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# Acoustics Australia





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## LETTERS



## PAPERS



## TECHNICAL NOTES



## REGULAR ITEMS



*Cover design: Helena Brusic*

# MESSAGE FROM THE PRESIDENT

Hello all,

Time is flying fast and we are already half way through this year. We recently had a Federal Council phone meeting and discussed our Federal Budget, subscriptions and the potential use of some of our funding to support research in areas of interest to acousticians. Watch this space for more details. We also noted that our membership is now 619 and, of these, 304 have logged in to our website and 315 have never logged in at all. Note that we are also currently looking at upgrading our website and hope to be able to report on this very soon. In relation to our website, one very interesting statistic is that page requests for the Journal back issues are very high, not only from Australia but also worldwide. In the period from October 2011 to May 2012, there were 455,315 requests for journals or articles. Note also that the proceedings from the recent conferences Acoustics 2011: Breaking New Ground and Acoustics 2012 Fremantle: Acoustics, Development and the Environment, have been added to our website, as well as individual journal articles are now available on-line since journal volume 23 no. 1 in 1995! So there is a wealth of information available via our website so please make use of this valuable resource.

We are fast approaching this year's Annual Conference which is being hosted by our South Australian Division from November 17-20, 2013. The venue is the McCracken Country Club in picturesque Victor Harbor. A number of workshops have been organised. Associate Professor Con Doolan is running a workshop on Flow Induced Noise while Professor Lex Brown is running an International Research Workshop focussed on Children and Noise. Dr Carl Howard will also be holding the inaugural meeting of "Australian Educators in Acoustics". In addition, we will have many themes covered including low frequency and wind turbine noise, architectural and building acoustics, underwater acoustics, marine environmental and bioacoustics, and environmental acoustics. There is also a large exhibition which will showcase new acoustical products and instrumentation. At the time of writing this message, there were only four booths left!

Also please note that AAS AGM will be at the Conference on Tuesday afternoon 19th November and that although the venue accommodation is fast filling, there are many accommodation venues close by that are listed on our web site.

Please go to the following website to register and find out more details. http://www.acoustics.asn.au/joomla/acoustics-2013.html

I look forward to seeing you all in Victor Harbor.

*Norm Broner*
*President AAS and President Internoise 2014*

# MESSAGE FROM THE EDITOR

A wise person worked out early on that the best way to get me to do any work was to dangle the incentive of a conference. If it hadn't been for the wisdom of my wonderful supervisor, Prof. Jie Pan from the University of Western Australia, I don't think I ever would have finished my PhD and subsequently had an academic career. 15 years later I am still enjoying conferences as much as ever – how could I not surrounded by such tall good-looking guys! This photo was taken at the recent ICSV20 conference in Bangkok. Am looking forward to my next conference already.

*Nicole Kessissoglou*

*(left to right) Marcus Güttler, Max Stanton, Tristan Lippert, Nicole Kessissoglou, Tom Basten, Frank Stalter, Marten Nijhof, Herwig Peters.*

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# LETTER TO THE EDITOR

**Paul Miskelly, Mittagong, NSW**
**paul.miskelly@aapt.net.au**

Response to article by S. Cooper, "Wind farm noise - an ethical dilemma for the Australian Acoustical Society?", *Acoustics Australia* **40**(2), 139-142 (2012)

I would like to add to the discussion raised by Mr Steven Cooper's article "Wind farm noise – An ethical dilemma for the Australian Acoustical Society?", *Acoustics Australia* **40**(2), 139-142 (2012).

Letters from Kym Burgemeister and Marshall Day Acoustics (both in *Acoustics Australia* **40**(3), December 2012), each make the important point that there is a need to *balance* the impacts from any given technology on local residents against its benefits to the wider community. It is the latter – the supposed benefits or, "*the greater good*", provided by wind farms - that I wish to address here.

The wind industry and its academic supporters tell us that, while individual wind farms produce an electricity output that is variable, that by spreading a number of wind farms across a wider region, their combined output becomes sufficiently smoothed so that, with a little balancing from gas turbine generation, the combination can readily replace coal-fired powerstations. See, for example, Diesendorf [1]. Thus, we are told, wind farms provide a direct benefit to the wider community by reducing the $CO_2$ emissions from fossil-fuelled generation. This, I understand, has become a generally accepted view among the government policymakers involved in the wind farm approval process.

I am an electrical engineer. I thought to test this smoothing hypothesis [2]. The operator of the eastern Australian grid, the Australian Energy Market Operator (AEMO) publishes, on a daily basis, the previous day's operational data for all registered generators. The data is published as the output at each 5-minute data point for all such times in the 24-hour period, that is, 288 data points for each of more than 300 generators on that grid. There is now a total of some 2700 MW of installed wind farm capacity spread across the eastern Australian grid, the most far-flung grid on earth, over a region that is 1200 km in its east-west extent, and some 500 km over a north-south extent, making this network of wind farms one of the most widely dispersed in the world. I thought to analyse the performance of this network of wind farms for the full calendar year 2010. At that time, the total installed wind capacity totalled a little over 2500 MW, incidentally making it a larger capacity than that of a single coal-fired powerstation.

In light of the generally accepted view stated above, the results were little short of astonishing. Not only is there no appreciable smoothing of the wind farm output, but there occur very frequently through the course of a year what can only be termed common-mode failures of the entire wind farm fleet. I chose a figure of 2% of installed capacity as the minimum acceptable level of output. (I am advised by other electrical engineers that this figure is "kind" to the wind industry: those others would have chosen a figure of 5% as the minimum acceptable figure, a figure that would have resulted in an even worse result.) Using my 2% minimum figure as the failure criterion, the wind farm fleet failed on some 109 occasions in 2010. To provide some perspective, the unscheduled outage of one major conventional powerstation once a year would be deemed unacceptable. This is a direct comparison, in terms of presently installed wind farm capacity, but, in addition, the failure of the entire conventional generation fleet – for this is also a direct comparison – on any single occasion, would be regarded as catastrophic, and would result in a national inquiry.

In addition to the common-mode failure, there were numerous occasions through the calendar year where wind output dropped rapidly from high values, requiring the rapid response of fast-acting gas turbine generation to fill the gap. The rapid response requirement results in inefficient operation of such plant, resulting in excessive $CO_2$ emissions at such times. Inhaber [3] determined, using a conservative approach, that as a result of this excessive fuel consumption, that where wind installed capacity approaches 20 percent of total installed generation capacity (the Federal government's renewable energy target), any resulting $CO_2$ emissions saving is completely nullified by the inefficiencies resulting from the frequent, rapid ramping.

