Acoustics Australia

- Digital Audio Coding
- Musical Sounds
- Dolphin Sonar
- A Dinner Gong
- MARCS Laboratories

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Vol 32 No 2

CONTENTS

August 2004

PAPERS

FEATURE

Cover illustration: MARCS at the Sydney Opera House (see page 69)


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From the President

The two year term of office for President is both long yet short, and as my term comes to an end, I thank you all for your support. I wish to thank Councillors, Divisional committees and many others who had input into the successful running of the Society. Without this continued donation of time and resources the society will not function.

Issues worth special mention through my term include the successful running of WESPAC8 - the 8th Western Pacific Acoustics Conference "Acoustics on the Move" held in Melbourne on April 7-9, 2003. This international conference was organised and hosted by the Victorian Division with the driving force of Charles Don. The successful bid for ICA 2010 is a significant achievement and will require the support of all the membership during the coming years. Acoustics Australia continued with its high quality publication and I must thank the Editorial Team that has produced it in a timely and professional manner. The continued time and effort invested by this team is not always apparent from the outside but we all should be aware that this quality of publication does not create itself. As with all voluntary tasks more help will always be welcome and all members can contribute by providing information, technical notes and drumming up some advertising support. The Member Survey and establishment of Priorities provide guidance for all those involved with the operation of the Society. The successful annual conference Acoustics 2002 in Adelaide is being followed by ACOUSTICS2004 at the Gold Coast. The theme is Transportation Noise & Vibration the New Millennium and it is being held 3–5 November 2004. Ensure you attend and catch up with your colleagues from around the country.

On the administration side there has been a lot of work updating the web page and a brief tour reveals a lot of information about how the society operates and what it has achieved. Most forms and reports are available through it and the Member database is now functioning. All members should log in to ensure that their details are up to date and change them when necessary. In future there will be only one database for contact details and e-mail addresses and if member details are incorrect it will be due to the member not maintaining them. Keep yours up to date. As we continue towards the "paperless office" the web and email will become the main means of communication between the Society, its members and Divisions.

Remember, there is always space for willing helpers, so become involved with your Society and lets all help it grow.

Ken Mikl

From the Editor

Welcome to the August 2004 issue! For those who have not actually thought about it, let me explain something of our "Editorial Philosophy". Acoustics Australia does not aim to be a Primary Research Journal in competition with Journal of the *Acoustical Society of America, Journal of Sound and Vibration, or Applied Acoustics.* Rather we aim to publish refereed papers of good scientific or technical quality that are chosen for their interest to members of the Society. We also include Technical Notes and discussion items in Acoustics Forum that are not sent to external referees but dealt with by the Editorial Committee. Finally, we publish items of general interest on new books and on new equipment, and news of recent and future events associated with the Society or with the international acoustics community.

Each year we publish a Special Issue dealing with a topic of interest and often relating to a conference held very recently. This year's special issue was published in April and contained five important papers on the subject of mechanical vibration, derived from a conference on that topic held on the Gold Coast in November 2003. The Special Issue for 2005 will be on Musical Acoustics and will rely fairly largely upon papers in the sessions organised by the Society and presented during the National Congress of the Australian Institute of Physics, to be held in Canberra in January 2005.

With the present issue we revert to our usual practice of presenting a balanced spread of articles ranging over various aspects of acoustics. There is one paper on the psychophysics of musical perception, one on compressed coding for the production of CDs, one on the remarkable sensitivity of the sonar of dolphins, and one on the practical aspects of making a dinner gong. In addition there is an extended paper that presents to the wider acoustics community the activities and plans of the MARCS Auditory Research Centre at the University of Western Sydney.

Finally, let me remind you all of the two great Australian conferences with acoustics content coming up in the not very distant future. In the period 3–5 November 2004 there is the Society's annual conference, to be held at Surfer's Paradise on the Gold Coast with the theme *Transport Noise and Vibration – The New Millennium* but also with sessions on Underwater Acoustics and on Architectural and Building Acoustics. Then from 31 January to 4 February 2005 in Canberra the Society is participating in the National Congress of the Australian Institute of Physics, which has the theme *Physics for the Nation,* and is organising sessions on Acoustics and Music. We hope to see you at one or both of these!

Finally, if you have something you would like to share with the rest of the Australian acoustics community – a new scientific or technical advance, a significant project completed, an interesting piece of news, or a controversial opinion for discussion, please get in touch with us. The Editorial Committee is waiting to hear from you!

Neville Fletcher

LOW-BITRATE CODING OF SOUND AND IMPLICATIONS FOR HIGH-QUALITY DIGITAL AUDIO*

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ABSTRACT. The Compact Disc launched in the early 1980s has set a worldwide standard for high-quality digital audio. Its sampling rate of 44.1 kHz and 16-bit word size appeared to provide enough bandwidth and dynamic range to capture every sound feature perceivable by the human ear. Although digital technology today is capable of delivering an even larger dynamic and spectral range at relatively little cost, there has been an ever-increasing need to produce digital sound of CD-equivalent quality at much lower bitrates than the CD's 1411 kbit/s. Examples of instances requiring lower bitrates are digital audio broadcast, fixed-head digital tape recorders, recordable mini-disc, multi-channel cinema sound, and internet audio. The original bit-compression schemes of transform and subband coding developed in the late 1980s, both exploiting the known perceptual limitations of the human auditory system, have now mostly been merged into a number of standardized complex coding systems that combine the best elements of the original schemes. Other techniques have been developed such as model-based parametric coding where not the waveform but rather the controlling parameters of a waveform-generating model are encoded and transmitted. Synthesis of the intended sound wave is then accomplished at the decoder end. This technique leads to additional bit savings and gives the end-user control over playback conditions like the positioning of sound sources. This feature is not only potentially useful for music playback in the home, but also for live electronic concerts, teleconferencing, and multi-user voice communication systems.

1. INTRODUCTION

High-quality audio is a concept that is not exactly defined and not always properly understood. To some it refers directly to physical similarity between a real sound field and its electro-acoustical reproduction. Within this viewpoint, acoustical knowledge and electronic technology are the only limiting factors preventing audio quality from being perfect. To others, however, audio quality refers to the audible similarity between a real-life sound event and an electronic reproduction. Given this viewpoint, the human auditory system with all its limitations has become an essential factor determining audio quality.

If one measures the locus of all pure tone frequency–intensity combinations that a young and healthy listener is able to perceive without experiencing discomfort, one typically finds a frequency range extending from 20 Hz to 20 kHz and an intensity range from about 10 to 110 dB. Frequency spectra of musical instruments and the human voice, considered all together, largely cover this frequency range. The dynamic range of a full symphony orchestra is, depending on size and concert hall acoustics, somewhere between 90 and 100 dB, although dynamic ranges of individual instruments are considerably smaller [1]. Figure 1 shows the typical frequency-intensity area for human hearing and, inside that area, a conservative estimate of the area covered by conventional music.

For digital storage and reproduction of sound with a bandwidth of 20 kHz, one needs a sampling frequency of at least 40 kHz. To reproduce sounds with a dynamic range of 96 dB without audible quantization noise, one needs a sample word size of at least 16 bits. These considerations led to the Compact Disc format proposed by Philips and Sony in the early 1980s, and standardized in 1987 [2]. The CD uses 16-bit words and a sampling frequency of 44.1 kHz. The total bitrate in 2-channel CD audio is therefore a little over 1.4 Mbits/s. On the market, the CD replaced the LP ('vinyl') format in a few years, and quickly became an informal standard for audio quality across the world.

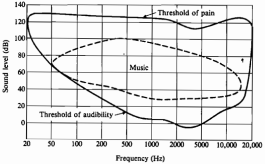

Figure 1: The auditory frequency and dynamic range. The 'music' range is represented by the dashed area. (From Rossing [42])

In subsequent years, storage and transmission of audio bits became progressively cheaper through advances in digital technology and, consequently, a mostly technology-driven

* Reprinted from the Proceedings of the International Congress on Acoustics ICA-2004, Kyoto, Japan

push towards an even higher digital audio quality standard ensued. In some studies, evidence was found that sound components well above 20 kHz can cause measurable effects on a-EEG activity [3], difference tone perception [4], and other perceptual features [5,6]. The Acoustic Renaissance for Audio (ARA) Foundation, residing at Meridian Audio Ltd in the UK, has been systematically pushing to establish stricter standards for high-quality audio on high-density CD carriers [7]. Critical evaluations of claims that sound components above 20 kHz are perceivable have, in some cases, uncovered artifacts like equipment distortion as the cause of the observed effects [8]. Therefore, it seems fair to conclude that psychophysical evidence in favor of a stricter and more bit-intensive CD standard is not overwhelming. It should be recognized, however, that for purposes of documenting and archiving sound recordings for later analysis, a more broadband format than the present CD standard would have its merits.

In many applications of digital audio other than compact disc or digital audio tape, there is an inherent bandwidth limitation that impedes or prevents straightforward application of the CD standard. Examples are digital audio broadcast (DAB), digital compact cassette (DCC), cinema sound (Sony Dynamic Digital Sound, Dolby ProLogic, Dolby Digital), and sounds transmitted over the internet. In some cases, the required audio bandwidth cannot be reached because of hardware limitations (DCC), and in other cases, there is bandwidth competition with other desirable features of the communication such as video quality or the number of available radio stations. In all of these cases, one ideally wants to reduce the audio bitrate of 1.4 Mb/s to something considerably lower, without affecting perceived sound quality. A straightforward approach of lowering the sampling rate would immediately be audible as a reduction of brightness due to the absence of high frequencies, whereas a reduction of the word size would result in (occasional) audibility of undesirable sound effects caused by quantization noise.

Bitrate reduction without audible sound degradation can be achieved by (a) reduction of statistical redundancy, and (b) minimization of perceptual irrelevance. Both processes are independent and cumulative. The first process removes redundancies in the code, is a strictly statistical mathematical operation, and results in *lossless* code (i.e., a code from which the original sound waveform is completely reconstructible). The second process exploits known limitations of our hearing system in order to remove inaudible sound elements and to selectively allow quantization noise that cannot be perceived. This results in a lossy, but at the same time, a *perceptually transparent* code. From such a code, the original sound waveform cannot be exactly reconstructed any more, but a listener will not be able to tell the difference between the original and the bitrate-reduced versions. Finally, if bitrates are reduced to a level where audible sound elements are removed or audible quantization noise is added, the result will be a *lossy* and *perceptually degraded* code.

2. TRANSPARENT CODING

A general scheme for perceptually transparent coding is shown in Figure 2. A running waveform is first subjected to a time—frequency analysis, shown in the top-left part of the figure, which can take several different forms depending on the coding strategy. Short time segments, comparable to the temporal resolution of our hearing system, are passed through a set of uniform or nonuniform. bandpass filters, as is done in *subband coding* [9]. The output parameters of this time-frequency analysis block in this case are the time responses of each subband to each input segment. *Transform coders* typically use some unitary transformation into the frequency domain, for instance a cosine transform [101. In this case, the output parameters are represented by a set of Fourier coefficients for each transformed time segment. Both subband and transform coders are purely waveform--based and do not assume any knowledge of how the waveform was produced.

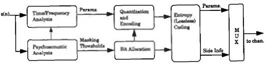

Figure 2: General scheme for perceptual entropy coding. (From Painter and Spanias [15])

Another analysis strategy is to assume that the sound waveform is, in principle, made up of simple building blocks that can be modeled and adequately described by a limited set of parameters. *Parametric coders* try to describe the input sound segments in terms of sinusoidal parameters [11], combinations of sinusoids and noise [12], or frequency-modulated sinusoids [13]. Output parameters then take the form of sinusoidal amplitudes and frequencies, noise intensities and bandwidths, and modulation indices. Another example is the linear predictive coder [14] that has become very successful in speech but less successful in music coding. It is based on the idea of a source-filter model of sound, where the source is either a periodic pulse or noise and the filter is an n-pole linear filter. Such a sound model seems a reasonable representation for the human voice and, to some extent, for harmonic musical instruments. It appears less appropriate as a model for inharmonic tonal instruments such as bells, gongs or other percussion instruments.

It is not the purpose of this presentation to provide a complete list of all the coding strategies that have been proposed or implemented. A detailed and elaborately documented review of this material was recently provided by Painter and Spanias [15].

The next general operation is the quantization and encoding of the parameters yielded by the time—frequency analysis. The quantization process is adaptive, where the number of bits available for encoding parameters of each time frame is determined by the results a psychoacoustic analysis shown in the lower portion of Fig. 2. This analysis typically assesses (1) which parameters can safely be dropped because their effect is inaudible, and (2) how much quantization noise can be allowed for signal parameters within that time frame. The principal psychoacoustic effect exploited is simultaneous masking, where the perception threshold for quantization noise in any frequency band is raised by the simultaneously presence of a sound signal [16]. Forward and backward masking [17], in which a sound signal masks quantization noise in succeeding or preceding time frames, can, in principle, also be included. In two--channel stereo sound, masked thresholds for quantization noise may be lowered by binaural masking release [18], resulting in a somewhat higher bit allocation in comparison with mono sound. On the other hand, signal information in the different sound channels can, to some extent, be combined, lowering the over-all bitrate [19,20]. For multi-channel 3D sound presentation, sound channels may also be partially combined [20], and even transformed into a single monaural channel plus a stream of binaural parameters, as is done in *Binaural Cue Coding* [21]. This leads, on the one hand, to a very low bitrate/sound quality ratio and, on the other hand, provides more flexible control over source positioning at the output in comparison with conventional multi-channel coding and transmission techniques.

In the final stage of the general encoder of Fig. 2, all mathematical and statistical redundancies that are left in the signal code are removed. This stage of the processor is *lossless,* whereas the previous stages of the coder yielded a *lossy* code. Hence, the end result is a *lossy* code. Whether or not this code is perceptually transparent depends primarily on the available bitrate (i.e., the number of bits that can be allocated in each time frame). It also depends, however, on details of the design of the time/frequency analysis block and on the finesse and correctness of the psychoacoustical model that is used.

If available bitrates are too low in comparison with the degree of sophistication of signal preprocessing and psychoacoustic analysis, coding artifacts will occur that are likely to affect the perceived sound quality in a negative way. Subjectively, such artifacts may sound like brief irregular bursts of audible noise, distortion of timbre, or disturbances in perceived location of sound source images. A particularly well-known and frequently occurring artifact is *pre-echo.* This happens when time frames are chosen too long with respect to aural time resolution, and a strong music signal onset (i.e., a castanet or bell sound) falls in the later portion of a time frame. Because allowable quantization noise estimated by the psychoacoustic model is spread over the entire time frame, it is likely to be audible just before the onset. To estimate obtainable limits for transparent coding, Johnston [22] developed a general computation scheme for *Perceptual Entropy* based on quantitative psychoacoustic models for simultaneous and non-simultaneous masking. He predicted that, for a variety of coders, bitrates could be typically reduced to about 2.1 bits/sample (at sampling rates above 40 kHz) before perceptual transparency is lost. Early listening tests with an MPEG-1 layer I coder and simple fixed--difference or adaptive forced choice procedures between full 16-bit/sample and bit-compressed signals, yielded difference detection thresholds between 2 and 3 bits/sample, dependent on music fragment [23]. Subsequent technical developments of combining the best elements of different coding strategies, however, have yielded hybrid coders that are able to produce perceptually transparent codes at rates less than 1 bit/sample [24].

3. EXAMPLE OF A SUBBAND CODER

This section describes the principles of a specific subband coding scheme known as MUSICAM [25] that became the basis for the ISO/IEC MPEGI layers I and II standard [26]. To help understand how this coder works, a schematic representation of a 3-tone music signal against a quantization noise background typically produced by a PCM (i.e., CD-type) coder is shown in Fig. 3. It shows the masked threshold of our auditory system, given three masking tones at 250, 1000 and 4000 Hz and at a sound pressure level of approximately 70 dB. One can see that at the three tone frequencies, auditory threshold has been raised by about 50 dB compared with the threshold in quiet. Quantization noise for some arbitrary bitrate has been represented by a spectrally flat band of noise, which is an oversimplification because quantization noise, although

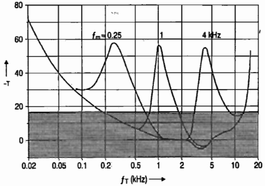

Figure 3: Threshold (LT) of a test tone in quiet and in the presence of three simultaneous masking tones. Shaded area represents arbitrary quantization noise.

spectrally broadband, is typically neither flat nor stationary. From the figure, it is clear that, for this situation, the quantization noise is audible in the valleys of the masking profile, at approximately 800, 2500, and 10000 Hz.

