com.au

AUSTRALIAN ACOUSTICAL SOCIETY ENQUIRIES

NATIONAL MATTERS

* Notification of change of address
* Payment of annual subscription
* Proceedings of annual conferences

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SOCIETY SUBSCRIPTION RATES

For 2009/10 Financial Year:

Fellow and Member	$130.00
Graduate, Associate and Subscriber	$100.00
Retired	$40.00
Student	$30.00

Including GST

DIVISIONAL MATTERS

Enquiries regarding membership and sustaining membership should be directed to the appropriate State Division Secretary

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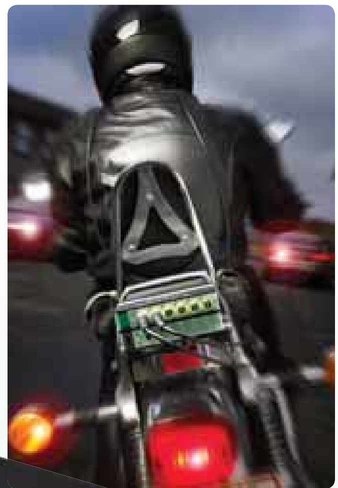
RUGGED, REMOTE RECORDING

LAN-XI NOTAR™

BG 1742 – 11



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