This result is due entirely to the prevailing meteorology: the frequent passage of large high-pressure systems cause occasions where the wind is not blowing anywhere across the entire grid [2]. As a result, increasing the wind farm fleet is no solution to the common-mode failure problem. Indeed, continued increase in installed capacity would merely result in the increased risk of catastrophic grid collapse, as a consequence of the increased absolute magnitude of the swings in wind farm output. Miskelly and Quirk [4] also address the impact of these wind-caused sudden variations in demand.

These findings concur with the empirical observations being made in both the UK and Germany, for example, where there is a new understanding, based on operational experience, that wind energy is both not decreasing $CO_2$ emissions to any appreciable extent, but is also placing the continued operational security and reliability of those countries' respective grids under increasing strain.

I respectfully suggest to members of the AAS that, in the light of these findings, it is time to give serious consideration to the possibility that the "*greater good*" to be had from grid-connected wind farms is not only minimal, but that it is indeed likely to be non-existent. Therefore, any noise impacts on nearby residents resulting from the operation of wind farms are totally unacceptable.

As a result of these findings the ethical issue raised by Steven Cooper takes on a new importance: given that it is clear from these findings that wind energy technology on the eastern Australian grid is a colossal failure in terms of meeting its stated objective, I suggest that currently-misguided policy strategies by governments require a robust response from AAS members. For example, in NSW, as a first such response, the present exemption of wind farms from the stringent requirements of the NSW INP now require, I suggest, that AAS members practising in that jurisdiction lodge objection regarding that exemption as a matter of urgency with the relevant departmental Directors-General.

*Paul Miskelly*

BE, MEngSc (both degrees in Electrical Engineering)

## REFERENCES


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[2] P. Miskelly, "Wind Farms in Eastern Australia – Recent Lessons", *Energy & Environment* **23**(8), 1233-1260 (2012)

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# Inter-Noise 2014

## MELBOURNE AUSTRALIA 16-19 NOVEMBER 2014

The Australian Acoustical Society will be hosting Inter-Noise 2014 in Melbourne, from 16-19 November 2014. The congress venue is the Melbourne Convention and Exhibition Centre which is superbly located on the banks of the Yarra River, just a short stroll from the central business district. Papers will cover all aspects of noise control, with additional workshops and an extensive equipment exhibition to support the technical program. The congress theme is *Improving the world through noise control*.

**Key Dates**

The dates for Inter-Noise 2014 are:
Abstract submission deadline: 10 May 2014
Paper submission deadline: 25 July 2014
Early Bird Registration by: 25 July 2014

**Registration Fees**

The registration fees have been set as:

| | | |
|---|---|---|
| Delegate | $840 | $720 (early bird) |
| Student | $320 | $255 (early bird) |
| Accompanying person | $140 | |
| Congress Banquet | $130pp | |

The registration fee will cover entrance to the opening and closing ceremonies, distinguished lectures, all technical sessions and the exhibition, as well as a book of abstracts and a USB containing the full papers.

The Congress organisers have included a light lunch as well as morning and afternoon tea or coffee as part of the registration fee. These refreshments will be provided in the vicinity of the technical exhibition which will be held in the main foyer of the Congress Centre. Expressions of interest in participating in the exhibition have already been received from overseas and local exhibitors and there is also the possibility of gold, silver or bronze sponsorship. For more details refer to the Congress website or contact Dr Norm Broner, NBroner@globalskm.com

The Congress Banquet is not included in the registration fee, however, as it will have a strong Australian theme and feature the opportunity for delegates to take photographs of themselves with native Australian animals, it should prove to be a major attraction.

**Technical Program**

After the welcome and opening ceremony on Sunday 16 November, the following three days will involve up to 12 parallel sessions covering all fields of noise control. The first Plenary lecture will be by Prof. Jung-Woo Choi from Korea on an emerging topic: *Sound Sketch: its theory and application using speaker arrays*. Prof. Lex Brown from Australia will close the technical program by reviewing: *Soundscape planning as a complement to environmental noise management*.

The Keynote lectures will cover four important areas of the Congress, notably Aircraft Noise, Active Noise Control, Wind Turbines and LFN as well as Building Acoustics /Noise Effects on Humans.

At this stage, over 100 international and Australian based experts have agreed to help with the organisation of the 80 proposed special sessions, covering all the major areas of noise control. The following broad topics are in the early planning stage:

- Active Noise Control (4 sessions)
- Aeroacoustics (6 sessions)
- Building Acoustics (10 sessions)
- Human Reaction - Occupational Noise (7 sessions)
- Industrial Noise and Vibration (6 sessions)
- Maritime – Underwater (5 sessions)
- Road/Vehicle Noise and Vibration (10 sessions)
- Wind Turbine and LFN (5 sessions).

A more complete listing of the session topics will be progressively added to the Congress website.
Abstract and paper submission, as well as registration, will also be through the Congress website, which is
**www.internoise2014.org**

# AN ON-DEMAND SIMULTANEOUS ANNOYANCE AND INDOOR NOISE RECORDING TECHNIQUE

**Con J. Doolan and Danielle J. Moreau**
**School of Mechanical Engineering, The University of Adelaide, Adelaide, Australia**

A novel methodology is presented for the simultaneous measurement of noise and personal annoyance at the exact times that the affected person is annoyed. The system is described and applied to a test case, a farmhouse close to a wind farm where the resident claims to be annoyed by noise. The system was successfully able to characterise the level and spectral content of the noise in the house when the resident was annoyed, and there was some correlation with personally recorded annoyance level. As the system cannot identify noise sources, no conclusions can be made about noise source; however, the methodology is shown to be a useful aid for diagnosing the type and severity of an indoor noise problem. To help interpret the results from this type of testing, a discussion concerning the subjective nature of noise annoyance is presented before some suggestions are made for further improvements in the measurement system.