Figure 4 illustrates how one could ideally shape the quantization noise so that it is always below the masked threshold created by the signal components. Careful inspection reveals that the total quantization noise power in Fig. 4 is larger than that in Fig. 3, but nevertheless the noise shown in Fig. 4 will be inaudible. It is also clear that, because a music signal continuously changes with time, the quantization noise shaping should be adaptive and be updated very frequently.

In order to achieve this, the music signal is cut into short (typically 8-ms) time frames, and each time frame is decomposed into a set of responses of a bank of 32 constant-bandwidth (700-Hz) polyphase quadrature filters. Referring back to Fig. 1, this is done in the 'time/frequency analysis' block of the coder. The output 'parameters' are the (temporal) filter response waveforms. All responses are critically subsampled at 1400 Hz, so that at the output of the filter bank the total bitrate is still the same as at the input. Parallel to this filtering operation, the 'psychoacoustic analysis' block performs an FFT on the time frame and computes a masking profile from this spectral representation based on elementary psychoacoustic rules. Next, from the total number of bits available for that particular time frame, bits are allocated to each response signal in a manner that varies inversely with the

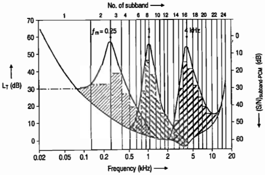

Figure 4: Same as Fig. 3, with quantization noise divided over 24 subbands. (From Wiese and Stoll [25])

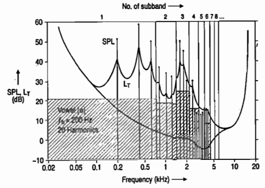

Figure 5: Spectrum of a vowel, masking pattern, and quantization noise as allocated in an MPEG 1 -layer II coder. (From Wiese and Stoll [25])

amount of masked threshold elevation at the center frequency of the passband for that signal. Since allocation of less bits implies generation of more quantization noise, the overall result will be that the spectral profile of the noise will generally conform to the masking profile of the signal, as is shown in Fig. 4.

Figure 5 illustrates the actual performance of the coder for a female-voice vowel segment comprised of 20 harmonics. Shown are (a) the spectral components of the vowel sound, (b) the masking profile of this vowel segment computed by the psychoacoustic analyzer, and (c) the spectrum of quantization noise allowed by the coder. One can observe that bitrate saving is achieved in two ways: (a) by allocating fewer bits in subbands where the masking profile is high, and (b) by allocating no bits at all where all signal components within a subband are inaudible, as is the case in subband 5. It is also clear that, if one wants to maintain perceptual transparency, any uncertainty about the correctness of the psychoacoustic model will require a larger safety margin between computed masked threshold and quantization noise levels. Therefore, the quality of the psychoacoustic model has a direct influence on the transparent bitrates that a coder can ultimately achieve. Therefore, continued efforts are made to further improve psychoacoustic masking models [27].

4. SUBJECTIVE EVALUATION OF SOUND QUALITY

Because perceptual entropy coders exploit limitations inherent to the human auditory system, classical measures of sound quality such as signal/noise ratio or total harmonic distortion are inadequate and irrelevant. Therefore, the performance of coding schemes is typically evaluated with the use of human listeners.

If one only wants to evaluate whether a sound from a particular coder at a certain bitrate is perceptually transparent, the simplest and most relevant method is a forced-choice discrimination test. The underlying thought here is that, if listeners cannot aurally detect the difference between a sound signal from an accepted quality standard such as a CD and the same sound signal from a lower-bitrate coder, then the two signals are perceptually equivalent in all respects and therefore the lower-bitrate code is perceptually transparent. The fastest form of such a discrimination test is an adaptive 2-down I-up comparison method [28] in which each trial consists of a presentation of a reference and a coded signal in random order. Listeners have to respond on each trial what the presentation order was, and typically will receive response feedback. A run will typically start with a low bitrate for the coded signal, so that differences between reference and coded signals are not difficult to perceive. After two successive correct responses, the bitrate of the coded signal is increased by a certain increment, making the discrimination task more difficult. This procedure is continued until an incorrect response is given, whereupon the bitrate of the coded signal is decreased by one increment. Somewhere the adaptive trace will end up oscillating between increments and decrements, at which point the adaptive run is ended. The bitrate at which oscillation of the trace occurs is considered the perceptual discrimination threshold and corresponds to a 71% correct discrimination level if fixed-difference runs were used.

Although forced-choice discrimination methods are straightforward in classical psychophysics, their application in the present case is not without problems. Firstly, meaningful comparison of music fragments requires some minimum length of these fragments, which interferes with short-term memory and often causes listener fatigue. Secondly, coding artifacts have many different perceptual forms and can occur at random times, so that the subject is in constant uncertainty about *when* to listen, *what* to listen *for* and, in cases of spatial sound, *where* to listen. To make matters worse, discrimination cues will not only quantitatively change when the bitrate is changed, but may also change qualitatively. This makes adaptive discrimination runs difficult to perform, and often causes unstable non-converging traces and significant learning effects.

A more popular test method is the double-blind A-B-C hidden reference paradigm, standardized in ITU-R Recommendation BS. 1116 [29], which is used for the evaluation of both perceptually transparent and degraded codes. On each trial, expert listeners hear three identical music fragments A, B, and C, where A is always the high-quality reference and B and C are, in random order, the reference and the coded signal. Listeners must (a) identify which of B and C is the hidden reference, and (b) rate the difference between B and C on a 41-point scale. A variant on this method is ITU-T Recommendation P.800/P.830, which uses a 7-point scale and is intended for more heavily degraded signals. Incorrect identification of the hidden reference automatically assigns the highest rating ('perceptually equivalent') to the coded signal for that trial. Average scale values assigned to perceived differences provide an indication of the amount of perceptual degradation of coded signals. This method of rating is important for situations where the highest possible quality of audio is desired, but perceptual transparency cannot be achieved because of technical limitations.

5. OBJECTIVE MEASURES OF SOUND QUALITY

Given the many problems encountered with subjective quality testing, including the finding that 'expert' listeners tend to be systematically biased in their sensitivity to particular types of perceptual degradation [30], it is not surprising that several attempts have been made to develop objective measures for perceived sound quality [31–33]. A fundamental problem underlying all these attempts is the fact that, since perceptual entropy coding is based on imperfections of the human ear, an accurate and complete model of human hearing is required to build a reliable and objective evaluation algorithm. It is obvious that a complete model of human hearing does not exist and will not exist for some time to come. Moreover, there is always a hidden danger that an 'objective' evaluation algorithm adopts the same or a similar hearing model as was used in a particular coding algorithm, since all perceptual entropy coders use some kind of psychoacoustic model. If this is the case, the evaluation algorithm is likely to favor the coder that uses the same hearing model, and will therefore be everything but 'objective'.

Despite the inherent pitfalls and potential shortcomings of objective evaluation methods, an international standard for objective sound quality testing has been developed and agreed upon between 1998 and 2001 by the International Telecommunications Union as Recommendation ITU-R BS.1387 [34]. This standard contains a psychoacoustic model yielding quantitative measures for the internal representation of signal features, and a cognitive model for describing feature extraction and combination. It incorporates six earlier proposed perception and cognition models, specifically the Disturbance IndeX (DIX) model [35], the Noise-to-Masked Ratio (NMR) model [32], the Objective Audio Signal Evaluation (OASE) model [36], the Perceptual Audio Quality Measure (PAQM) model [37], the PERCeptual EVALuation (PERCEVAL) model [38], and the Perceptual Objective Measurement (POM) model [39]. The essence of the evaluation model is that it always compares a degraded signal with a high-quality reference, and estimates specific loudness differences in each critical band following a classical method outlined by Zwicker and Feldtkeller in 1967 [16]. These quantities are then weighted and combined, and ultimately yield a single quantitative measure representing the perceptual distance between the reference and the coded signal.

6. CONCLUDING REMARKS

One should always keep in mind that lossy, perceptually transparent codes are meant to be listened to only. They should never be used for documentation purposes in which sounds are recorded and stored for physical analysis at a later time. The fact that such codes are transparent to our ears does not imply that they are transparent to acoustical or signal processing analysis procedures.

The original waveform-based coding techniques basically allow only playback of a music signal as it was recorded. Modem model-based parametric techniques allow, besides considerable bit savings, a great deal of flexibility and control on the playback site [40,41].

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DISCLAIMER

Opinions, interpretations and conclusions contained in this letter are those of the author and are not necessarily endorsed by the U.S. Army and/or the Department of Defense.

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ACOUSTICAL FEATURES OF MUSICAL SOUNDS

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ABSTRACT: In searching for significant features of musical sounds, it is necessary to convert basic physical spectral data into psychophysical measures. Preliminary analysis and organisation of the data takes place in the cochlea where incoming acoustic waves are filtered and converted into digital nerve impulses. These impulses are passed on to the brain where they are processed in a number of stages of increasing sophistication. Time and frequency analysis occurs simultaneously allowing continuous assessment of both the characteristics of the starting transient and the eventual 'steady' sound. The tristimulus method of analysis emulates this process by measuring the changes that occur with respect to both time and frequency.

1. BASIC PHYSICAL ANALYSIS

In order to study the acoustical output of a musical instrument, a set of representative sounds may be recorded and displayed as a time display, a frequency display or a combination of both using a sonograph [1]. Such records are purely physical in that they are independent of the properties of the ear and brain. For studies of instrument mechanics or the effects produced by rooms, this type of record is adequate. Quantities that may be measured include the mean sound level of the complete sound, the durations of the starting transient, steady state and decay together with spectrum levels at specific times. To have relevance to musical problems it is necessary to conduct further processing, in particular, to convert the measurements into psychophysical units and subject them to data reduction.

2. PSYCHOPHYSICAL MEASURES

When the processing functions of the ear and the brain are taken into account, features of the sound may be extracted that can be described as being psychophysical, implying that they are a combination of physical measurements and data originating from experiments involving human judgments. Preliminary processing takes place in the cochlea where the frequency sensitive hair cells are organised into a set of band-pass filters, called critical bands. When excited, the hair cells generate digital nerve impulses that are transmitted via the acoustic nerve to a series of auditory centres in the brain. The critical bands play a primary role in determining loudness, pitch and timbre. They determine spectral weighting, masking and some timing functions leading to measures including the mean loudness of the complete sound, band loudness spectra at specific times and band loudness derivatives as a function of time (these provide estimates of start-times and rise-rates for the partial tones present in the sound). The critical bands have time constants of approximately 10 ms at low frequencies falling to less than 1 ms at high frequencies. Their bandwidths are such that, for most musical sounds, the first five or six partial tones fall within separate bands – an important occurrence for the assessment of both pitch and timbre. For the higher partial tones, two or more fall within a particular band requiring their intensities to be combined. *[In practice, it is often convenient to use a set of one-third octave band filters. Response times are similar to those for critical bands except for the range below 400 Hz where one-third octave filter response times are longer than those for corresponding critical bands].*

Processing by the brain is an involved and only partially understood domain. It is generally agreed that the organisation of neurons in the cochlea by their frequency sensitivity is maintained through all the auditory stages up to the auditory cortex. The brain includes stages involving computation, comparison, correlation and integration coupled to a very sophisticated memory system. The result is the ability to perform rapid evaluation of a range of features of a sound. Pitch, loudness and timbre are assessed as well as more subtle aspects such as sharpness [2], roughness [3] and features of the important starting transients such as early noise, inharmonic components and dominant tones. The overall effect of all this activity is to produce a 'sound image' or 'acoustic template' that is stored for future reference and identification of sounds.

Data reduction plays an important role in reducing each image to its essential properties. For instance, in assessing the pitch of a complex sound, the first few partial tones that fall in separate critical bands play a dominant role in allowing the brain to form a harmonic template of the sound and assign a pitch [4]. This has the effect of reducing a large amount of spectral data to a single entity. Similarly, in the case of timbre, it is important that the first 5 or 6 partial tones fall in separate critical bands before the signals are passed on to the brain for assessment. In the tristimulus method described below, the large amount of spectral data associated with each note at any given time is reduced to 3 timbre coordinates. A further important property of the ear-brain system that increases overall processing economy is adaptation wherein certain cells respond mainly to changing signals, reducing their activity when no new information is presented.

A basic task for the investigator is to identify the essential features in each type of sound. For musical sounds, the relative roles of the 'steady sound' and the transients must be assessed. *[The term 'steady sound' is not particularly accurate since there are significant variations of loudness, pitch and timbre forming components of vibrato to which the brain is sensitive].* In the case of timbre, some investigators (including Helmholtz [5]), have maintained that the brain uses different procedures for assessing steady-state and transient

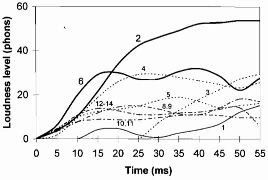

Fig 1 Starting transient growth curves for an open organ pipe (Principal 8', pitch G4, Marcussen organ, Spånga church, Stockholm). The time is measured from the first onset of sound from the pipe. For this pipe, there is some initial noise in the bands containing partial tones 8&9, 10&11, and 12-14. Initially, partial tones 6 and 2 (shown by heavy lines) are dominant. After a slow start, the fundamental increases in strength up to 200 ms. The second harmonic remains the strongest component.

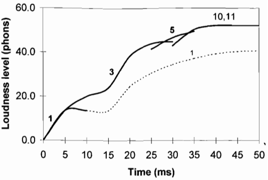

Fig 2 Dominant partials for the starting transient of an oboe note E4 flat. The order of dominance is the fundamental (0-5 ms), partial 3 (5-28 ms), partial 5 (28-34 ms), partials 10,11 (44 ms onwards). The later level of the fundamental is shown by the dotted curve, reaching a level of 40 phons in the steady state.

sounds. Other workers consider that these two aspects should be considered together, timbre being a composite property of the whole sound. It is not clear how the brain processes and coordinates information relating to the starting transient and the steady state as different parts of the brain take part. It will be shown below that both aspects of timbre can be combined in the same experimental procedure.

3. ANALYSIS PROCEDURE FOR LOUDNESS AND TIMBRE

For the analysis of a complete musical note, the sound is digitised, filtered into critical or one-third octave bands and stored in a computer for processing [6]. As the sound builds up, measurements are made at 5 ms intervals using a sliding Hanning window of 10 ms equivalent bandwidth [6]. Starting transients typically occupy between 30 and 80 ms with some string sounds extending to 300 ms. The output of each filter band is converted into linear loudness units (sones) and then into logarithmic loudness level units (phons). *[The logarithmic phon scale is more useful for displaying low-level values than the linear sone scale.]* A productive way to present the basic data is in the form of spectrum growth curves for the partial tones present.

In evaluating the critical band loudness response it is immaterial where a particular tone falls within a band or whether the tone is harmonic or not. If two or more tones fall within the same band they are not heard as separate tones; their intensities are summed to find the equivalent loudness within that band. For many musical instruments the partial tones are harmonic although they often deviate from this strict condition in the first few milliseconds as the standing wave system is being established. There are consequently initial non-harmonic tones present in some instruments, such as the 'mouth tones' heard in the flute and flue organ pipe [1]. Percussion instruments generally have non-harmonic partial tones. Under normal listening conditions, low-level initial tones are difficult to hear as they are easily masked by background noise, especially by wind noise in the case of organ pipes.

4. INITIAL PROCESSING OF LOUDNESS AND TIMBRE IN THE INNER EAR

Growth curves and dominant partials

An organ pipe has been chosen as an example of a musical sound having moderate harmonic development and which is not dependent on the manner of playing by a performer. A comparison will be made with a note played on a representative orchestral instrument, an oboe. Figure 1 shows a full set of growth curves from the onset of sound up to 55 ms for a Principal pipe, pitch G4, from the Marcussen organ in Spånga church, Stockholm. The G4 pipe was selected from the 8 ft Principal rank of pipes. The first 5 partials fall in separate critical bands, partials 6 and 7 in the next band, then 8 and 9 together, 10 and 11 , 12 to 14, 15 to 18, etc. For present purposes, there is too much information in Figure 1 from which essential features need to be extracted.

The dominant partial for a given time value, an important quantity in the assessment of the starting transient, is revealed by the envelope of the growth curves (shown as heavy curves). The dominant partials in the starting transient for the Principal G4 pipe are the 6th (from onset to 18 ms) , and the 2nd (from 18 ms onwards). The fundamental is slow to develop taking at least 200 ms to reach a level comparable with that of the 2nd harmonic.

There is no regular pattern in the order of appearance of the partial tones from pipe to pipe even within the same rank of pipes. For instance, for the C4 pipe from the same 8 ft

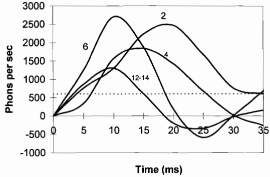

Fig 3 Derivative curves for the four fastest rising partials of the Principal G4 pipe. The dotted horizontal line at 600 phons/s is a threshold level for the measurement of start-times and times to steady state. The position of the peak of each curve gives the maximum rise-rate and the corresponding time.

Principal rank, the dominant partial order is 6th, 2nd , 3rd and fundamental while for the E4 pipe the order is 5th, 2nd , and fundamental. The voicing in this organ is in accord with baroque organ styling. A different dominant partial order would be expected from pipes voiced in a 'romantic' manner [1].

Figure 2 shows the starting transient dominant partials for a note (E4 flat) played on an oboe (data from Moorer and Grey [7]). The order of appearance of dominant partials is the fundamental (0-5 ms), 3rd (5-28 ms), 5th (28-34 ms) and partials 10 and 11 (44 ms onwards). As with the organ pipe, this is a record of a single sound and is not necessarily characteristic of all oboe notes. On wind and string instruments, there are considerable tonal differences between different notes played on the same instrument or between the same notes played by different players.

Properties of the starting transient

For a given sound, the rate of rise of loudness for each partial tone is different. There are two related parameters that are important in the assessment of the starting transient [5]: the maximum rise-rate and the time to reach a steady sound for each partial. Both quantities may be measured from derivative curves, based on the measured loudness differences at 5 ms intervals. Figure 3 shows smoothed derivative curves for the four fastest rising partials of the Principal G4 pipe. From these curves the following values are found:

	start-time (ms)	max rise-rate (phons/s)	time to max rise-rate (ms)	time to steady sound (ms)
Partial #6	2.5	2750	10.5	18.5
Partials #12-14	3.0	1300	9.5	15
Partial #2	4.0	2500	19	33
Partial #4	6.5	1800	14	25.5

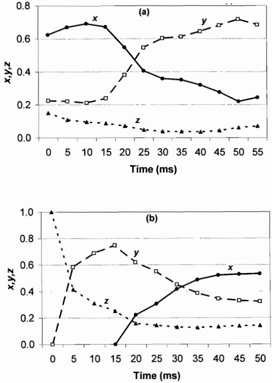

Fig 4 Variation of tristimulus coordinates x, y, z with time for (a) Principal G4 open organ pipe, (b) oboe note E4 flat. For (a), x starts high then decreases, y starts low then increases, z remains relatively steady. For (b), z starts high then falls rapidly, y increases rapidly at first while x starts late but then dominates.

The starting time for a given partial is taken as the time when the loudness level reaches 3 phons above background (a difference of 3 phons in a time interval of 5 ms is equivalent to 600 phons per second). Maximum rise-rate corresponds with the first peak of the curve (zero slope) while the duration of the starting transient corresponds with the return of the curve to the threshold value of 600 phons per second.

For the Principal G4 pipe, the starting transient duration is less than 50 ms for most partial tones but is approximately 120 ms for partial #2 and in excess of 200 ms for the fundamental. Further examples of starting transient durations include a stopped organ pipe (Gedackt G4) 40 ms, the previously quoted oboe note 35 ms, a reed organ pipe (Vox Humana G4) 30 ms, a clarinet note 45 ms and a viola note 65 ms.

Transition from starting transient to steady state

With musical sounds, time and frequency aspects need to be considered together. One of the remarkable properties of the ear and brain is their ability to process time and frequency information simultaneously. As Gabor [8,9] pointed out, there

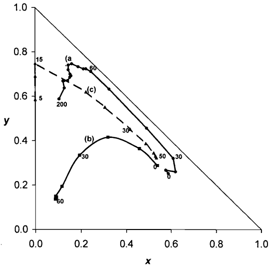

Fig 5 Tristimulus diagram with y plotted against x for (a) the Principal G4 pipe, (b) a Gedackt G4 pipe, and (c) the oboe note E4 flat. The Principal pipe starts with a bright sound (x=0.59, y=0.25, z=0.16) and finishes with stronger y and increased fundamental (x=0.10, y=0.60, z=0.30). The Gedackt pipe starts brightly (x=0.54, y=0.29, z=0.17) and finishes with a strong fundamental in the steady state after only 60 ms (x=0.09, y=0.14, z=0.77). The oboe note starts with fundamental only and finishes brightly (x=0.51, y=0.33, z=0.16).

is no limit to the accuracy of a time or frequency measurement carried out in isolation (1 variable) but when carried out simultaneously (2 variables) the results are limited by the uncertainty principle. Gabor introduced the concept of an 'elementary signal' or 'logon' of area Δt.Δf, where Δt is the effective duration and Δf is the effective frequency resolution. A logon is the smallest allowable quantum of information governed by the uncertainty principle $\Delta t.\Delta f \geq 1$. A profitable application of logons is in the analysis of musical starting transients [10].

During the starting transient, the signals change rapidly with time, hence maximum time resolution is required. In this phase, a typical logon for sampled filter measurements would have Δt = 10 ms and Δf = 100 Hz. Once the steady state is reached, changes with respect to time slow down. Sensitivity to pitch changes then becomes more important requiring maximum frequency resolution. In the steady state, accurate pitch recognition would require $\Delta f \leq 10$ ms with a consequent expansion of the time resolution to $\Delta t \geq 100$ ms.

It can be concluded that the cochlea filters play a fundamental role in analysing both the starting transient and steady state parts of a musical sound. The brain then interprets this basic data in terms of a number of more sophisticated concepts.

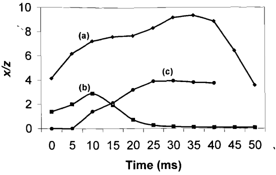

Fig 6 Graph of brightness (x/z) as a function of time for (a) the Principal G4 pipe, (b) the Gedackt G4 pipe, and (c) the oboe note E4 flat. For both organ pipes the brightness diminishes as the fundamental tone grows whereas the oboe note behaves in the opposite sense.

5. TIMBRE PROCESSING IN THE BRAIN

Tristimulus method

The human brain is particularly adept in reducing the large amount of spectral information involved in listening to music. In the musical assessment of timbre, even though at least 10 separate factors may be involved [10], three parameters have been found to lead to an adequate practical description [10, 11]. In the tristimulus method, band spectrum measurements are reduced to three normalised coordinates derived from the following three groups [12]:

Group 1: the loudness of the fundamental tone in sones,
Group 2: the loudness of partial tones 2, 3 and 4,
Group 3: the loudness of partial tones 5 and greater.

Because of its importance as a reference point both for pitch and timbre, the fundamental is the only tone included in group 1. For groups 2 and 3, the loudness of each group is computed using Stevens Mark 7 procedure [13].

Three tristimulus coordinates, x, y and z, are then defined as:

$$x = N(5,n)/N, \quad y = N(2,4)/N, \quad z = N(1)/N \qquad (1)$$

where $N(1)$ is the loudness of the fundamental, $N(2,4)$ is the loudness of partials 2 to 4, $N(5,n)$ is the loudness of partials 5 to n, and $N = N(1) + N(2,4) + N(5,n)$. The coordinates x, y and z describe timbre only; they are normalised for loudness (since $x+y+z=1$) and for pitch (since the fundamental tone is used as a reference pitch). Two-dimensional graphs may be drawn for any pair of coordinates such as x versus y or x versus z. The method may be applied to steady sounds or to changes occurring during starting transients.

Figure 4 shows the variation of x, y and z with time for (a) the Principal G4 open organ pipe and (b) the oboe note E4 flat. The tonal behaviour during the starting transient is quite different for these two sounds. For the Principal pipe, the proportion of high-frequency partials (x) falls rapidly up to 100 ms, the mid-frequencies (y) grow rapidly during the first 60 ms and then remain at a high level while z remains relatively low, reaching its maximum value at 200 ms. For the oboe note, z starts high, since the fundamental is the first partial to start, the mid-frequencies (y) rise rapidly then diminish while the high-frequency components (x) start slowly but dominate after 30 ms. The tone is bright in the steady state where the coordinate order is x, y and z.

Figure 5 shows a tristimulus diagram in which y is plotted against x for (a) the Principal G4 open pipe, (b) a Gedackt G4 stopped pipe by way of comparison, and (c) the oboe note. The Principal pipe starts with a bright sound (x = 0.59, y = 0.25, z = 0.18) and finishes in the steady state (200 ms or later) with much reduced x, a stronger y and an increase in z (x = 0.10, y = 0.60, z = 0.30). The Gedackt pipe starts brightly (x = 0.54, y = 0.29, z = 0.17) and finishes with a strong z in the steady state after only 60 ms (x = 0.09, y = 0.14, z = 0.77). The oboe note starts with fundamental only and finishes brightly at approximately 40 ms (x = 0.51, y = 0.33, z = 0.16).

Figure 6 shows the value of x/z as a function of time for the sounds shown in Fig 6. The ratio x/z measures the contribution of the upper partials compared with the fundamental and is a useful quantity describing the tonal balance or 'brightness' of the sound. *[While some writers dislike the term 'brightness' when applied to a sound, it is a term often used by musicians in contrast to a 'dull' sound. Bright sounds have high x/z values, dull sounds have low value.]* Both the Principal G4 and Gedackt G4 pipes show early high values that gradually decrease as the fundamental tone becomes more dominant. The behaviour is the reverse for the oboe note.

Sensitivity of the tristimulus method

The tristimulus coordinates are very sensitive to small changes in the spectrum of the sound. The method is particularly useful for studying small tonal changes due to differences in playing techniques or in instrumental conditions. It is possible to measure changes as small as the just noticeable differences (JND) in timbre for instrumental tones. According to Coltman [14], for skilled musicians, the JND in timbre for flute sounds corresponds to a change of only 1 dB in the level of a given harmonic. For less skilled listeners, the JND corresponds with a change of about 3 dB. Coltman found that the smallest JND values were observed with strong harmonics such as the second.

Consider the following spectrum (typical of a flute note played forte):

Harmonic:	1	2	3	4	5	6	7
Level (dB):	40	30	25	23	10	10	5

For this sound, x = 0.115, y = 0.442, z = 0.443.

If the level of the second harmonic is progressively reduced by 1, 2 and 3 dB, the y coordinate reduces by 0.015, 0.029 and 0.044 respectively with corresponding percentage changes of 3.3%, 6.6% and 9.9%. With stable sounds, the tristimulus method can record changes of the order of 2-3%.

Changes in the level of the second harmonic directly affect only data in the second group and hence the y coordinate. However, changes also occur in x and z as a consequence of the normalising condition $x + y + z = 1$.

6. CONCLUSIONS

Tonal properties of both the starting transient and the steady state can be measured using a single method in which sampled spectrum data are recorded at 5 ms intervals. Conversion into psychophysical measures facilitates interpretation of the roles played by the cochlea and the brain. Corresponding to frequency analysis performed by the cochlea, the growth of the partial tones with time, the presence of dominant partial tones and time measures associated with the starting transient may be extracted. Further analysis in the brain produces estimates of loudness and pitch (both one-dimensional) and of timbre (multi-dimensional). Application of a tristimulus method reduces timbre to three dimensions allowing for useful graphical presentation.

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THE SONAR OF DOLPHINS*

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ABSTRACT The sonar of dolphins has undergone evolutionary re-finement for millions of years and has evolved to be the premier sonar system for short range applications. It far surpasses the capability of technological sonar, i.e. the only sonar system the US Navy has to detect buried mines is a dolphin system. Echolocation experiments with captive animals have revealed much of the basic parameters of the dolphin sonar. Features such as signal characteristics, transmission and reception beam patterns, hearing and internal filtering properties will be discussed. Sonar detection range and discrimination capabilities will also be included. Recent measurements of echolocation signals used by wild dolphins have expanded our understanding of their sonar system and their utilization in the field. A capability to perform time-varying gain has been recently uncovered which is very different than that of a technological sonar. A model of killer whale foraging on chinook salmon will be examined in order to gain an understanding of the effectiveness of the sonar system in nature. The model will examine foraging in both quiet and noisy environments and will show that the echo levels are more than sufficient for prey detection at relatively long ranges.

1. INTRODUCTION

Research on the dolphin sonar system has been conducted over three decades and have increased our knowledge of their system. However, our knowledge has not matured to the stage at which a sonar can be constructed that can mimic the capabilities of the dolphin sonar system. Our research have shown that the properties of the dolphin sonar are fairly ordinary yet dolphins can perform astonishing target discrimination tasks. Most of the sonar characteristics in this paper are associated with the Atlantic bottlenose dolphin (*Tursiops truncates*).

2. CHARACTERISTICS OF THE DOLPHIN SONAR SYSTEM

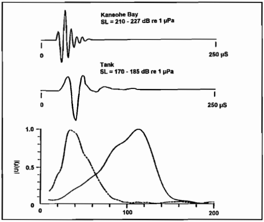

Fig. 1 Representation sonar signals of bottlenose dolphins

The broad frequency range and excellent sensitivity of hearing are two unique characteristics of dolphins. Dolphins can hear from 100 Hz to 150 kHz [1]. This is a range of 12 octaves and represents the widest frequency extent of any animal. The best sensitivity is about 40 dB re 1 mPa, which is comparable to low noise broadband hydrophones.

Bottlenose dolphins emit short broadband clicks having peak frequencies as high as 120-130 kHz [2]. Signals duration vary from 40 to 70 μs, having 4 to 10 positive excursions. Peak-to-peak source levels between 210 and 227 dB re 1 μPa have been measured [2]. Two sonar respectively. The directional projection and reception characteristics of bottlenose dolphin are not exceptional compared to many technological sonar.

The peripheral auditory system can be modeled as a bank of contiguous filters. At a frequency of 120 kHz, the Q of the filter is about 7 associated with a bandwidth about 17 kHz, not a very narrow filter [2].

3. DISCRIMINATION CAPABILITIES

Perhaps the most intriguing feature of the dolphin sonar system is the ability of echolocating dolphins to perform fine discrimination between different target. Three experiments will be discussed. The first involved blindfolded dolphin discriminating between the material composition and thickness of circular metallic plates [3].

Metallic Plate Discrimination

With the standard target being a 0.22 cm thick 30-cm diameter copper plate, three dolphins could discriminate between the standard and aluminum and brass plate of the same diameter and thickness. The dolphins could also discriminate copper plate of different wall thickness. Echoes from some of the plates obtained with a simulated dolphin echo system are shown in Fig. 2. When the incident signal was normal to the disc, the echoes showed no differences. However when the incident signal was 14° from normal, signals could enter the disc and propagate to the end and back, causing the echoes to have a structure related to both the thickness and material compo-sition of the plates.

* Reprinted from the Proceedings of the WESPAC-VIII, Melbourne, April 2003

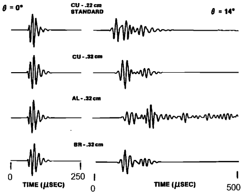

Fig. 2 Echoes from four plates at normal and 14° incidence. The 14° echoes are at least –30 dB weaker than normal incident echoes [2].

Wall Thickness Discrimination.

The second experiment involved the discrimination of wall thickness differences between a standard (6.35 mm wall, 3.81 cm OD, 12.7 cm length) and comparison cylinders with both thinner and thicker walls but the same OD and length [2]. At a range of 8 m the dolphin 75% correct response threshold occurred at wall thickness differences of -0.23 and +0.27 mm. Echoes from the standard and the comparison having a -0.3mm wall thickness difference are shown in Fig. 3. The dolphin probably used the 600 ns difference between the first and second highlights and/or the shift in the spectra.

Material Composition.

The third experiment involved a dolphin echo-locating at a range of 8 m and discriminating the material composition of solid 7.62-cm diameter spheres [4]. The dolphin could discriminate between the standard stainless steel sphere from spheres of the same diameter but composed of brass, aluminum and nylon. One again, differences in the echo structure of the targets were the probable cue.