## INTRODUCTION

Traditional means of measuring noise in residents' homes affected by environmental or other noise may not have the required fidelity to capture important features of noise character. Some aspects, such as low-frequency noise or short-duration events, are not able to be resolved from standard techniques that rely upon 10-minute averages and A-weighting. However, it is difficult to record noise in sufficient detail in the field to resolve these effects due to large data storage and post-processing requirements. Annoyance events may occur infrequently, at random, or when particular weather conditions are present. Sometimes the source of the annoying sound is unknown; hence characterisation in terms of spectral content, annoyance and time of day may lead to its identification or at least a quantification of its severity. Continuous recordings in these situations are sometimes impractical (due to data storage issues, unreliability of the recording method and long data processing times) so a different methodology is needed. To overcome these issues, a new resident-controlled noise and annoyance recording system has been devised and is presented in this paper.

The aim of this paper is to describe a new methodology to record noise and annoyance in situations where the observers perceive unwanted, external environmental noise that may not be easily characterised or analysed by traditional means. The technique records time-series recordings that allow analysis of the signal using a variety of post processing techniques. In order to demonstrate the implementation of the methodology, preliminary results from a trial of the system in a home near a wind farm are presented and show the type of data that is obtained and the different ways it can be analysed. Note that the noise and other data presented in this paper are for a variety of situations where the resident *perceives* that the wind farm is annoying them. For a review of wind turbine noise perception, annoyance and low frequency emission see [1]. As the authors did not have direct control of the noise source attributed by the resident to annoyance, the recorded noise data cannot be directly attributed to the wind farm. However, the data is important as it characterises (in terms of level and spectral content) noise at the exact time observers find it annoying, whether it is caused by turbines, wind noise or another source.

## METHODOLOGY

The system was designed to be placed in a resident's home and operated by them when they noticed and were annoyed by environmental noise. Importantly, the resident rates the annoyance level of the noise using a ten-point scale, where 1 represents not-annoyed and 10 represents the highest level of annoyance. The resident is also able to provide comments describing the character of the noise source or any other information of interest (e.g. weather conditions).

The system uses a Brüel & Kjær 4958 20 kHz precision array microphone connected to a 4mA constant current microphone signal conditioner. This microphone has a flat frequency response over the 10 Hz–20 kHz frequency range and was held approximately 1.5 m from the floor (in a separate room to the other components of the system) in a microphone stand with a 105 mm diameter wind sock placed on it. The output of the microphone and signal conditioner was amplified using a Krohn-Hite Model 3362 Dual Channel Filter before recording the signal using a LabJack U3-HV data acquisition device. The system records 10 seconds of time-series signal at a rate of 12 kHz onto the hard drive of a laptop computer connected to the data acquisition device. A value of 10 seconds is recommended for infrasound measurements in ISO:7196 Annex A [2].

A software interface was programmed in the Visual Basic 6 language. An easy interface between the resident and the data logging system was required so that the system is as user friendly as possible for people who were unfamiliar with computers.

Prior to commencing the acoustic tests, the system was calibrated using a pistonphone in the anechoic chamber at the University of Adelaide. Additionally, the noise floor of the system was measured and is shown in Figure 2.

## TEST CASE

The system was placed in a home that was situated approximately 2.5 km west of an operational wind farm in Australia. The home was of weatherboard construction, located on flat farm land and surrounded by a few large trees. The microphone and wind sock were placed in the centre of a room, approximately 1.5 m from the walls, pointing at a partially open window that faced the wind farm. This room also contained a single bed. All other system components were placed in a neighbouring room from which the resident would operate the system. As the noise levels were low, it was necessary to isolate all of the other system components from the microphone to prevent extraneous noise sources (e.g. the computer fan) from being recorded. The results shown in this paper were recorded by a single resident and taken over a 24 hour period from 22/4/2012 to 8/5/2012. A total of 53 recordings were derived from the test and will be used to demonstrate the system.

In this paper, spectral data are presented in one-third-octave band and narrowband format with a frequency resolution of 2 Hz. Narrowband spectra have been calculated using Welch's averaged modified periodogram method of spectral estimation with a Hamming window function and 75% overlap. According to Bendat and Piersol [3], the 95% confidence interval on the narrowband autospectral density is -1.2/+1.4 dB/Hz. One-third-octave band spectra have been calculated using a filter bank from time-series data.

## RESULTS

Table 1 provides a summary of the results obtained during the test period. It has a column describing the average noise level (for various weightings) for each self reported annoyance level and the number of samples collected at each annoyance rating. The final column states all descriptive comments relating to the wind farm and the resident's perception of the noise, just before the noise was recorded.

Table 1. Summary of results

| Annoyance/ Location | dB(Z) | dB(A) | dB(Z) 10-30 Hz | Number of samples | Selected Comments |
|---|---|---|---|---|---|
| 1 | 51 | 31 | 50 | 2 | Hardly turning |
| 2 | 54 | 33 | 53 | 11 | Turning slowly / quiet hum / murmur from turbine |
| 3 | 53 | 31 | 52 | 7 | Faint rumbling can be heard |
| 4 | 55 | 32 | 54 | 11 | Slowly moving / thumping / rumbling noise / humming noise / can hear a rumbling? |
| 5 | 57 | 33 | 57 | 11 | Turbines moving slowly / rumbling |
| 6 | 54 | 31 | 53 | 7 | Turbines turning quite fast, not as much wind by house / slowly spinning |
| 7 | 54 | 31 | 54 | 2 | Can feel pounding / turning strongly |
| 8 | 67 | 34 | 66 | 1 | Loud thumping / rumbling |
| 9 | 56 | 31 | 56 | 1 | Roaring, rumbling noise |
| Noise floor | 39 | 30 | 36 | 1 | |

Table 1 also lists the overall sound level of the equipment noise floor measured in the anechoic chamber at the University of Adelaide (referred to as 'Noise floor'). The table shows that for all annoyance ratings, the overall sound levels measured in the resident's home are significantly above that of the noise floor. The A-weighted noise measurements sit only just above the noise floor indicating that the majority of the noise measured is at low frequencies.

The total number of samples measured in the home is small, therefore any conclusions are limited to this data set and cannot be made general to a resident's perception of wind farm noise. The data do give an insight into the character of noise that a rural resident perceives as annoying and the operation of the noise recording system itself.