4. USE OF SONAR IN THE WILD

Sonar experiments with captive dolphins and artificial targets have pro-vided much information on capabilities but did little toward under-standing their use of sonar in the wild. Signal measurements of wild dolphins have shown that source level increases in a 20 log R manner, where R is the target range. This variation of source level can be considered a form of time-varying gain for a sonar system that has little control of the receiver gain. Therefore, instead of varying the receiver gain, the transmission level is varied. When a dolphin forage for a fish school, the volume reverberation level of the school decreases as a function of 20 log R. Therefore, the level of the echoes from a fish school will be nearly constant with range.

Recently work my colleagues and I have performed involved modeling the use of sonar in foraging killer whales. Killer whales in British Columbia waters typically forage for chinook salmon that swim between a depth of 30 and 50 m.

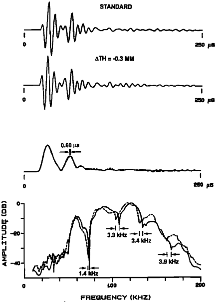

Fig. 3. Echoes from standard and comparison (-0.3 mm thinner wall) targets on the left. Envelopes of the echoes in top right and spectra in bottom right.

Our model is one in which the killer whale is at a 1 m depth and the salmon at 50 m directly ahead of the killer swimming away. Our field measurements showed signals had source levels that varied as 181.4 + 20 log R with a center frequency close to 50 kHz. By computing the target strength of a 0.7 m Chinook salmon as a function of angle we can estimate the levels of the echoes returning to the whale as a function of range. At a horizontal distance of 100 m between the whale and the salmon, the peak-to-peak echo level will be approximately 78 dB re 1 μPa. The threshold of hearing at 50 kHz is approximately 50 dB so that the echo is over 28 dB above the whale's threshold of hearing.

5. CONCLUSIONS

Although the dolphin auditory system is not highly tuned and the receiving and transmitting beam patterns are not very narrow, the dolphins can accomplish fine discrimination of targets. The use of broadband, short duration signals with good time resolution properties is probably the single most important feature that allow dolphins to make fine discrimination. The high mobility of dolphins and perhaps coupled to good spatial auditory memory are also important properties that enhance the dolphin discrimination capabilities. With their excellent sonar discrimination and detection capabilities, dolphins have no problems detecting prey at sufficiently long ranges to ensure successful foraging.

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DESIGNING AND MAKING A CEREMONIAL DINNER GONG

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ABSTRACT. The acoustic and aesthetic design of a ceremonial dinner gong for the fiftieth anniversary of University House at the Australian National University is described. The gong is made from polished stainless steel with a walnut frame constructed from the same timber used in building the Great Hall, and the shape of the frame echoes the slightly sloping walls of the Hall in which it will be housed and used.

1. INTRODUCTION

The Australian National University in Canberra was established in 1947 as the first and only purely post-graduate University in Australia, offering only PhD degrees at a time when these were offered by no other Australian University. In 1960 it amalgamated with the Canberra University College, then part of the University of Melbourne, to form the current Australian National University, which offers degrees at all levels, though the Research Schools of the original ANU still maintain something of a separate existence within the whole and cater only for graduate students.

In 1954 there was established, within the original ANU, a residential College for students and staff with the title of University House, and in March 2004 this institution celebrated its fiftieth anniversary. In the tradition of University Colleges in England, University House has always been a meeting place for students and senior members of the academic staff, and a feature of this collegiate life has been the formal dinners held each week in the Great Hall. This hall is notable for its architectural simplicity and subtlety, and also for the grand panelled painting covering the wall behind High Table, painted by renowned Australian artist Leonard French. To celebrate this anniversary, University House commissioned the design and construction of a ceremonial dinner gong to rest on High Table on formal occasions and to punctuate the procedures. The design and construction of this gong was delegated to the workshops of the Research School of Physical Sciences and Engineering (R.S.Phys.S.E.) in the University, and this note describes the steps taken to complete this honorific task.

2. ACOUSTIC DESIGN

A ceremonial dinner gong is a work of both acoustic and visual art and, while the two necessarily interact, they are also largely separate. In the first place, the sound expected of a dinner gong is not well defined. Provided it is pleasant and impressive, the sound is taken to characterise the particular gong. There are, however, certain acoustic principles that can guide the design in order that the resulting sound may be characterised as "pleasant and impressive".

The first design decision is the overall sound balance. In this, a gong is generally intermediate in sound between a bell, with well-defined tonal qualities, and a cymbal with an abundance of high and closely spaced vibrational modes. A bell achieves its tonal quality by being cast to a highly curved shape with very thick walls, so that wall stiffness dominates the behaviour and makes it almost completely linear. Under these conditions, the lower mode frequencies are well separated and clearly defined, and the bell-maker spends much effort in tuning their relative frequencies to near-harmonic ratios. At the other end of the scale, a cymbal is nearly flat and has free edges. When struck close to the edge with a hard stick, as is usual, many higher modes with nodal diameters are excited and the sound is "shimmering" rather than tonal. To add complication, the thinness of the cymbal means that vibrational amplitudes can be large relative to the thickness of the metal from which it is made, so that there is considerable nonlinear interaction, giving rise to sum and difference frequencies and to energy transfer between modes. All this contributes to the bright and incisive sound.

A gong sound lies between these two extremes, and can vary widely from one design to another. The metal sheet from which the gong is constructed is thicker than that of a cymbal but thinner than that of a bell, and the edge of a gong is almost invariably turned down to stiffen it against high-frequency modes with nodal diameters. In addition, a gong is usually struck with a padded hammer so that the impact is spread over an appreciable area, thus inhibiting the excitation of very high-frequency modes.

There is another feature of the sound to consider, and that is its evolution through time. The sound of a bell simply decays away, with higher partials decaying faster than those of lower frequency. A cymbal, on the other hand, tends to transfer energy from lower to higher modes because of vibrational nonlinearity, and it almost appears that the audible sound level increases momentarily before decaying. Gongs can be built to behave in either way. Large, thin, nearly flat gongs such as the Chinese tam-tam, the profile of which is

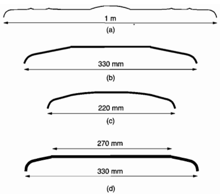

Figure 1. Profiles of (a) a Chinese tam-tam, (b) a downward-gliding opera gong, (c) an upward-gliding opera gong, (d) the gong constructed for University House. (Drawings are not to scale.)

shown in Fig. 1(a), can be nearly a metre across and not much more than 2mm in metal thickness, with a small central dome, a turned-down edge, and rings of hammered bumps. It is struck in the centre by a large and softly padded hammer, and actually behaves most impressively in this way, with the sound building up from a low-pitched boom to a shimmering rush over a period as long as two seconds.[1] The huge gong at the beginning of J. Arthur Rank films from the 1950s is a contrived example, though in this case it is not the real sound of the gong that we hear but rather an "artistic impression" constructed by the percussion section of a large English orchestra.

There is another feature of gongs made from metal of intermediate thickness and fairly flat shape that should be noted. If the gong surface is flat, under no initial tension, and essentially clamped round the edges by the turned-down lip of the gong shape as in Fig. 1 (b), then vibrational motion, as imparted by a hammer blow near the centre, will stretch the metal radially and introduce tension that tends to return the surface to a flat shape. This amplitude-dependent restoring force, which varies as the cube of the displacement, is in addition to the linear restoring force due to metal stiffness, so that the total restoring force F for a central displacement x looks like

$$F(x) \approx AEd^3a^{-4}x + BEda^{-4}x^3$$

where E is the Young's modulus of the plate material, d is its thickness, a is the gong diameter, and A and B are positive constants of order unity that depend upon the mode involved. The first term refers to bending stiffness and the second to displacement-induced tension. This means that the vibration frequency at large amplitude will be higher than that at small amplitude, so that the pitch of the gong will begin a little high and glide back towards its nominal value as the vibration decays, a sound that is striking in Chinese operas [2], but not very pleasant in a dinner gong. From the form of $F(x)$, such behaviour becomes noticeable once the amplitude of the vibration becomes comparable with the thickness of the metal from which the gong is made.

There is another Chinese gong that exhibits the opposite effect. If the metal is thin and the gong is very slightly domed to a height about equal to the metal thickness as in Fig. 1 (c), then the vibration frequency actually falls when the amplitude of the oscillation becomes comparable with the height of the dome. The physical analysis is rather more complicated [2], and derives from the fact that, in the absence of plate stiffness, the dome has two positions of stable equilibrium, normal and inverted, with an unstable equilibrium for a flat configuration in between. The sound of such a gong glides upwards as the sound decays, and this makes a dramatic contrast to the sound of the downward-gliding gong. Again, however, this effect is not appropriate for a dinner gong.

It turns out that there are two possible solutions that will maintain the gong pitch nearly level during the decay of the sound. One is to make the gong from rather thick metal so that the vibrational amplitude is always less than the metal thickness and the gong behaves almost in the same way as a bell. The second is to make the gong with a domed shape and to ensure that the height of the dome is much greater than the greatest vibrational amplitude that will be achieved. Other features of the sound, such as overtone pitches, also depend upon the metal thickness and shell shape, so that there are many things to be considered, and the design ultimately depends upon tradition and upon subjective judgment.

In developing the gong design for University House, the second of these approaches was initially tried, the dome height being about 10 mm, and this worked quite well. In the final design of Fig. 1 (d), however, the first approach was used because of the availability of appropriately thick sheet metal. There is another difference between the outcomes in the two cases, assuming that the gong diameter remains constant, and that is the effect on perceived pitch. The first solution, using thinner metal, leads to a lower pitch than the second, though the dome curvature tends to work against this.

The other acoustic adjustment that can be made is a little more subtle. If the gong is struck near its centre with a soft hammer, then several vibrational modes are excited and the listener will notice at least the lowest two of these, with the perceived pitch being determined largely by the frequency of the second mode, as in bells. It is important that the relation between the pitches of the first and second modes is heard as pleasant, which usually means a simple integer ratio between the frequencies. In our first experimental model, with slightly domed metal about 1.4 mm in thickness, the initial pitch relationship was not good, so this was modified by incising a series of deep rings into the metal at a distance about one-quarter of the radius from the centre. These incisions, being made near a node of the second mode but closer to an antinode of the first mode, shifted their relative frequencies to a much more pleasant relationship. For reasons of visual appeal and material availability, however, the gong was ultimately made from polished stainless steel sheet 2 mm in thickness and, because of the much greater stiffness, doming was no longer necessary. It turned out, also, that the frequency relation between the lower modes was initially pleasant, so that the incised-ring adjustment was no longer required. This feature, however, imparted a very striking and appropriate visual

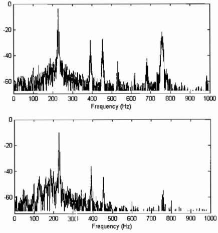

Figure 2. Relative sound pressure level spectra in dB about 1s after the strike (upper curve) and about 6s after the strike (lower curve).

appearance, and was retained for this reason in the form of lightly etched circles.

When the spectrum of the gong sound, excited by a central blow from the soft hammer, is examined, it is notable that it evolves significantly over a time of about 2 seconds. The bright initial sound has a pronounced fundamental at 226 Hz accompanied by a series of approximately evenly spaced upper partials, as shown in the upper panel of Fig. 2. Unfortunately there was no opportunity to examine the associated vibrational modes in detail, so that only a few general remarks can be made. The geometry of the gong is also too complex for a simple analysis to be helpful, though a finite-element analysis would give some insight. Where axially symmetric modes are concerned, the downturned rim of the gong acts simply as a small added mass and effectively the gong diameter a little with an essentially free condition on the extended boundary. For modes with nodal diameters, however, the rim is elastically stiff in the tangential direction, so that the edge of the gong is essentially clamped. Matters are then further complicated by the 30 mm sloped edge to the main gong plate.

The frequencies and relative frequencies of the modes are given in Table 1. If a mode with m nodal circles and n nodal diameters is represented by the notation (m,n), then approximate calculations suggest that the fundamental peak at 226 Hz is the (1,0) mode and that at 758 Hz the (2,0) mode, both being strongly excited by the central impact of the soft striker. The prominent second mode at 388 Hz is probably the antisymmetrical (1,1) mode excited by slight asymmetry in the strike. The strong modes at 454 and 682 Hz are puzzling in that they are almost exact integer multiples of the fundamental frequency. An intriguing possibility is that these are the result of nonlinearity and the abrupt change in slope of the gong surface towards its outer edge.[3]

The sound decay of the gong is a two-stage process. For the first two seconds or so, the decay rate is about 3 dB/s, while for the remaining time it is only about 1 dB/s, the faster initial decay being associated with the more rapid attenuation of the upper partials in the sound. The actual sound spectrum at two times after striking is shown in Fig. 2. The upper panel gives the spectrum about one second after the strike, while the lower panel is about six seconds after the strike. Clearly the upper partials decay much more rapidly than the fundamental.

The several harmonically related partials, indicated with an asterisk, define the subjective pitch of the gong, while the strong inharmonic partial with frequency ratio near 1.7 relative to the fundamental probably contributes largely to the slightly bell-like sound. As in carillon bells, the perceived pitch is more nearly that of the third partial at 454 Hz rather than that of the fundamental. The fact that there are several strong inharmonic partials in the initial strike note does not lead to a discordant sound, since these are well-spaced pure tones, and discords arise from rapid beating between the overtones of complex tones.[4]

Table 1. Prominent partials in the sound

Frequency (Hz)	226	388	454	530	618	682	758
Frequency ratio	1.0*	1.72	2.01*	2.35	2.73	3.02*	3.35
Harmonic ratio	1 : 1	—	2 : 1	—	—	3 : 1	—

3. VISUAL DESIGN

Visual design is a rather more subtle matter than acoustic design in this case. Gongs are traditionally usually made from brass or bronze that has been cast or hammered to shape, depending upon its thickness. This results in a mottled surface appearance and, in addition, these metals rapidly tarnish to a rather undistinguished patina. In contrast, a material such as stainless steel will retain its original surface appearance for a very long time without further attention. For this reason, and since the University House authorities had asked for an engraved crest on the gong, it was decided to make it from highly polished stainless steel sheet and to create a surface pattern by a mixture of etching and abrasion. The availability of appropriate sheet material of 2 mm thickness and the desirability having a flat surface for visual reasons also led to the adoption of the second acoustic design alternative. The final visual appearance of the gong incorporated the University House crest on a polished background, centred and surrounded by circles, further surrounded by broad area roughened by abrasion with fine glass beads, and then a final ring with concentric abrasion produced by fine emery paper.

A gong is not, however, simply the metal vibrating element, but involves also the means by which this element is supported. The design adopted recognised the geometry of the University House Great hall architecture, and in particular the gently sloping walls, and these are echoed in the shape of the walnut timber frame in which the metal gong is mounted.

Figure 3. The completed ceremonial gong, seen against the great panelled painting decorating the end wall of the Great Hall of University House. The design and construction team (L to R) comprised Tony Barling, Stephen Holgate, Steve Brooks, Neville Fletcher, and Ron Cruikshank, (absent: Anthony Mackey and Tony Cullen) all from R.S.Phys.S.E. at ANU. (Photograph by Tim Wetherell)

The mounting of the gong within the wooden frame is unobtrusive and simply requires, from an acoustic point of view, that it does not interfere with the vibration of the gong. The mount therefore consists of two loops of nylon cord passing through small holes in the gong rim and supported by hooks screwed into the wooden frame.

The completed gong, together with members of the design and construction team, is shown in the photograph of Fig. 3. The shape of the wooden frame is clearly seen to match that of the Great Hall, while the surface decoration of the metal gong shows up the University House crest to perfection.

4. CONSTRUCTION

The gong was constructed, as mentioned before, from 2 mm stainless steel sheet, polished on one side. To create the necessary profile with smoothly down-turned edges, a wooden disc about 30 cm in diameter and 20 mm thick was made that was a replica of the required inside shape. This was fixed in the spindle of an appropriately large lathe and the circular steel plate was clamped against it using a steel disk to prevent slippage and to protect the surface of the central area that was later to be decorated with the University House crest. The steel was then spun against the wooden template using a lubricant and a polished bronze forming tool to give a narrow sloping band and then a smoothly turned-down edge as shown in Fig. 1(d). The engraved design was produced externally by a screen printing type process, the design then being etched into the surface using an acid solution.

The timber frame was made in the School's carpentry shop. We were very fortunate in being able to secure some thick walnut planks that were left over from the original building of the Great Hall, and these were both symbolically appropriate and also an excellent match to various exposed timbers and furnishings in the Hall. The shape of the frame, as seen in Fig. 3, echoes the profile of the Great Hall, and is so constructed that the gong can be easily carried using the hand-grip at the top of the frame. The gong is supported in the frame by a light flexible cord passing through two holes in the gong rim and secured to two hooks in the frame. The striker for the gong was made from the same walnut as the frame, and the head from ironbark as a bobbin with a high-density foam layer wrapped with woven glass fibre fabric to give a narrow cylinder about 60 mm in diameter. It sits on two supports at the rear of the frame.

5. CONCLUSION

The ceremonial gong is the gift of Ms Pauline Griffin, Honorary Fellow and former Pro-Chancellor of the University, and was formally presented to University House at a commemorative dinner on March 31, 2004. It will repose in a specially built cabinet near the High Table, to be used on formal occasions, both in the Hall and elsewhere within University House. The team from the Research School workshop is proud to have been associated with its design and construction.

ACKNOWLEDGMENTS

It is a pleasure to recognise the contributions of Ron Cruikshank, Stephen Holgate and Tony Barling (metalwork), of Steve Brooks and Anthony Mackey (woodwork), and of Tony Cullen (striker). Stephen Moore, from UNSW@ADFA, assisted with the spectral analysis.

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Feature Article

Note: Feature articles provide information about interesting and important acoustical organisations andfacilities in Australia. They are not formally peer-reviewed.

RESEARCH AT MARCS AUDITORY LABORATORIES, UNIVERSITY OF WESTERN SYDNEY

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ABSTRACT: MARCS Auditory Laboratories is a University Research Centre at the University of Western Sydney. MARCS specialises in research on speech, music, and auditory processes. This paper elaborates particular research strands in MARCS – Auditory Processes and Speech Technology, the Baby Lab, Communicative Musicality, Human Factors and Performance, Music Cognition, Second Language Acquisition, and Speech and Language, and describe current projects in each of these strands. This is followed by a brief history of the Centre and its role at UWS.

1. MARCS AUDITORY LABORATORIES

MARCS is a research centre for basic and applied research on auditory perception and cognition, with particular focus on the significant domains of speech and music. Since the inception of MARCS in 1999 it has developed a dynamic and vibrant research culture that has attracted researchers with common interests in auditory research. Our members' backgrounds vary - experimental psychology, various branches of linguistics, music, electrical engineering, and we work with people from a range of disciplines, including music, linguistics, phonetics, computer science, human performance, and engineering, using our skills in cognitive science, research design, statistical analysis, and computer programming. So, while MARCS is a broad church, it has a particular methodological leaning (laboratory-based experimental psychology and related approaches) and subject matter (auditory research) which direct our enterprise.

In this paper, we first discuss various representative research projects at MARCS, and follow this with some background information regarding the University of Western Sydney, and MARCS.

2. REPRESENTATIVE RESEARCH AT MARCS

A range of research is conducted at MARCS, and the breadth of work is conveniently collected under our 7 labs, the Auditory Processes and Speech Technology, the Baby Lab, Communicative Musicality, Human Factors and Performance, Music Cognition, Second Language Acquisition, and Speech and Language. Examples of recent research projects in these labs are provided in the following sections.

Auditory Processes and Speech Technology Lab

Lab Leader: Dr Jörg Buchholz

Modeling the auditory signal processing of reverberant sounds

The well-known cocktail party effect shows that we humans monitor various concurrent sounds in our environment, and parse their spatial and information content. For machines and in mathematical modelling, cocktail-party processors are often applied to separate the superimposed signals. Although such techniques show promising results in simple scenarios, they usually fail in realistic – especially reverberant – environments. However, the human listener is able to perform this complex signal separation task, by applying sophisticated monaural and binaural signal processing. Hence, it is very important to understand and to model these auditory processes in order to employ them in modern cocktail-party processors.

In this regard, the phenomenon of room reflection masking has been investigated in our lab and a model describing the underlying auditory processes has been proposed. The proposed model is composed of four main stages: (a) bandpass filterbank stage, (b) Signal Dependent Compression (SDC) stage, (c) Equalization-Cancellation (EC) stage, and, (d) decision device stage. According to this model-structure, the SDC stage represents the important monaural mechanisms, and the EC stage the important binaural mechanisms. The model has been adjusted to simulate known psychoacoustic data on a test reflection being masked by a direct sound as well as being masked by the direct sound plus an additional reflection.

With reference to the first masking condition, it was found that an auditory model, which is solely based on monaural processes, could successfully describe the involved auditory processes. With reference to the second condition, it was found

The MARCS artificial head "Gustav" listening to a performance in the Sydney Opera House.

that binaural processes also need to be considered, processes that seem to be successfully described by the employed EC-approach. Furthermore, the proposed auditory model has been employed to predict the masking effect of a complete room impulse response, revealing that monaural processes play an important role in such complicated conditions.

Baby Lab

Lab Leader: Dr Christine Kitamura

Reorganisation of speech perception in infancy

Young infants understand aspects of speech well before they can speak themselves. Newborns perceive a wide variety of speech sounds but then between 6 and 9 months they begin to focus especially on the sounds that are used in their own language and disregard those used in other languages. This has been found for consonants and vowels, but the development of the perception of tone in infancy has been relatively neglected. Tone, the use of pitch movements and associated cues to convey meaning at the lexical level, is used in Asian tone languages such as Cantonese, Madarin and Thai, various West African register languages, and pitch-accent languages such as Japanese and Swedish, languages spoken by over half the world's population. Now PhD student, Karen Mattock, using a reinforced head-turn procedure, has established that Chinese babies continue to focus on tone variations in words, as they get older, while Australian English babies (for whom lexical tone is phonologically irrelevant) reduce their attention to tones between 6 and 9 months. However, when the same tones are presented as musical pitches, Australian English infants' attention remains unchanged between 6 and 9 months. Further studies will be conducted on infants' use of tone information to learn new words. The results have implications for second language learning, identifying language deficits, designing appropriate interventions, and for machine recognition and synthesis of Chinese.

Communicative Musicality Lab

Lab Leader: Dr Stephen Malloch

Communication and education: the case of Teacherese

A major factor in children's early development is the nature of the interactions with those who feature significantly in their lives – primary care givers, peers and mentors. Recognition of the inter-connectedness of a child's cognitive, social and emotional development highlights the interpersonal or intersubjective nature of communication and education. What encourages a person to engage with another? Certainly the content of the exchange will be important, but what of the underlying dynamics of the interchange?

We investigated the relationship between rated degree of engagement between primary-school classes and their teachers and timing characteristics of the verbal exchanges in these classes. This research comes out of previous work suggesting that communication between a parent and a young infant takes place through the intentions (underlying impulses for action) and affect carried by the 'music-like' qualities of their joint vocalisations in combination with the joint 'dance-like' gestures of their bodies and facial movements within a shared sense of time – communicative behaviour called communicative musicality. Stephen Malloch and Rudi Crncec from the Communicative Musicality Lab teamed up with Catherine Scott from the School of Education at the University of New England. Using the model of *communicative musicality* as a theoretical framework, the aim was to investigate the hypothesis that the more engaged the students are with the teacher, the more 'harmonious' the classroom interaction. A more harmonious interaction was defined as one with fewer interruptions and overlapping turns and fewer instances of student-to-student talking while the teacher was talking. Video recordings were collected of three teachers from three schools instructing 7/8 year olds. Extracts from these video recordings were rated for class engagement by seven expert raters. Measurements were made of timing categories in the vocal interactions between teacher and students, and ratings and measures were statistically compared. Significant correlations were found between ratings and those vocal timing categories related to teacher-student interaction. As the communicative musicality model would predict, ratings of low engagement were associated with vocal timing that suggested a more disrupted classroom interaction style, and ratings of high engagement were associated with vocal timing that suggested a more 'harmonious' interaction style.

Human Factors and Performance Lab

Lab Leader: Dr Mark Wiggins

Information acquisition, expertise, and decision-making in advanced technology environments

There is considerable evidence to suggest that human operators rely on a series of visual, auditory, and/or tactile cues as the basis for their decisions. These cues are extracted from a complex array of stimuli within the operational environment and their significance is usually determined by previous experience. Although the importance of cues has been recognised for some time, the acquisition of cues and the way in which cues interact to impact performance has been difficult to establish. Our present research involves the development and evaluation of a theoretical model of cue acquisition and integration as the basis for improving human performance within applied industrial environments such as aviation. In our most recent study, 50 pilots were asked to conduct a simulated in-flight diversion, the information pertaining to which was presented in one of three formats. These formats were based on three different types of decision heuristics (rules of thumb). The results indicated that the successful integration of cue-based information was dependent upon a number of factors, including the proximity and perceived relationship between different forms of cue-based information. We also observed differences between experienced and inexperienced pilots in terms of the ease with which they perceived the task. Specifically, experienced pilots preferred an environment in which task-related information could be acquired quickly and efficiently, whereas inexperienced pilots preferred an approach in which they could plan the sequence with which information would be presented. While the latter was more time-consuming, it resulted in more accurate results than might otherwise have been the case. Overall, the results suggest that the process of information acquisition is mediated, in part, by previous experience in the domain. This has important implications for the future development of decision support systems.

Music Cognition Lab

Lab Leader: Dr Kate Stevens

Using novel and familiar melodies to investigate episodic memory for music

The Music Cognition Lab applies methods from experimental psychology to investigate perception, cognition, and production of music. Western tonal music is an effective stimulus and tool to examine general auditory perceptual and cognitive processes as it is highly structured, gives rise to learned expectancies, and can be either novel or familiar. Research topics covered by our six postgraduate research students include pitch and time perception, synchrony and additivity of visual and auditory cues in marimba performance, attentional capture for expressive movement, the role of contour in recognition of speech and music, and development of a psychometric scale for measuring psychological responses to dance and music.

A recently published experiment used novel and familiar melodies to investigate episodic memory for music. Melodies were presented either once or three timers, either on Day 1 or Day 2, and participants made judgments about the recency and frequency of presentation of the melodies. Differences emerged between accuracy of judgments of recency and frequency, and this interacted with whether melodies were novel or familiar. More specifically, the results indicated that episodic recognition of novel melodies is based more on a generalized "feeling of familiarity" than a specific episodic memory. Frequency information contributes more strongly to this generalized memory than recency information, and formation of an episodic memory for a melody depends either on the overall familiarity of the stimulus or the availability of a verbal label.

Other developments in the Music Cognition Lab include the validation of the Audience Response Tool (ART) developed by PhD student Ms Renee Glass. The ART records open-ended and rating scale responses to live performance of dance or music. Responses include cognitive, interpretative, aesthetic, and affective responses. The ART is now also available on hand-held computers and, as well as recording questionnaire data *after* a performance, real-time continuous measures can be made as a performance unfolds. The computerised version of the ART and continuous recording devices were trialed during a performance of Sue Healey's *Fine Line Terrain* at the Sydney Opera House on July 2, 2004.

Second Language Acquisition Lab

Lab Leader: Mr Bruno Di Biase

Italian L2 learning programs in primary schools

The Second Language Acquisition Laboratory conducts investigations in two broad areas: acquisition of second language (L2) in instructional environments, and bilingual first language acquisition. Both strongly relate to the multilingual nature of Australian (particularly urban) environments and have implications for the success of language instruction and learning in educational settings (especially with the increasing presence of overseas students), as well as for the maintenance of languages in bilingual families and community environments. The Lab is currently investigating acquisition of Chinese-English (Ruying Qi), Japanese L2 (Satomi Kawaguchi), Japanese-English (Yuki Itani-Adams), Arabic L2 (Stuart Campbell) Italian L2 and Spanish L2 (Di Biase), and English L2 (Campbell, Di Biase, Kawaguchi). The Lab's theoretical orientation is mainly based on Processability Theory (PT) devised by Pienemann and others. This provides a cross-linguistic instrument for measuring language development as well as a common point of reference for language processing in both corpus-based approaches and on-line experimentation.

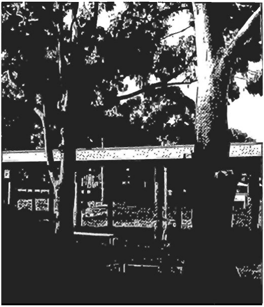
A large project, recently completed with ARC support, by Bruno Di Biase and Patrizia Berti along with other researchers and teachers involved a classroom-based longitudinal study of Italian L2 learning programs in three primary schools. The effectiveness of form-focusing techniques in instructed language development was assessed via videorecordings taken over an 18-month period, using metrics provided by PT. It was found that L2 programs in schools provide considerable lexical support but little, if any, grammatical development. This may be partly accounted for by the decidedly limited exposure children may have to the L2 in the school environment. However, on the other hand, the project shows that even moderate doses of (developmentally moderated) form-focused instruction appear to bootstrap grammatical development in the learners.

Speech and Language Lab

Lab Leader: Prof Denis Burnham

Seeing is Hearing

Optical information from facial movements of a talker contributes to speech perception not only when acoustic information is degraded or when the listener is hearing-



The delightfully rural campus of the University of Western Sydney in Bankstown.

impaired, but also when the acoustic information is clearly audible. This is most clearly shown in the classic McGurk effect, in which dubbing the auditory speech syllable [ba] onto the lip movements for [ga] results in the emergent perception of "da" or "tha". Intuitively one would assume that there is little or no visual speech information for the pitch variations associated with lexical tone in languages such as Cantonese and Thai. However, recently at MARCS labs we have found that Cantonese perceivers identify better than chance the Cantonese word being said from visual information alone, when given 6 alternatives with identical phonetic, but differing tonetic information. Additionally, visual speech information augments auditory discrimination for pairs of Cantonese words differing only in tone for Cantonese perceivers, for tone language perceivers unfamiliar with Cantonese tones (Thais), and even for non-tone language (Australian English) perceivers! Currently we are, with our collaborators at ATR labs in Japan, conducting speech production studies using a combination of signal processing, biological, and behavioural techniques (including OPTOTRAK to track the movement of the face during speech) to identify the essential characteristics of tone, affect, and identity and develop a comprehensive model of auditory-visual speech processing and communication. This research will have implications for understanding of the basis of auditory-visual perception and production in tonal languages and in affective communication, facilitate links between neurophysiological processes and auditory-visual speech processing; and contribute to applications in automatic person recognition, automatic speech recognition, text-to-speech systems, and talking head aids for the hearing impaired.

3. BACKGROUND

The University of Western Sydney

Established in 1989, the University of Western Sydney (UWS) early on realised that research concentrations in niche areas were both in line with emerging Department of Education, Science and Technology, and Australian Research Council (ARC) policy, and an effective way to realise this goal in a new university. UWS provides internal funding to its research centres, and encourages both internal grant applications through, and higher degree research places at these centres. By such means UWS has facilitated the development of a small number of research centres of excellence, and MARCS Labs is one of these.

MARCS Auditory Laboratories

UWS is situated on six campuses, and MARCS is situated on the Bankstown campus. MARCS has significant affiliations and collaborations with other UWS groups – the School of Psychology, the School of Languages and Linguistics, the Centre for Advanced Systems Engineering, and the Precision Robotics Research Group.

MARCS Auditory Laboratories grew out of the Macarthur Auditory Cognition Laboratory (MACL), which was formed in 1996 under the direction of Dr Kate Stevens. Identifying a niche in Australian research, Dr Stevens applied for University funding for MACL in 1997, obtained funding in 1998 and attracted Denis Burnham from the University of NSW as inaugural director. In 1999 MARCS began life as a Research Centre at the then University of Western Sydney, Macarthur, and continued as such until in 2001 when the three members of UWS amalgamated and MARCS became a University Research Centre at the new unified University of Western Sydney.

Support from the University of Western Sydney, and the College of Arts, Education and Social Sciences, and hard work by our collaborators and a dedicated bunch of academic, research and support staff in MARCS has allowed MARCS to grow and augment their external funding, increase PhD and Research Masters student load and timely completions, and build our equipment and lab infrastructure.