While the levels of noise measured in the home are low, the unweighted data show an increase with annoyance rating, although care must be taken for data at the highest annoyance ratings due to low number of repeat data measurements. Figure 1 plots the mean overall sound levels using three different weightings (Z, C and A) against annoyance rating over the frequency range of 10-1000 Hz. Regression and correlation coefficients for this data are given in the figure caption. The Z (unweighted) and C weighted data show an overall increase with annoyance rating while the A weighted data do not. This is because the majority of the acoustic energy is contained in the lower frequencies. This can be illustrated by examining Figure 2, which shows the single sided power spectral density versus frequency of recordings at various annoyance ratings. The figure shows that as annoyance increases, energy levels increase in the 10-30 Hz band as well as increasing levels of broadband energy to 1000 Hz, the most of which occurs at an annoyance rating of 8. The levels of this spectrum are higher than others suggesting that additional noise sources may be contributing to this measurement. Figure 2 shows that the noise environment is low at high frequencies and that the levels are close to the noise floor of the measurement system at frequencies above 200 Hz.

Note that the peaks at 50 Hz and its harmonics are due to electrical interference. These components have not been removed from the noise measurements as this study is

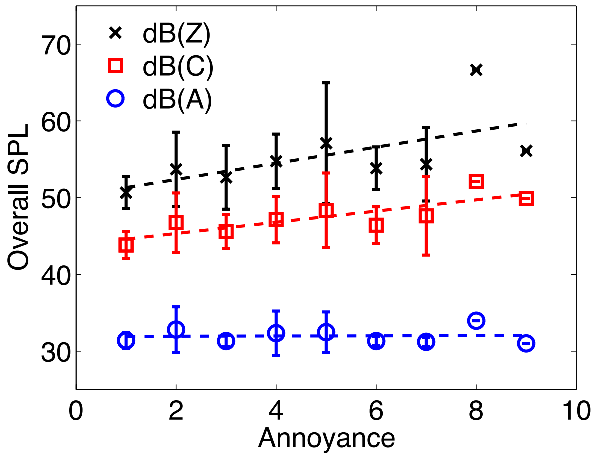


Figure 1. Overall sound pressure level versus annoyance rating by the resident. The levels were calculated over the 10-1000 Hz frequency range. Error bars indicate standard deviation. Regression and correlation coefficients for unweighted data: $b = 1.1$, $p = 0.63$; C-weighted data: $b = 0.73$, $p = 0.83$ and A-weighted data: $b = 0.01$, $p = 0.03$.

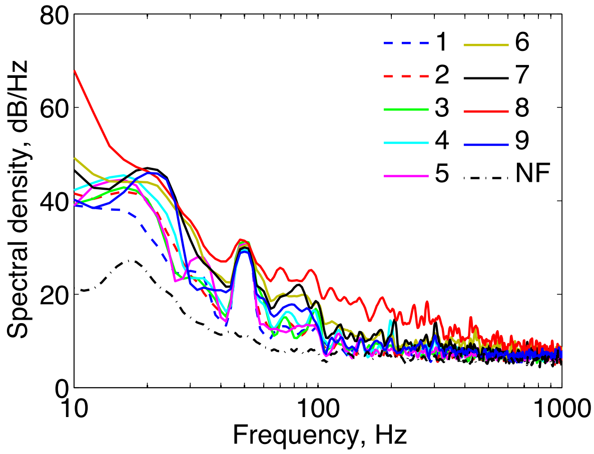


Figure 2. Power spectral density (unweighted) of the acoustic data for various resident-rated annoyance levels. NF indicates noise floor.

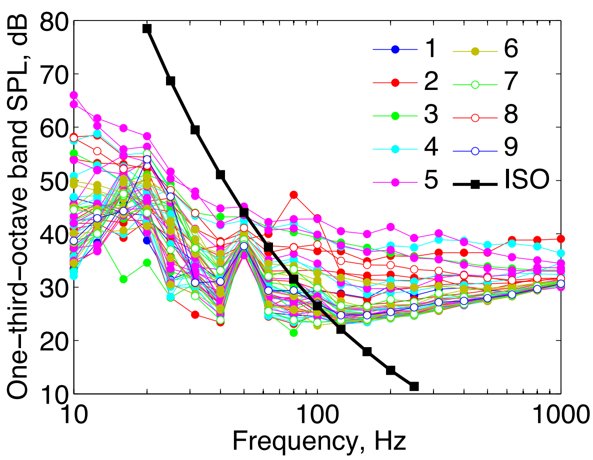


Figure 3. One-third-octave band spectra (unweighted) for all annoyance ratings. ISO refers to the median threshold curve in ISO:226 [5]

comparative and the acoustic energy at 50 Hz and harmonics does not change with each reading, so these noise components do not affect trends in the data. This is confirmed by an examination of the data in Table 1; the difference between the 10-1000 Hz and 10-30 Hz unweighted levels show 1 dB or less variation, showing that noise above 30 Hz is not affecting the trends in Figure 1.

The descriptive comments provided in Table 1 show that the resident is able to perceive unwanted and annoying noise, whatever the source is, and describe it. The comments suggest that the noise is perceived as thumping, rumbling, pounding and roaring. It is possible that acoustic energy below 10 Hz may be responsible for thumping noise; however, future measurements with new microphones capable of measuring below 1 Hz will be performed to help resolve this issue as well as determining what frequency content is responsible for the rumbling and roaring. For a discussion of wind turbine noise sources see [4].

The overall levels are low and are at the limits of detectability. Figure 3 shows all measured noise spectra presented in one-third-octave bands compared to the curve representing the median hearing threshold as listed in ISO:226 [5]. The recorded noise only just exceeds the mean hearing threshold at low frequencies between 50 and 100 Hz. At such low levels, individual differences in hearing sensitivity will make large differences in the rating of annoyance. Further, the levels in the 10-30 Hz band are about 20 dB or more below the ISO curve. A recent review by Leventhall [6] examines the link between low frequency noise and annoyance. The major conclusions from the review are that annoyance by low frequency noise is individual due to a combination of personal and social (non-acoustical) moderating influences. Personal sensitivity to low frequency noise can be influenced by age, gender and social context as well as the ability to cope with an external background stressor, such as noise. Further, Leventhall [6] suggests that there is a possibility of a “learned aversion” to low frequency noise so that a person may be able to develop an enhanced perceptibility to low frequency noise by focussing on it over long periods of time. Thus the sensitivity of a person to low frequency noise is highly individualistic and relates not only to the noise levels but the context of the person’s life that affects the personal and social moderators that influence their sensitivity and reaction.