MARCS now consists of around 60 members, the Director, Professor Denis Burnham, and Deputy Director, Dr. Kate Stevens, the newly appointed Professor Cathi Best, 5 members from the School of Psychology and the School of Languages and Linguistics, 2 Research Fellows, 3 Postdoctoral Fellows, 1 Honorary Adjunct Professor, 3 Honorary Adjunct Fellows, 20 Higher Degree (PhD or Masters) Students, in any year around 10 Honours or Graduate Diploma Students, affiliates from the Precision Robotics Research Group, and the Centre for Advanced Systems Engineering, 6 Research Assistants, a Business Manager, a Technical Services Team Leader, 2 Software Engineers, and an Administrative Assistant. There are 15 testing laboratories and accompanying office space and recent university grants of space and funds have enabled the initiation of a building program (due for completion in September, 2004) which will double lab and office space. Facilities consist of sound proof booths, speech and music recording labs, speech/music analysis editing and synthesis facilities, purpose-built infant testing labs, simulation lab, movement analysis lab incorporating PEAK Motus and Optotrak movement tracking devices, a Portable Audience Response Facility, an anechoic chamber and 3D audio lab, and an EEG/ERP (event related potential) lab.

Organisations Associated with MARCS

MARCS is the administrative hub for the Australian Music & Psychology Society (AMPS). AMPS hosts seminars on the nexus between psychology and music, drawing an audience of staff and graduate students from psychology, music, education, physics, architecture, and acoustics.

MARCS Director Denis Burnham is the President of ASSTA, the Australian Speech Science and Technology Association, the peak speech science and technology association in Australia. He is also Vice Chairman of AVISA, the Auditory-VIsual Speech Association, which is a Special Interest Group of ISCA, the International Speech Communication Association (ASSTA's international counterpart).

An Invitation ...

MARCS is always ready to collaborate with people in areas of mutual interest; that is how we and others learn and push back the frontiers of science. We invite researchers, industry personnel, individuals to contact us about possible joint projects, consultancies, contract research. Consider the research reported above and find out more on our web page. You might like to visit MARCS and give an informal talk at one of our MMMs (Monday MARCS Meetings – see web address below), have a look around, and talk about your research and possible collaborations. (We are just 25 minutes from the city, and right next to the M5!) For all these or any other questions please contact MARCS Director, Professor Denis Burnham at d.burnham@uws.edu.au

We look forward to hearing from you.

Web Addresses

MARCS Auditory Laboratories,
UWS: http://marcs.uws.edu.au/
Monday MARCS Meetings:
http://marcs.uws.edu.au/events/mmm/index.htm
Centre for Advanced Systems Engineering, UWS:
http://www.uws.edu.au/research/researchcentres/case
Precision Robotics Research Group, UWS:
http://www.uws.edu.au/about/acadorg/cste/research/training/researchlinks/prrg#2
Australian Music & Psychology Society (AMPS):
http://marcs.uws.edu.au/links/amps/ See this website and/or contact the convenor, Kate Stevens (kj.stevens@uws.edu.au) to receive details of AMPS seminars.
Australian Speech Science and Technology Association:
http://www.assta.org/
AVISA, the Auditory-Visual Speech Association:
http://marcs.uws.edu.au/links/avisa/Default.htm
ISCA, the International Speech Communication Association:
http://www.isca-speech.org/

Publications

A list of publications from MARCS researchers, classified by field, can be viewed on the web-site http://marcs.uws.edu.au/

INTERLUDE

MY FAVOURITE ACOUSTICS BOOKS

For those of you who have not edited a journal such as *Acoustics Australia* — and I expect that is nearly everyone — there are matters that are hidden from view, or at least that should be. Have you ever noticed that every issue of the journal has a number of pages that is a multiple of four? How is this managed? It would perhaps be relatively simple to do if the Editors possessed a pile of material that was awaiting publication from which to select items of appropriate length and style, but unfortunately this is rarely the case. So, with Marion Burgess away overseas attending the INTERNOISE conference in Prague, I am faced with an issue that is one page short of the necessary quadruple multiple, and there is nothing in the IN-tray with which to fill it! Hence this interlude.

What, I asked myself, would be an appropriately interesting filler? As I look around my office I see bookshelves filled with volumes on many branches of physical science, and among them many volumes on acoustics. The collection, of course, reflects my own individual interests, so that there are large gaps — environmental acoustics being the most noticeable one. But what is left? I have a good collection of books dealing with the fundamentals of the subject, but particularly books on musical and biological acoustics. I hope you will be interested if I share with you my views on some of the best of these. If I have found them interesting and helpful, I am sure you will too.

General and Historical Acoustics

Several excellent books have been written about the history of acoustics, which goes back to the time of Pythagoras some 2500 years ago. Among those I find most informative and interesting are Frederick Hunt's *Origins in Acoustics* [1] and Robert Beyer's *Sounds of Our Times* [2]. In addition, of course, I have copies of Rayleigh's *The Theory of Sound* (1894), Helmholtz's *On the Sensations of Tone* (1885) and James Jeans' *Science and Music* (1937), all available as Dover reprints.

Acoustical Fundamentals

Apart from numerous conference proceedings, the most general set of books I have is the *Encyclopedia of Acoustics*, a modern comprehensive treatment of the subject edited by Malcolm Crocker [3]. This is where I go to find out in detail about many subjects with which I am not familiar. It covers everything from fairly basic to highly applied in a compact and informative manner.

If I wanted to remind myself of fundamentals, then I would consult *Fundamentals of Acoustics* by Kinsler and Frey, or the more recent edition with two extra authors [4], an easily readable and comprehensive book designed for advanced undergraduates. More technical and more detailed is the excellent *Acoustics* by Leo Beranek, reprinted in soft covers by the Acoustical Society of America [5].

Of course, if I want to delve quite deeply into fundamentals, then the best place is the classic *Vibration and Sound* by P.M. Morse [6], or perhaps the expanded version *Theoretical Acoustics* by Morse and Ingard [7], although these may be too mathematical for most people.

Musical Acoustics

Because this is one of my areas of special interest, my shelves are heavily laden with related books. Three books by Australians are notable among the general or introductory texts: *Measured Tones* by Ian Johnston [8], and the two volumes *Acoustics Applied to Music* and *Orchestral and Keyboard Instruments* by Howard Pollard [9]. I recommend all these highly.

Ascending the ladder of complexity come excellent books by Arthur Benade [10], John Backus [11] and Cornelius Nederveen [12], and then, for those not put off by mathematics, a very successful book [13] of which I am co-author, and which I often need to consult. For those interested in the acoustics of bowed-string instruments, there is a superb and comprehensive collection of papers published by the American Acoustical Society [14]. There are also many books on my shelves about individual instruments.

Music and mathematics have been closely linked since the time of Pythagoras, and a classic and innovative modern exposition, building on the work of Helmholtz, is given by William Sethares in *Tuning, Timbre, Spectrum, Scale* [15], complete with a CD of examples.

Biological and Auditory Acoustics

From a human perspective, acoustics is ultimately concerned with hearing, and I have several excellent books on this subject, of which I mention just *Psychology of Hearing* by Brian Moore [16]. For those concerned with the hearing-impaired, a most important recent book *Cochlear Implants* by Australian implant pioneer Graeme Clark [17] gives an excellent treatment.

Turning to the wider field of animal bioacoustics, there is an excellent survey from a biological perspective in Williams Stebbings' book *The Acoustic Sense of Animals* [18], and I have examined the whole subject from a more physical viewpoint in my own book *Acoustic Systems in Biology* [19].

Architecture

My interest in architectural acoustics largely focuses upon spaces such as concert halls designed for music listening, while I realise that many practical acousticians must be more concerned with factories, offices, and homes in noisy environments. In my reading within this limited area, two books stand out. The first is by Jurgen Meyer, *Acoustics and the Performance of Music*, and is soon to come out in an expanded edition if it has not already done so [20], and the second is by Leo Beranek, *Concert Halls and Opera Houses* [21]. Both are superbly interesting reading and highly recommended. I would just like to see one written about gothic cathedrals!

Conclusions

So this is a list of about twenty books on my shelves that have particularly caught my attention over the years. I hope that my mention of them here might suggest a few that would repay your own attention.

Neville Fletcher

References

1. F.V. Hunt *Origins in Acoustics* Acoustical Society of America, New York 1992
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3. M.J. Crocker (ed.) *Encyclopedia of Acoustics* (4 vols) John Wiley, New York 1997
4. L.E. Kinsler, A.R. Frey, A.B. Coppens and J.V. Sanders *Fundamentals of Acoustics* Wiley, New York 1982.
5. L.L. Beranek *Acoustics* Acoustical Society of America, New York 1986
6. P.M. Morse *Vibration and Sound* Acoustical Society of America, New York 1976
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8. I. Johnston *Measured Tones* Institute of Physics Publishing, Bristol and Philadelphia 2002
9. H.F Pollard *Acoustics Applied to Music* (1999) and *Orchestral and Keyboard Instruments* (2002) H.F. Pollard, Cronulla NSW
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11. J. Backus *The Acoustical Foundations of Music* John Murray, London 1970
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17. G. Clark *Cochlear Implants* AIP Press/Springer, New York 2003
18. W.C. Stebbins *The Acoustic Sense of Animals* Harvard University Press, Cambridge Mass, 1983
19. N.H. Fletcher *Acoustic Systems in Biology* Oxford University Press, New York, 1992
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21. L. Beranek *Concert Halls and Opera Houses* Springer-Verlag, New York 2004

Book Reviews

Fundamentals of Noise and Vibration Analysis for Engineers

2nd Edition
Michael Norton and Denis Karczub

Cambridge University Press, Cambridge, 2003, 652 pp (soft cover), ISBN 0521499135. Distributor Cambridge University Press, Private bag 31, Port Melbourne, Vic 3207, Tel 03 8671 1411, www.cambridge.org/aus/ Price A$150 approx.

The first edition of this book, with Norton as the sole author, was published in 1989 and quickly became recognised as a classic text. The review in Acoustics Australia, vol 18, no 3, p77 concluded with the comment that *"The book is a must for every engineering library"*. At that time this book filled a niche in that it deals with the inter-related fields of noise and vibration in the one volume – and this still applies today. This second edition maintains the structure of the original edition but is significantly updated to take into account the developments in the 1990s.

Each chapter has an introduction outlining the aim and placing it in context. References and nomenclature are included at the end of each chapter. This is fine for the references but it is a little tedious to find the nomenclature page for the relevant chapter when one is used to finding the nomenclature for the entire book in the one location. This minor annoyance was easily overcome with the use of plastic tags.

Chapter one provides the basis for understanding of mechanical vibrations. It is by far the largest chapter in the book and sets a strong foundation for the subsequent content. Chapter two deals with the fundamentals of sound waves through to flow acoustics. The third chapter provides a link between acoustics and vibration by discussing the interactions between sound waves and structures. Noise and vibration measurement and control procedures are discussed in Chapter 4. It is a challenge to cover all this in around 80 pages. Inevitably some material must be omitted and the limitations are given in the introduction to this chapter. It is surprising that in a book for engineers mufflers and outdoor propagation have been omitted. Chapters 5 on signal analysis and 6 on statistical energy analysis represent major updates from the first edition. Chapter 7 deals comprehensively with pipe flow noise and vibration although it is headed as a case study. The final chapter discusses noise and vibration as diagnostic tools and also has substantial updates to deal with modern integrated condition monitoring systems. Just before the appendices there are sets of problems for each of the chapters. An appendix gives the answers but without an explanation so if the number does not match the student must work out where they went wrong.

This book is well written, is easy to read, has good clear diagrams and contains a wealth of material. The highlighting of the integration of acoustics and vibration throughout the book is of great assistance. This second edition has been updated while keeping the original aim in mind. It is highly recommended for undergraduate students, post graduate students and practising engineers who undertake any work in noise and vibration.

Marion Burgess

Concert Halls and Opera Houses Music, Acoustics and Architecture

2nd Edition
Leo Beranek

Springer Verlag, USA, 2004, 661 pp (hard cover), ISBN 0-387-95524-0. Distributor DA Information Services, 648 Whitehorse Rd., Mitcham 3132, Tel 03 9210 7777, Fax 03 9210 7788, www.dadirect.com, Price A$136

This second edition is updated since the first edition published by the Acoustical Society of America in 1996. It is a wonderful book to browse through. The paper has a good feel to it and the uncommon font gives an artistic effect while being very clear. The layout is crisp, the diagrams are clear and complement the black and white photos.

Two short introductory sections provide an overview of music, acoustics and the language of musical acoustics. Then follows over 400 pages on 100 concert halls. Twenty six of these are in the US with the remainder from around the world. Those featured from our region include the Sydney Concert Hall and the Christchurch Town Hall. For each hall there is a short summary of the background of the hall including its intended uses. Any unusual constructional elements are discussed. Subjective assessments are given including in most cases impressions from the author following his attendance at concerts in the hall. The basic structural details are summarized and a table provides the technical details – both physical dimensions and some acoustic properties. More comprehensive details of the acoustic properties of all the halls are given in the Appendices. Plans and sections to essentially the same scale allow fast comparison of the various halls. The photographs are well selected to highlight the key features of each hall.

Following the details of the hundred concert halls are two chapters providing more information on the requirements and assessment of concert halls and of opera houses. The short bibliography focuses on the key books and papers which have contributed to the field.

It was in 1962 that 'Music Acoustics and Architecture' by Beranek was published. At that time it provided a summary and assessment of the performance of concert halls and became a valued reference book. This latest book builds upon that basis and integrates the knowledge that has been gained during the intervening decades. It is surely a required reference book for any students of architectural acoustics and for any consultants and designers involved with design and assessment of concert halls and opera houses. Anyone interested in acoustics and music would also enjoy reading this book.

Marion Burgess

Marion Burgess is a research officer in the School of Aerospace, Civil and Mechanical Engineering, UNSW at ADFA

News

John Wasserman has joined Wilkinson Murray as an Associate. John is a mechanical engineer with 16 years experience in the public and private sectors. He has been involved in the consulting industry for nine years providing consulting services in acoustics and vibration. John has worked with the NSW State Government for the last seven years, initially as Manager of Noise Assessments for the EPA and for the last two as Manager, Transport, for the Major Infrastructure Assessment Area of the Department of Infrastructure, Planning and Natural Resources (formerly Planning NSW).

Violin Competition

The Violin Society of America International Competition for new violins, violas, cellos, basses and their bows will be held 8-14 Nov 2004 in Portland Oregon. This competition is held biennially and offers the opportunity for makers from all countries to compete. The purpose is to inspire the creation of outstanding quality instruments and bows. More information on this event from www.vsa.to

Occupational Noise Update

The **Revised National Code of Practice** for Noise Management and Protection of Hearing at Work [NOHSC:2009 (2004)] 3rd Edition was declared on 2 June 2004 following a period of public comment and endorsement by the Workplace Relations Ministers ' Council (WRMC) in May 2004. The code is available from www.nohsc.gov.au/OHSInformation/NOHSCPublications/#5

A full review of the National Standard for Occupational Noise and National Code was endorsed by the WRMC at its May 2004 meeting. A Noise Review Reference Group formed from the state jurisdictions, the ACCI and the ACTU will commence work on the scope of the review in November 2004. Issues expected to be included are:

- Introduction of an action level at $L_{Aeq,8h}$ of 80 dB(A) and $L_{C,peak}$ of 135 dB(C);
- Non-auditory effects of noise;
- Combined effects of noise and ototoxic agents;
- Acoustic shock;
- Responsibilities of designers, manufacturers and suppliers of noise sources;
- Ultra- and infra-sound; and
- Matters raised in public comment on the 2004 Code.

Any comments should be forwarded to NOHSC for consideration during this review.

UK Draft Regulations and Guidance Material released for public comment by the UK Health and Safety Commission to implement the EU Physical Agents (Noise) Directive can be seen from: www.hse.gov.uk/consult/condocs/cd196.htm

Review of AS/NZS 1269:1998 Standards Australia will be considering the public comments received during the above review and the revised editions of the five parts of the standard are likely to be ready by the end of the year.

Pamela Gunn

Local Government Noise Guide

The NSW Department of Environment and Conservation (DEC) has released the Noise Guide for Local Government. The Guide aims to provide practical guidance to council officers in the day-to-day management of local noise problems and in the interpretation of existing policy and legislation. The Department has also released five new brochures addressing neighbourhood noise, barking dogs, vehicle noise, alarms and noise abatement orders. For more information check out www.epa.nsw.gov.au

En Health Report

The report on The Health Effects of Environmental Noise other than Hearing Loss has been released in May and is available for download from http://enhealth.nphp.gov.au/council/pubs/pdf/noise.pdf. This report was developed for the enHealth Council by the New South Wales Health Department, with funding provided by the Australian Government Department of Health and Ageing. The primary aim of the report is to provide a review of the health effects and the measures which can be directed at environmental noise management. This 88 page document will be of interest to all those involved with any aspect of environmental noise.