The subjective nature of an individual’s annoyance rating is illustrated in Figure 4. Here, all single-sided power spectral density results collected for an annoyance rating of 5 are presented. Most spectra have the same shape, showing a broad peak over the 10-30 Hz range and some broadband energy below 1000 Hz. However, some results show higher levels again and are entirely broadband in nature. Thus, the rating of annoyance may be influenced by the particular time of day or personal situation the resident finds himself or herself in. For example, the annoyance to a low level noise may be higher at night than in the day, due to the masking effects of background noise or a personal judgement that it can be noisier in the daytime. Alternatively, if the resident is stressed by other personal or social factors, a lower level noise may be rated as more annoying than at a time when these factors are not

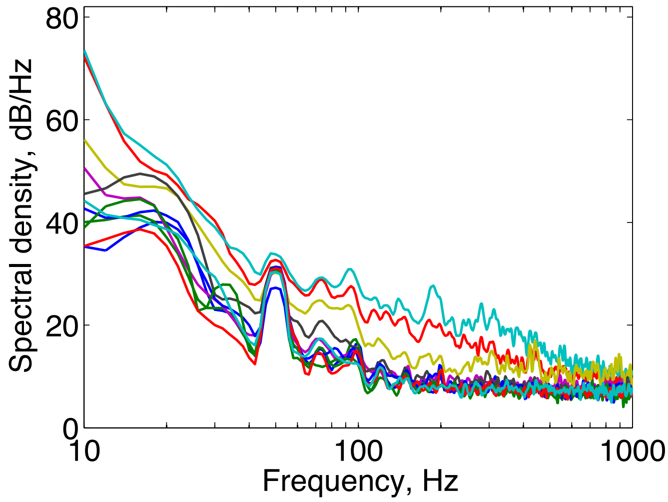


Figure 4. All power spectral density results (unweighted) for acoustic signals rated with annoyance = 5 by the resident

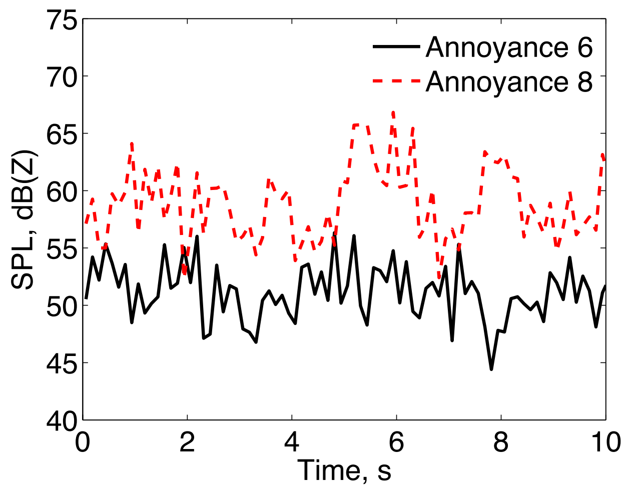


Figure 5. 125 ms time averaged (FAST) unweighted time series sound pressure data for two resident-rated Annoyance levels. The data were band-passed over 10-1000 Hz

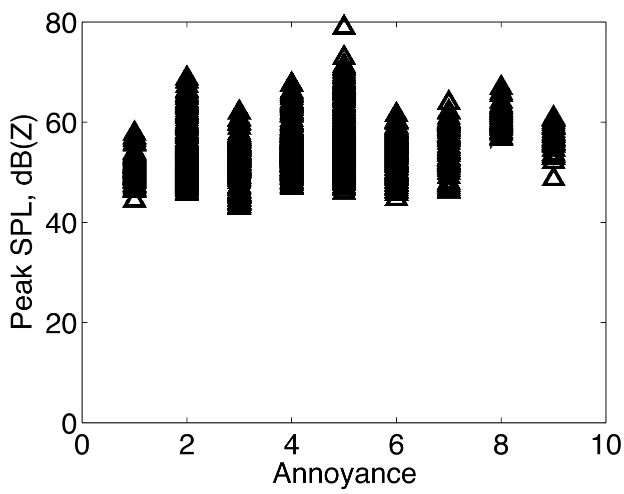


Figure 6. Peak SPL (dB(Z)) from each 125 ms time averaged time series

present. Confusing the issue further is the possible incorrect identification of the noise source by the resident. Annoyance and how noise is related to its causation is a complex psychological process, confounded by many subjective non-acoustical variables (e.g. [7]) which helps explain the variation in the results. It also makes the process of identifying the causes of noise annoyance difficult and the reduction of personal annoyance even more so, especially at low noise levels.

Another factor that may influence a person's sensitivity to low frequency noise is level variation [8]. Figure 5 shows the 125 ms time averaged unweighted sound pressure data for two resident-rated Annoyance levels. The mean level is different for each Annoyance, however, there is significant (up to 10 dB) level variation in each signal. The period of this level variation changes throughout the noise recording and is not associated with blade pass frequency.

To further investigate the link between level variation and annoyance, a peak detection algorithm was used to extract each peak from each 125 ms time averaged data record. These peaks are plotted against Annoyance rating in Figure 6. There is considerable scatter in the data and no trend can be discerned.

The depth of level variation, defined here as the difference in dB between the maximum and minimum levels in each 125 ms time-averaged data record ($\Delta L$), is plotted against Annoyance rating in Figure 7. While there is much scatter, there is no trend with Annoyance. Further, the degree of modulation (m) can be used to characterise the depth of level variation [8]. The degree of modulation is defined by

$$\Delta L = 20\log_{10}\left[(1 + m) / (1 - m)\right] \quad (1)$$

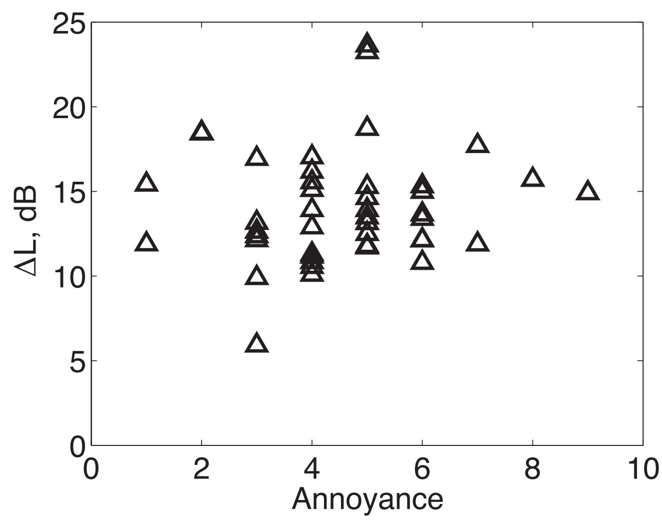


Figure 7. Depth of level variation versus resident rated Annoyance level

Figure 8 plots the mean value of m for each Annoyance rating. This result, and those in Figures 6 and 7, show that there is significant level variation in the recorded signals, but the degree of modulation is relatively uniform for each Annoyance rating and no trend with annoyance can be found. While an interesting result, further studies are required to determine whether the presence of level variation is needed to make this type of low frequency noise more perceptible or annoying, or if it is the solely a function of overall level.

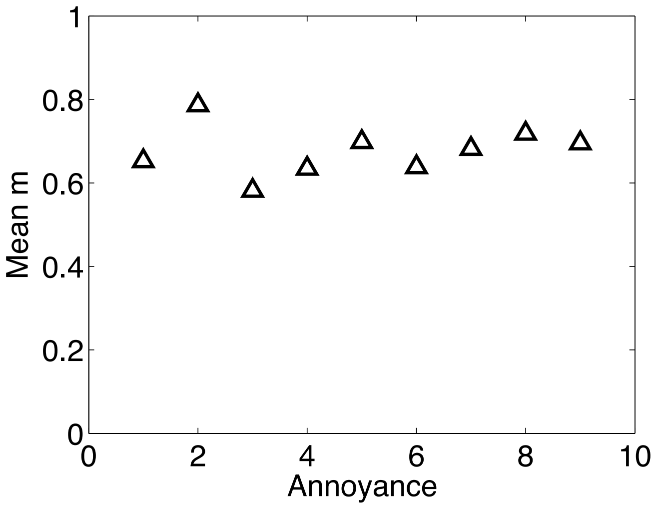


Figure 8. Mean values of the degree of modulation (m) versus resident rated Annoyance level

## SUMMARY

This paper has described a new methodology for recording noise and annoyance within homes or workplaces affected by unwanted noise. The technique records time-series microphone data that allows analysis using a variety of post processing techniques.

A test case, a home near a wind farm, was presented to demonstrate the use of the technique. No link can be made between the noise data and the operation of the turbines; however, the data presented gives an insight into the type and level of noise experienced by residents and that they personally attribute to wind turbines. Measurements show an increase in the overall mean Z (unweighted) and C weighted sound level with annoyance rating. No increase was, however, observed in the mean A weighted sound level and this is due to the majority of the acoustic energy being contained in the lower frequencies and the noise levels being close to the noise floor of the measurement system at higher frequencies. In particular, the energy levels within the 10-30 Hz band were observed to increase with annoyance rating. Additionally, significant level variation was detected in the noise signals; however, no trend with annoyance was observed.

## FUTURE WORK

This study has measured the noise that a resident *attributes* to a wind farm. The question remains whether the source of this noise is actually the wind farm. It is possible that this noise is just wind noise from foliage and building facades, or another source. Taking simultaneous measurements of wind speed and direction with noise and resident rated annoyance would determine if noise level was more strongly correlated with wind speed. Cooperation with the wind farm operators is also desirable to obtain on/off noise measurements; however this is not always possible, so additional work is needed to determine the source location and strength using instrumentation at the measurement location.

Future measurements with the system will incorporate use of a microphone capable of measuring below 1 Hz to capture noise over a larger frequency range than is reported in this study. Another improvement is the incorporation of a high-resolution data acquisition system that will eliminate the need for an amplifier. A weather station located near the home would also be beneficial to record local meteorological conditions that will help identify wind noise. Multiple microphones would also be desirable to measure the variation throughout the home. Longer measurement and averaging periods should also be investigated.

## ACKNOWLEDGEMENTS


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# ASSESSMENT OF AN ACOUSTIC SCREEN USED FOR SOUND EXPOSURE MANAGEMENT IN A PROFESSIONAL ORCHESTRA

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It has been shown that orchestral musicians risk damage to their hearing from workplace noise exposure. Personal, administrative and engineered control measures strive to reduce exposure to the musicians while having minimal impact on the musicians' ability to produce music to the highest standard. Acoustic screens form part of a range of controls used to manage sound exposure, but their construction and placement needs to be carefully considered in the orchestral setting. Existing acoustic screens for use in orchestras have been shown to be ineffectual, to exacerbate risk of injury to some players through qualitative and quantitative changes to the acoustic environment or suffer other practical limitations such as obscuring sight lines. This study reports on an acoustic screen currently in use designed to protect musicians in front of high volume instruments while having minimal impact on players of these high volume instruments through the use of appropriately arranged diffusive and reflective materials. Sound level testing was carried out with the screens in various positions in the orchestra pit with both pink noise and high level instruments used as a sound source. Additionally sound levels were monitored during orchestral performances to assess the impact of the screens on actual exposure levels. Results indicate the screens effectively reduce exposure to those in front of high-level instruments while having a negligible impact on those producing these high sound levels.

## INTRODUCTION

Orchestral musicians are at risk of noise induced hearing loss, with the level of risk they face dependent upon instrument played, repertoire and venue [1,2]. Those most exposed are musicians who work in the orchestra pit, most particularly the members of the brass section [2,3].

While it has been shown in several studies that brass players are significantly more exposed than the musicians around or in front of them (unless in extremely close proximity) those directly in front are clearly at some degree of risk from the output of the brass section. In addition these players also experience high levels of discomfort and anxiety when the brass section is playing at high volume [1-3].

There are a wide range of sound exposure control measures currently in use in Australia's orchestras, which include education programs, administrative controls and personal protective devices [4]. Engineered controls such as risers, orchestral layout and acoustic screens are also an important tool in any orchestral hearing conservation strategy but need to be approached with care. Orchestras often use large perpendicular sheets of Perspex arranged in front of the brass (often also used to shield musicians from the drum kit during 'fusion' performances involving pop/rock bands and orchestras) and/or provide personal screens made from various materials for individual musicians. These include single sheets of Perspex mounted on a stand, wrap-around absorptive screens also mounted on stands and also screens that attach to the musicians' chair and adjust with the player's position.