Acoustic Inventions

The ABC TV program, New Inventors, has featured at least two acoustic inventions. Chris Field was the successful winner of his heat for the Silenceair. This is a device that provides natural ventilation in buildings and permits more natural airflow than any other wall vent. It is an alternative ventilation system for both residential and commercial buildings. Chris now goes into the finals for the program

Another was from David Telfer with an antibio device for controlling bacteria. According to the program, a low frequency variable sound irritates bacteria and viruses in water and controls them. He stated that it does not kill baceria - just irritates them and it is used to purify water in swimming pools and for drinking.

You can read more about these and other inventions from www.abc.net.au/newinventors/ inventions/

INC Move

In May 2004 the INC office moved to 63-71 South Park Drive, South Dandenong, Victoria, Australia, 3175. Phone - +61 3 8710 7400 Fax - +61 3 8710 7499 www.inccorp.com.au

New Products

BRUEL & KJAER Two-plane Balancing Consultant

The recently released Two-plane Balancing Consultant Type 7790A is an intuitive and effective tool for in-situ (field) single-plane and two-plane balancing of rotating machinery. A task-orientated user interface guides you quickly and safely through the necessary steps for setting up, measuring, validating and reporting. Fast trim balancing using stored rotor data is also supported. The balance quality can be determined according to established balance quality grades (ISO 1940-1) or according to maximum machine vibrations. The balancing procedure can be FFT-based or based on order tracking for the most accurate results. The Type 7790A is one of many rotating machinery analysis and machine diagnostics tools offered on the PULSE™ multi-analyzer platform.

PULSE Lite Pocket Noise and Vibration Analysis

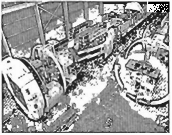
Brüel & Kjaer has released the PULSE™ Lite, a full featured analyser in a small size. It gives the power of a Brüel & Kjaer CPB, FFT or Run-up/Run-down analyser in a package small enough to fit in your laptop bag's accessory pocket. That is why it is nicknamed the "Pocket Analyzer". PULSE Lite is not only small and powerful; it is also easy to use. It incorporates over 60 years of Brüel & Kjaer's knowledge in sound and vibration testing and into a system that is simple to operate. PULSE Lite also has a growth path if your needs change. Data and projects from PULSE Lite are 100% compatible with PULSE so upgrading is painless.

Information from local Brüel & Kjær representative or www.bksv.com.au.

Meeting Reports

TUNNELLING NOISE AND VIBRATION MANAGEMENT

I am reminded of an entry in the UK yellow pages that tickled me some years ago. Under "boring", the entry said "see civil engineers"! Well, Barry Murray's presentation to the Christmas breakfast on 5th December last year discussed, among other things, tunnel boring machines (TBM's), but was anything but boring. In fact, in addition to a splendid breakfast and an opportunity to catch up with colleagues, around 60 AAS members were treated to a riveting overview of the noise and vibration issues that arise on projects that involve tunnel construction close to occupied buildings.



Barry described the applicable noise and vibration criteria and the groundborne vibration propagation mechanisms involved. He then showed examples of roadheaders and TBM's – the two dominant sources of tunnel construction vibration – and discussed typical levels observed at various distances from the workface. The machines are impressive (see TBM in accompanying picture) and the noise and vibration management challenges are significant.

As if that wasn't enough, Barry then illustrated the issues associated with the "surface works" that go hand in hand with tunnel construction projects. After the talk, delegates were able to join Barry on a visit to the Cross City Tunnel spoil-handling site at Bourke Street. The nearest residences are within 30 to 40m of this facility and, once again, the noise control challenges are significant given the 24hr tunnelling operation.

Dave Anderson

PERCEPTION AND HEARING

Members were treated to (not one, but) two interesting technical talks at the NSW Division meeting on Tuesday 22 June 2004. First Donald Woolford gave an overview of Auditory Perception, Psychoacoustics and Practice. The talk illustrated the wide range of practical uses for psychoacoustics (the basic science to quantify auditory sensation) from the design of hearing aids, measuring equipment, audio equipment and transducers, research into music perception, design of enclosures for music, speech and sound recording, to the preparation of standards documents.

The history of early work in psychophysics is fascinating, recording Kruger's observation in 1743 that the strength of a sensation grows in proportion to an increase in stimulus strength; Weber's work in 1834 in establishing the notion of just noticeable differences (JND's) in discriminating intensity changes; and Fechner's logarithmic law in 1860 relating sensation and stimulus magnitude. In the mid 1900's, Stevens proposed a general psychophysical law, and its application to three sensory modalities was presented. Earlier psychophysics relied more on comparisons between various stimuli, but modern technology and the computer has greatly increased possibilities.

Don distinguished between psychological and physiological acoustics, the overlap between the two disciplines, and the range of psychoacoustic parameters that include loudness, pitch, masking, timbre, localization and temporal aspects. The pure tone equal loudness curves first developed by Fletcher and Munson in 1933 were presented to show the relation between physically measurable sound pressure level and frequency in relation to the subjective parameter of loudness level. This example displays the absolute threshold for binaural free field listening, and thresholds for discrimination and pain. Don then described the comparatively recent development of "cognitive psychology", which is a more holistic approach, aiming to articulate the structure of human thought and action regardless of biological or social cause. The playing of a musical instrument and recording a symphony orchestra were described as highly developed cognitive skills, where the sensation of sound is the basic common element.

Don reminded us of some interesting aspects of hearing damage, namely:

- That hearing damage can result in distortions, loss of hearing acuity and sometimes an unusual growth in the sensation of loudness.
- That despite a mild to moderate rise in absolute threshold, sounds presented well above absolute threshold, such as those typically heard by an orchestral player, may not be adversely distorted.
- The presence of a hearing impairment can introduce an annoyance component in the sensation of normal sounds and exacerbate sounds that are already annoying.
- The quantification of the psychological concept of annoyance for particular sounds is difficult, not only because of the actual sound structure and its temporal characteristics, but for variability in susceptibility between different individuals in such as adaptation, habituation, and individual states of hearing.

Two psychoacoustic research studies were cited and outlined: One, to investigate the sensation of auditory spaciousness in a concert hall (1), the other a study in categorical perception for various kinds of sounds (2). Because of its profound application to auditory theory and psycho-physiological practice, Don's concluding remarks briefly addressed the Critical Band Concept, used to explain such aspects as loudness, pitch, frequency analysis, masking, phase, and timbre of musical sounds (3).

1 "Auditory spaciousness: some further psychoacoustic analyses" JASA Vol. 80 (2) August 1986

2 "Spectral-temporal factors in the identification of environmental sounds" JASA Vol 115 (3), March 2004

3 Scharf B (1970) "Critical Bands" in "Foundations of Modern Auditory Theory" Volume I. Ed. Tobias J V

Dr Paul Niall then gave a very interesting talk on the System used for hearing loss assessments in compensation claims in NSW. Dr Niall is a Consultant Occupational and Audiological Physician and is Chief Medical Officer and Medical Referee with the Compensation Court of NSW. Paul explained some of the difficulties associated with hearing loss assessments, most notably the fact that there is no obvious date of injury. He then explained how initial attempts to quantify hearing loss had used speech tests because loss of speech perception is perhaps the most significant impact of hearing loss. However, this method had incurred a number of problems and Paul went on to outline the development of pure tone audiometry methods developed in Australia and the USA since the 1940's and discussed the current procedure, referred to as the NAL procedure, including: the low and high cut-off values used in the assessments; corrections applied for presbycusis; and the way in which binaural corrections are applied (weighted towards the "good ear").

Paul then explained how hearing impairment and compensation assessments are made in conjunction with the WorkCover guidelines, noting that certain slightly illogical changes to legislation were introduced in the 1990's following "touting" for business in the compensation claims business. An issue of particular difficulty is that of how liability can be shared between a number of previous employers.

In response to a question on how workplace vs recreational noise exposure is dealt with, Paul noted that research from the 1980's had indicated that noise induced hearing loss was overwhelmingly the result of workplace exposure, but acknowledged that workplaces have become quieter while perhaps recreational places (such as night clubs) have become noisier since that time.

Dave Anderson

WIND TURBINE NOISE FORUM

On Mar 24, the Vic division held a Wind Turbine Noise Forum jointly with the Environmental Engineering Society. Norm Broner (AAS Vic Div chairman) welcomed the speakers and the about 50 acousticians and engineers present.

First, Meagan Wheatley (Sustainable Energy Authority) spoke broadly about "Wind energy in Victoria". Electricity demand is at present increasing by 2 to 3% per year. The renewable energy target has been currently set at 10% by 2010, to include 600 wind turbines. It contains provision for generation also from geothermal, solar (both photo-voltaic and heat) and tidal sources, in addition to the current 5% from hydro-power. A Wind Atlas has been compiled to show wind speeds at 65m above ground throughout Victoria. For maximum turbine efficiency, wind in laminar flow is the most effective, with wind farm output varying as the cube of the wind speed. Guidelines for siting wind farms include that they are not allowed in National Parks, their visual aspect must be considered, and that electricity transmission lines longer than 30 km are costly.

The second speaker, Dr Graham Slack (Garrad Hassan Pacific P/L, wind energy consultants) described "Wind farm design and planning" as a process of continuing development. The present move is to larger turbines because the energy generated varies as the square of the wind blades' diameter, with diameters currently up to 20m. For maximum wind use, blades are of variable pitch with the generators driven via a variable speed gearbox. Though winds are variable, the wind turbines are stated to have a reliability of 97 to 99%. While this cost of generating electricity per kWh is continually declining, it is currently around 5c/kWh. As the blades rotate, a "hishing" sound (a broad band noise) is generated as they pass through the "shadow" of the turbine support column. Blade tip speeds, critical in this regard, are of the order of 60 to 80m/s. Helical gears minimize gear box noise. Each turbine is taken as a point source, with the noise levels quoted relative to the background noise (which may include wind noise).

In selecting wind farm sites, several design trade-offs must be made. The high elevation of turbines raises visual issues. With standards requiring rigorous noise limits, noise constraints can be severe.

The third speaker, Juliett Bird (National Trust), spoke in qualitative terms about reconciling the need to maximize wind energy use with various environmental sensitivities, and of balancing maximum benefits against monetary and environmental costs and minimum inconvenience. Suitable winds frequently occur in visually sensitive sites. Current weaknesses in assessment procedures include inadequate landscape assessment criteria (an EES is not always prepared), and insufficient environmental protection. Currently, protected areas comprise 15% of the state, including 43% of its coastline. Many environmentally sensitive landscapes are not yet officially recognized. With the very quick progress of the wind industry, care in preparing EESs is all the more necessary.

The fourth speaker, Dr Peter Teague (Vipac-Adelaide) gave an overview of Standards & guidelines for wind turbine noise. Currently, there were international standards, a Draft AS, and EPA guidelines. ETSU-R-97, Assessment & rating of wind farm noise, is a UK review document giving guidance on noise measurement and setting noise limits. IEC61400-11, on wind turbine generators, gives methods for data reduction, regression analysis, and obtaining tonal components for the adjustment of measured sound levels. The SA EPA 2003 Guidelines describe noise measurement, assessment and prediction procedures, and provide noise criteria. For wind speeds from 3 to 15m/s the criteria for turbine and background noise are, respectively, L_{eq}35 dB(A), and L_{90} + 5dB(A). An adjustment of + 5 dB(A) to the measured levels needing adjustment allows for the presence of tonal components and low frequency noise.

The fifth speaker, Gustaf Reutersward (Richard Heggie Assoc P/L), spoke about Wind farm noise assessment. The procedures described in the NZ standard, the Draft AS, and the SA EPA Guidelines are similar. Background noise levels are plotted as a function of wind speed. Extraneous noise data (from traffic, insects, frogs, etc) need to be removed. A specialized microphone wind shield, and a sheltered location for measuring wind noise are needed to eliminate microphone wind induced noise. Weather monitoring equipment is also needed in obtaining seasonal influences on wind data, including wind directional effects from trees, etc. Because wind noise varies with wind direction, wind data need to be separated into 90° or even 45° sectors. The hishing noise of blades passing the tower, detectable at up to 600m from it, depend on blade and tower design, with the level accordingly varying periodically by from 2 to 5 dB(A) attracting a 5 dB penalty adjustment). Complaints of this noise are a late night/early morning phenomenon — of a sleep disturbing nature.

During the subsequent discussion of how people react to noise, it was suggested that during the day transformer hum was more annoying than wind turbine noise. The effects of machine aging on wind turbine noise still need to be considered.

OCCUPATIONAL NOISE

On Jun 15 at 4pm, a joint meeting was held with the Australian Institute of Occupational Hygienists. Alex Simovski (Worksafe) described the revised Victorian Occupational Health & Safety [Noise] Regulations introduced in Jan 2004, the significant changes made, and the implications for employers and consultants in applying them.

In describing the 2004 regulations, Alex stated that, basically, they were set out in four main sections -- the Preliminary Part (1), including their objective (to reduce the incidence and severity of hearing loss resulting from excessive exposure to noise) and exposure standard, and Parts 2 to 4 describing the duties of plant designers, manufacturers and suppliers, of employers and of employees. While similar to those of 1992, they had superseded them on Jan 30, and differed from them in certain significant respects. These comprised removing the need for Worksafe to approve audiometrists and be notified of employee audiometric test results, deleting the hearing loss criteria (type A) related to age, and changing the maximum peak exposure level from 140 dB(lin) to 140 dB(C), though the 8-hour exposure standard remained unchanged at Leq = 85 dB(A) rms. Other changes were made to the methods of reviewing noise controls after an audiological examination, of recognizing representative noise assessments of an area, of consulting with employees in the absence of a Health & Safety Rep, and of modifying control hierarchies and written noise control plans.

Louis Fouvy

Future Conferences

ACOUSTICS 2004

The national conference for the AAS, ACOUSTICS 2004, will be held on 3-5 November 2004 at Surfers Paradise on Queensland's Gold Coast. The Conference theme is "Transportation Noise & Vibration, the New Millennium". Other major topics for the Conference will include Underwater Acoustics and Architectural and Building Acoustics but papers from all areas of acoustics will be included in the program. An exciting program has been developed with Plenary, Keynote and Invited speakers in addition to the contributed papers. There will be workshops on Wheel/Rail Noise & Vibration, New Building Code of Australia (BCA) and Building Design for Transportation Noise Control. Prior to the conference will be a Short Course on Environmental & Transportation Noise - Planning and Enforcement Issues

There will be a technical exhibition as well as a lively social program. So make sure you get your registration in soon.

Information on the conference from www.acoustics.asn.au

NOVEM 2005

NOVEM 2005, 18 to 21 April will be a specialised conference focused on emerging techniques in acoustics and vibration. This conference will be held in Saint Raphaël, a small town on the Azure Coast, not far from Nice International Airport. The objective is to promote discussion and exchange of ideas in a relaxed atmosphere. Four topics will be covered:

1) Prediction for noise design
2) Novel modelling approaches
3) Innovative material technologies
4) Advanced identification techniques

Abstracts can be submitted now and the full papers will be due by 14 Jan. Information from http://www.novem2005.com/

New Concepts for Harbour Protection, Littoral Security and Shallow-Water Acoustic Communication

This congress will be held in Istanbul, Turkey, 4-8 July 2005. The aim is to focus on recent advances in the areas of research and development relevant to harbour and coastline protection, security and communication in the undersea environment. Areas of interest include: Both active and passive acoustic techniques for harbour protection and/or barriers; Active and passive acoustic techniques for target detection, classification, and tracking in the littoral zones; Mine detection, classification, and mitigation in both littoral zones and harbours; Acoustic communications both in the very shallow confines of harbours and bays as well as the deeper littoral regions off the coast.