As practicing orchestral musicians devising and operating hearing conservation strategies, two authors of the current paper have identified several issues that frequently arise with the use and effectiveness of orchestral acoustic screens, some of which have been verified in the limited literature on the topic. Arrangement of large sheets of reflective material in front of highly directional instruments such as the brass reflects sound back to the musicians already at high-risk, increasing their sound exposure and producing significantly deleterious qualitative changes to the acoustic for these musicians. Libera and Mace [5] demonstrated small Perspex personal screens were relatively ineffective as protective devices due to a number of factors, including difficulty positioning the screens due to movement of the musicians and the necessity for very close proximity to the protected ear in order to effectively attenuate sound levels. Williams and Stewart [6] subsequently showed that these screens could increase sound levels by reflection for those musicians playing into them by as much as 3dB, effectively doubling these musicians' sound exposure. More absorptive screens have been shown to be effective as a protective device [6] and are an important tool for orchestras, but have been observed by the authors to be difficult for players of certain instruments to position effectively, often due to restriction of movement while playing – particularly for instruments such as upper strings and trombones. Further to this, when used by several players in a row these screens can create a large absorptive surface within the orchestra. Such a surface in close proximity to the musicians may cause the brass sound to lack power and potentially cause strain injuries as these players strive to project their sound to the conductor's podium and into the auditorium.

In addition to these concerns, any screen or other visual interference placed in front of musicians can cause problems with sight lines to the conductor and effectively isolate these musicians from the rest of the orchestra. To the knowledge of the authors there are no purpose-built orchestral acoustic screens effectively addressing these issues.

A potential solution is to create an acoustic screen using a combination of diffusion and reflection. Diffusion can be used to disperse sound and avoid reflection directly back to the players, while any directly reflective surfaces can use incident angles to also avoid significant reflection of the sound back to the musicians creating it and project sound out of the orchestra pit and into the auditorium.

The Queensland Symphony Orchestra (QSO) has one of Australia's most active and effective hearing conservation programs [4] and as such has been developing solutions to problems with orchestral screens for several years. Recently improvements were made to some of their bespoke acoustic screens in order to address issues players had been having seeing the conductor and the opportunity arose to test their effectiveness in situ.

The aim of this investigation was to determine whether a purpose-built acoustic screen in use in an orchestra pit was effective at mitigating sound exposure while having limited impact on those musicians creating the high sound levels.

## METHOD

The orchestral screens (Figures 1 and 2) are built from light timber with metal legs and Perspex panels.

The base consists of simple variable depth vertical panels (75 mm wide, 35 to 150 mm in depth) while the top section consists of a sheet of Perspex in a supported frame, capable of being angled to suit various situations. This top section has a long (800 mm) or short (400 mm) option, depending upon the nature of protection needed and visual practicability (sight lines to the conductor).

In order to gather data on attenuation, two orchestral trumpet players were recruited on a voluntary basis from the orchestra and undertook testing during a break in the opera being rehearsed at the time. The musicians were instructed to play a short duet (around a minute in length) several times at a 'loud' volume direction (*forte*) in their normal seated position within the orchestra pit.

Three Type 1 data-logging sound level meters (Casella CEL632) were set on stands, one at the chair of the musician directly in front of the trumpets at approximately ear level when seated (1260 mm), the other two within 100 mm of each trumpeter's right ear (Figure 3). The units were calibrated using a matching calibrator directly prior to the assessment. The entire testing session was recorded without adjustment and results were later analysed using proprietary software associated with the sound level meters (*CEL Insight*).

In total four setups were trialled:

1. Screen in place with long top section
2. Screen in place with short top section
3. Screen in place with no top section
4. Screen removed altogether

The musicians played the duet three times for each configuration and were instructed to ensure each performance of the excerpt was as close as possible in volume and tone. Each set of three performances were averaged (using the arithmetic mean) to reduce the impact of any individual variation by the players.

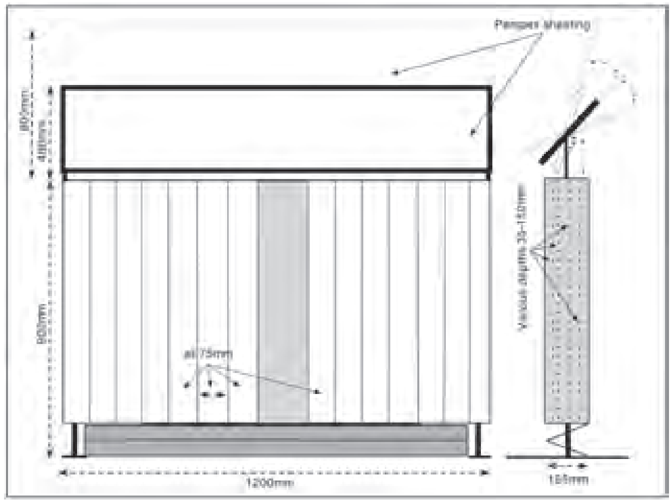


Figure 1. Illustration of diffusive/refractive orchestral acoustic screens

Figure 2. Orchestral acoustic screens setup for ballet rehearsal (short screen in place)

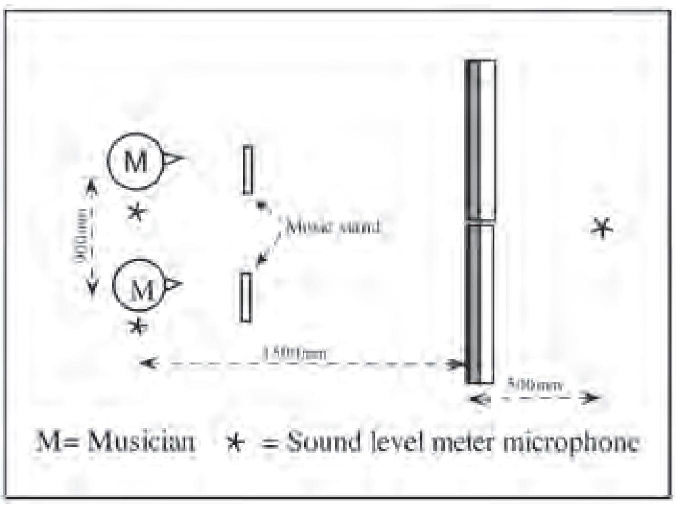


Figure 3. Illustration of recording method

In addition, a series of recordings were made in the four setups above using a loudspeaker positioned in the first trumpet position at seated head height delivering pink noise at approximately 80dBA. The pink noise was generated using the software ‘Noisy’ (version 1.2, produced by ‘Noisy Developers’, 2010) running on a laptop computer. The output level was calibrated using a CEL460 dosimeter in sound level meter mode, with its microphone positioned 100 mm in front of the speaker.