Areas of interest are not limited to acoustic techniques and all abstracts related to the overall goal of the conference will be considered. Deadline for abstracts 31 October 2004 and for papers 31 May 2005. Information http://www.tica05.org/

ICA 2010

The Australian Acoustical Society has been successful in its bid to host the ICA in Sydney in 2010. While it sounds a long time away the intervening years will quickly disappear. There are a multitude of reasons that make the ICA Congress a very special international acoustics event. One is that it is truly a congress and sessions cover the wide range of topics in acoustics. These include musical acoustics, speech, underwater acoustics, signal processing, noise and vibration to name just a few. The ICA has a tradition of having high quality plenary and distinguished presentations which provide a review of the current state of knowledge in the areas. Thus attending the congress is one way to keep up with what is happening in allied fields as well as just enjoying the opportunity to hear well presented papers on topics that may be in far different fields of acoustics from those in which you usually work. As well there are many parallel sessions with contributed papers and a technical exhibition. Overall the five day program allows the participant to become immersed in acoustics. Information about ICA 2010 will begin to appear on www.acoustics.asn.au soon so prepare for this event.

New Members

Member

Harvey Dillon (NSW), Rachel Foster (SA), Gayle Greer (NSW), Keith Hewett (NSW), Danny Kastak (NSW), Les Johnston (NSW), Gregory Stewart (NSW), Dianne Wiliams (Vic), Alex Zinoviev (SA)

Graduate Subscriber

Craig Maxwell (WA)

Student

Laura Brooks (SA), Yong-Keat Lee (SA), Vladimir Pavasovic (Vic)

Obituary

Peter Kotulski

Peter Kotulski was born on 6 November 1944, in Germany, where his Mum, a Polish teenager, was taken and forced to work during the WW II. The family came to Australia in 1951 and Peter was the dux of his final year at Marist Brothers' Catholic High School. He graduated from UNSW with a BSc in the late 60s. He worked for a couple of years at UNSW School of Physics and then moved to a much better paid (at that time) position with NSW Health Commission, where he was mostly involved in occupational noise issues. He joined the then Noise Branch of State Pollution Control Commission (SPCC), later Environment Protection Authority of New South Wales (EPA), soon after its creation in 1975, and worked there till the late 1990s. He then completed a diploma in teaching and taught Physics and Mathematics in Ashfield and later Wiley Park High Schools.

His knowledge was absolutely enormous and ranged from Science to Medicine, History, Geography, Religion and Politics, but the most impressive was his intellect: his ability to understand, discuss and analyse the most difficult scientific and technical concepts. He started feeling ill in late 2003 and, soon after was diagnosed with Melanoma. From December 2003 he was in constant and unbearable pain, and he had a number of devastating chemotherapy treatments. During this time he was incredibly brave thanks to his exceptionally strong and unconditional faith as a most devoted Catholic and the support from his family. He died on 26 May 2004 and left behind his wife, daughter and son.

Alex Jochelson

TENTATIVE FUTURE CONFERENCES

2005 WA	2006 NZ/AUS
2007 QLD	2008 NSW
2009 VIC	2010 ICA

ACOUSTICS IN THE BCA

On Wednesday 11 August 2004, approximately AAS members and guests attended a (standing room only) meeting to discuss the new acoustic provisions in the Building Code of Australia (BCA) at CSRs Pyrmont premises. The talks were given by Jon Farren of Marshall Day Acoustics (Sydney) and Michael Ryan of CSR. Jon gave an overview of the changes and added some valuable comparisons nased on his experience working with similar codes in the UK and New Zealand. Michael covered some of the 'deemed-to-satisfy' constructions and informed us that CSR are soon to produce the famous 'red book' updated and extended to give constructions that comply with the new BCA. And electronic version of the red book will also soon be available on CD.

Some of the major concerns raised during the question time session were:-

- Is the weighted normalised impact sound pressure level given in the BCA (not more than 62 dB) reasonable? Most delegates felt that this was not reasonable and the 55 dB used in New Zealand and by some Councils in NSW is a better option.
- Should the impact spectrum adaptation term (C_I) be positive (i.e. $L_{n,w} + C_I$) or negative (i.e. $L_{n,w} - C_I$) or just not used at all? It seemed that the positive sign is a possible error and perhaps the C_I should be dropped.
- Is the introduction of the spectrum adaptation term C_{tr} (i.e. $R_w + C_{tr}$) for duct, soil, waste or water supply pipe useful? The opinion was that the spectrum adaptation term for ducts and pipes does not reflect an improvement in the sound insulation criteria (as it does for walls) and should be dropped.
- Is flanking correctly covered in the BCA? It was suggested that this could be addressed if manufacturers publish suitable junction details for party-wall/corridor-wall and party-wall/external-wall.

Overall, the new BCA was welcomed due to the improvements in the weighted standardised level difference with the spectrum adaptation term ($D_{nT,w} + C_{tr}$) for walls and floors and adding the section on verification methods.

The evening concluded with further debate whilst enjoying an excellent spread of finger food (courtesy of CSR) and Indigo Ridge Wines.

Ken Scannell

FASTS

Science Meets Parliament has been postponed from August due to uncertainty around the election date. It will be held in March 05. The AAS is hoping that FASTS will launch the policy document on the Future of Acoustics in Australia during this event. Any comments on participation in this event and issues to be raised should be forwarded to GeneralSecretary@acoustics.asn.au

Chief Scientist FASTS has been very active on the issue of the Chief Scientist and are pleased with the outcome of the Government report. While the report stated that there was no evidence that Dr Batterham had behaved inappropriately or improperly as Chief Scientist it said that the public interest is not being served as long as the perception of a conflict of interest remains and is not properly managed and that the appearance of a conflict of interest has undermined public confidence in the Office of the Chief Scientist. Accordingly, it recommends restoring the position of Chief Scientist to a full-time appointment under public service conditions as a matter of importance and sound public administration.

FASTS strongly endorses the recommendation that the position be made full time although our rationale goes to the importance of science advice to Government rather than addressing conflict of interest issues (as that still exists whether the position be full or part time). We would prefer the position be statutory rather than a PSA position.

Eureka Prize for Baldwin Dr Ken Baldwin has won the 2004 Australian Government Eureka Prize for Promoting Inderstanding of Science for initiating and championing the annual FASTS' *'Science Meets Parliament'*. Congratulating Dr Baldwin on his win, the president of FASTS, Professor Snow Barlow said *Science Meets Parliament* "has become a standout event in the calendar of politicians and scientists. For two days a year, 250 scientists and more than 150 politicians discuss a whole raft of ideas in cutting edge science and how these can improve the quality of our lives, environment and economy. *Science Meets Parliament* is a two-way street that has transformed both politicians' understanding of the diversity and potential of science and scientist's understanding of parliamentary processes."

Dr Baldwin is the Chair of the FASTS' policy committee and a Senior Fellow and Deputy Director of the ARC Centre of Excellence for Quantum Optics at the Australian National University. The Australian Museum Eureka Prizes are the nation's most comprehensive science awards. Now in its 15th Year, the Eureka Prizes raise the profile of science in the community by acknowledging and rewarding outstanding achievements in research, innovation, engineering, training, journalism and education.

The AAS participated in *Science Meets Parliament* for the first time in 2003 and plans to continue to have some involvement — the next will be in March 2005.

AUSTRALIAN ACOUSTICAL SOCIETY
CODE OF ETHICS

1. **Responsibility**
   The welfare, health and safety of the community shall at all times take precedence over sectional, professional and private interests.
2. **Advance the Objects of the Society**
   Members shall act in such a way as to promote the objects of the Society.
3. **Work within Areas of Competence**
   Members shall perform work only in their areas of competence.
4. **Application of Knowledge**
   Members shall apply their skill and knowledge in the interest of their employer or client, for whom they shall act in professional matters as faithful agents or trustees.
5. **Reputation**
   Members shall develop their professional reputation on merit and shall act at all times in a fair and honest manner.
6. **Professional Development**
   Members shall continue their professional development throughout their careers and shall assist and encourage others to do so.

EXPLANATORY NOTES

1. Responsibility

In fulfilment of this requirement members of the Society shall:

(a) avoid assignments that may create conflict between the interests of their clients, employers, or employees and the public interest.

(b) conform to acceptable professional standard and procedures, and not act in any manner that may knowingly jeopardise the public welfare, health, or safety.

(c) endeavour to promote the well-being of the community, and, if over-ruled in their judgement on this, inform their clients or employers of the possible consequences.

(d) contribute to public discussion on matters within their competence when by so doing the well-being of the community can be advanced.

2. Advance the Objects of the Society

Appropriate objects of the Society as listed in the Memorandum of Association are:

Object (a)

To promote and advance acoustics in all its branches and to facilitate the exchange of information and ideas in relation thereto.

Object (e)

To encourage the study of acoustics, highlight excellence in acoustics and to improve and elevate the general and technical knowledge in any manner considered appropriate by the Society.

Object (g)

To encourage research and the publication of new developments relating to acoustics.

3. Work within Areas of Competence

In all circumstances members shall:

(a) inform their employers or clients if any assignment requires qualifications and/or experience outside their fields of competence, and where possible make appropriate recommendations in regard to the need for further advice.

(b) report, make statements, give evidence or advice in an objective and truthful manner and only on the basis of adequate knowledge.

(c) reveal the existence of any interest, pecuniary or otherwise, that could be taken to affect their judgement in technical matters.

4. Application of Knowledge

Members shall at all times act equitably and fairly in dealing with others. Specifically they shall:

(a) Strive to avoid all known or potential conflicts of interest, and keep employers or clients fully informed on all matters, financial or technical, that could lead to such conflicts.

(b) refuse compensation, financial or otherwise, from more than one party for services on the same project, unless the circumstances are fully disclosed and agreed to by all interested parties.

(c) neither solicit nor accept financial or other valuable considerations from material or equipment suppliers in return for specification or recommendation of their products, or from contractors or other parties dealing with their employer or client.

5. Reputation

No member shall act improperly to gain a benefit and, accordingly, shall not:

(a) pay nor offer inducements, either directly or indirectly, to secure employment or engagement.

(b) falsify or misrepresent their qualifications, or experience, or prior responsibilities nor maliciously or carelessly do anything to injure the reputation, prospects, or business of others.

(c) use the advantages of privileged positions to compete unfairly.

(d) fail to give proper credit for work of others to whom credit is due nor to acknowledge the contribution of others.

6. Professional Development

Members shall:

(a) strive to extend their knowledge and skills in order to achieve continuous improvement in the science and practice of acoustics.

(b) actively assist and encourage those under their direction or with whom they are associated to advance their knowledge and skills.

Diary

2004

30 Aug-1 Sept, Maastricht
Low Frequency 2004
http://lowfrequency2004.org.uk,
organiser@lowfrequency2004.org.uk,

8-10 September, Athens
From Scientific Computing to Computational Eng
http://ic-scce.upatras.gr/

14-16 September, Turkey
WSEAS Conferences
http://wseas-conferences.50megs.com

14 - 16 September. Loughboro
Int Conf Sonar Signal Processing & Symp Bio-Sonar Systems & Bioacoustics.
http://ioa2004.lboro.ac.uk

20-22 September, Leuven
ISMA2004
http://www.isma-isaac.be

20-22 September, Williamsburg
Active 2004.
www.inceusa.org

14-15 October, Nis
XIX Conference Noise and Vibration.
momir@znrfak.znrfak.ni.ac.yu

04 - 08 October, Jeju
8th Conference on Spoken Language Processing (Interspeech).
www.icslp2004.org

3-5 November, Gold Coast
Acoustics 2004
AAS Annual Conference
PO Box 760, Spring Hill, QLD 4004, AUSTRALIA,
www.acoustics.asn.au, aas2004@acran.com.au

8-9 November, Wellington,
17th Biennial Conference of the New Zealand Acoustical Society.
www.acoustics.org.nz

6 - 9 December, New Delhi
ACSIM 2004
4th Asia Pacific Conference on Systems Integrity and Maintenance
www.acsim.com/

08 -10 December, Sydney
10th Australian International Conference on Speech Science and Technology.
www.assta.org/sst/2004

2005

31 Jan-4 Feb, Canberra
AIP Conf, Physics for the nation
www.aip.org.au

19 - 23 March, Philadelphia
International Conference on Acoustics, Speech, and Signal Processing.
www.icassp2005.org

18 - 21 April, Saint Raphaël
Intl Conf Emerging Technologies of Noise & Vibration Analysis & Control.
goran.pavic@insa-lyon.fr

01 - 03 June, Hiroshima
1st Int Symp on Advanced Technology of Vibration and Sound.
dezima.ike.tottori-u.ac.jp/vstech2005

28 June - 1 July, Heraklion
Int Conf Underwater Acoustic Measurements: Technologies and Results
http://UAmeasurements2005.iacm.forth.gr

11-14July Lisbon
ICSV12
www.iiav.org, icsv12@ist.utl.pt.

19-23 March, Philadelphia
Int Conf Acoustics, Speech, and Signal Processing.
http://www.icassp2005.com

18-21 July, PenState
Int Symp Non Linear Acoustics.
atchley@eng.psu.edu

6-10 August, Rio de Janeiro
Inter-Noise 2005.
www.internoise2005.ufsc.br, samir@emc.ufsc.br

05 - 09 September, Bath
Boundary Influences in High Frequency, Shallow Water Acoustics.
http://acoustics2005.bath.ac.uk

11 - 15 September, Beijing
6th World Cong Ultrasonics (WCU 2005).
www.ioa.ac.cn/wcu2005

2006

15 - 19 May, Toulouse
IEEE International Conference on Acoustics, Speech, and Signal Processing
(IEEE ICASSP 2006).
http://icassp2006.org

26-28 June, Seoul
WESPAC9
www.wespac9.org

03 - 07 July, Vienna
13th International Congress on Sound and Vibration (ICSV13)
http://info.tuwien.ac.at/icsv13

28 November - 02 December, Honolulu
Acoustical Soc of America & Acoustical Soc of Japan Fourth Joint Meeting.
http://asa.aip.org

3-6 December, Honolulu
Inter-Noise 2006.
www.i-ince.org

2007

9-12 July, Cairns
ICSV14
n.kessissoglou@unsw.edu.au

2-7 September, Madrid
ICA2007
www.ia.csic.es/sea/index.html

2010

August, Sydney
ICA2010
www.acoustics.asn.au

Meeting dates can change so please ensure you check the www pages.
Meeting Calendars are available on www.icacommission.org/calendar.html and www.i-ince.org

AUSTRALIAN ACOUSTICAL SOCIETY - SUSTAINING MEMBERS

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INGLEBURN 2565

G P EMBELTON & CO PTY LTD
P O BOX 207
COBURG 3058

HPS TECHNOLOGY PTY LTD
17 BALLANTYNE ROAD
KEWDALE 6105

INC CORPORATION PTY LTD
63-79 SOUTH PARK DRIVE
DANDENONG SOUTH 3175

NOISE CONTROL AUSTRALIA PTY LTD
70 TENNYSON ROAD
MORTLAKE 2137

NSW ENVIRONMENT PROTECTION AUTHORITY
P O BOX A290
SYDNEY SOUTH 1232

PEACE ENGINEERING PTY LTD
P O BOX 250
NARELLAN 2567

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149 MAGOWAR ROAD
GIRRAWEEN 2145

SOUND CONTROL PTY LTD
61 LINKS AVENUE NTH
EAGLE FARM 4009

VIPAC ENGINEERS AND SCIENTISTS LTD
279 NORMANBY ROAD
PORT MELBOURNE 3207

WARSASH SCIENTIFIC PTY LTD
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AUSTRALIAN ACOUSTICAL SOCIETY ENQUIRIES

NATIONAL MATTERS

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

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Sec: Richard Devereux
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Fax: (07) 3217 0066
rdevereux@acran.com.au

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Department of Mech Eng
University of Adelaide
SOUTH AUSTRALIA 5005
Sec: Anthony Zander
Tel: (08) 8303 5461
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azander@mecheng.adelaide.edu.au

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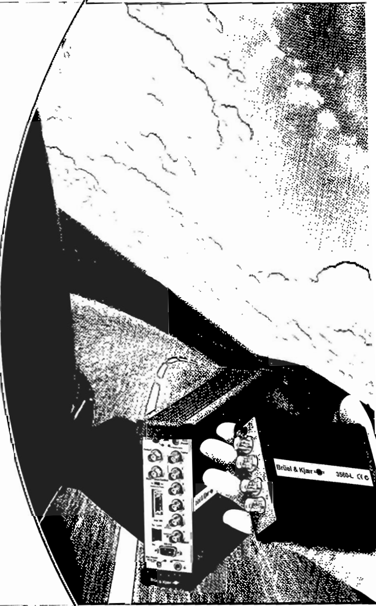

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