Sound exposure to orchestral musicians is a complex phenomenon involving acoustics, orchestral set-up, repertoire, direct sound from the instrument played as well as sound from adjacent musicians. As such a series of measures were also taken with the screens in place during complete orchestral performances to determine whether attenuation levels observed during the controlled investigation were maintained when many sound sources were present, including that of the ‘protected’ musician. These results were then compared to data from a previous study of sound exposure in this orchestra [2] prior to the introduction of the acoustic screens being investigated, ensuring orchestra size, instrument, orchestral set-up and venue were replicated.

All of these ‘real-orchestra’ measures were taken as part of the QSO’s ongoing noise monitoring program with procedures that have remained unchanged since the program commenced in 2004. Three CEL460 dosimeters were used, calibrated prior to and at the conclusion of each measurement. These were mounted on boom microphone stands in each instance, with the microphone positioned at ear level <30cm from the ear.

Throughout this report data is presented in dBC peak and dBALEQ, with the latter representing the equivalent steady state A-weighted sound level required to replicate the expended energy of the actual (fluctuating) measured exposure over the period of the assessment.

## RESULTS

The configuration with the greatest effective attenuation for either dBA or dBC peak levels for both sound sources was with the longer Perspex screen deployed (Tables 1, 2 and 3). With the trumpets as a sound source the greatest difference between the first trumpet and the screened position was 10.4 dBALEQ and 10.2 dBC peak. With pink noise as a sound source the greatest difference recorded was 20.4 dBALEQ between the pink noise generating speaker and the screened position. This reduction in level includes the effect of distance from the sound source, so in order to highlight the effect the screen had on sound levels in the screened positions, the difference between the unscreened levels at the ‘exposed’ position and the various screened levels has also been detailed in Tables 1 to 3, with the greatest effective attenuation of 4.1 dBLEQ and 5.3 dBC peak.

Table 1. Sound source – trumpet duet, various screen positions. The duet was played three times in each position with results for each set of three arithmetically averaged (the range of this data is indicated in brackets). All data is presented in dBALEQ

| Position | No Screen | Base Only | Short Screen | Long Screen |
|---|---|---|---|---|
| Trumpet 1 right ear (range) | 103.6 (0.6) | 103.3 (0.5) | 103.3 (1.4) | 102.3 (1.4) |
| Trumpet 2 right ear (range) | 101.1 (0.2) | 100.6 (0.4) | 100.9 (0.6) | 101.2 (0.8) |
| Screened position (range) | 97.3 (0.3) | 96.7 (0.3) | 93.6 (0.5) | 91.9 (0.5) |
| Difference between trumpet 1 and screened position | 6.3 | 6.6 | 9.7 | 10.4 |
| Effective attenuation by screen | -- | 0.3 | 3.4 | 4.1 |

Table 2. Sound source – trumpet duet, various screen positions. Peak readings for each setting. Highest recorded peak for each setting is reported. All data is presented in dBC

| Position | No Screen | Base Only | Short Screen | Long Screen |
|---|---|---|---|---|
| Trumpet 1 right ear (range) | 118.8 | 118.4 | 118.6 | 118.3 |
| Trumpet 2 right ear (range) | 118.7 | 118.6 | 118.7 | 117.8 |
| In front of screen | 113.9 | 112.8 | 110.0 | 108.1 |
| Difference between trumpet 1 and screened position | 4.9 | 5.6 | 8.6 | 10.2 |
| Effective attenuation by screen | -- | 0.7 | 3.7 | 5.3 |

Table 3. Sound source - pink noise at 80 dBA, various screen positions. All data is presented in dBALEQ. (* positioned slightly behind the sound source)

| Position | No Screen | Base Only | Short Screen | Long Screen |
|---|---|---|---|---|
| Reference microphone | 80 | 80 | 80 | 80 |
| Trumpet 1 position* | 72 | 72.1 | 72.1 | 72.2 |
| Trumpet 2 position* | 67.8 | 67.9 | 68.1 | 67.9 |
| Screened position | 65.4 | 65.2 | 61.1 | 59.6 |
| Difference between reference and screened position | 14.6 | 14.8 | 18.9 | 20.4 |
| Effective attenuation by screen | -- | 0.2 | 4.3 | 5.8 |

In any of the three screen configurations there was no appreciable increase in sound level (dBC peak or dBALEQ) observed to the trumpet 1 or 2 positions compared to readings taken without the screens in place.

As a significant contributor to sound exposure amongst many orchestral musicians is direct sound from their own instrument in addition to sound from adjacent instruments [1,2], further readings were taken during two orchestral opera and ballet performances to determine whether attenuation levels noted in the experimental set-up were similarly evident with the entire orchestra present. Results of these readings are presented in Table 4 together with levels recorded during a performance with no acoustic screen present.

## DISCUSSION

In order to determine the effectiveness of bespoke acoustic screens designed for use in an orchestra pit, sound levels were measured with two trumpet players playing a short, relatively loud duet with the screens either removed altogether or in various configurations. These measurements were then repeated using pink noise at 80 dBA as a sound source. To verify these results and explore exposure levels during actual performances, sound levels were also monitored during opera and ballet performances and compared with previously published data at these points from the same orchestra in an identical pit set-up without an acoustic screen present [2].

Results showed the screens are effective at reducing sound exposure without increasing exposure to the musicians 'upstream'. The screens are most effective while configured with a longer upper panel, however as this causes visual interference between the brass section and the conductor it is not ideal. As use of the smaller upper panel still provides adequate protection for the musician in front of the screen this has been chosen as the preferred option in the orchestra pit during performances.

An important factor in the reduction of sound levels is the distance between the sound source and the individual/s requiring protection. This distance alone significantly reduces the intensity of sound reaching those in front as seen in the column labeled 'No Screen' in Tables 1-3. In this case, with no screen present the two meters between the trumpet 1 position and the 'exposed' position effectively attenuated the sound level by 6.3 dB, accounting for a 75% reduction in sound exposure consistent with fundamental acoustic principles. This emphasises the crucial role distance plays in an orchestra as an exposure control measure. When developing hearing conservation strategies or proposing orchestra and band layouts, the combination of effective acoustic screens with distance is a very powerful tool in sound level reduction.

To more clearly illustrate the impact this level of sound attenuation has during an actual performance, Table 5 details expected levels in front of the screen based upon sound exposure at the right ear of cello 6 (rear desk) as reported in Table 4. Allowable recommended noise exposure is based upon 85 dBALEQ for eight hours, with an exchange rate of 3 dB. As players are often required to perform twice in one day (with performances up to and sometimes beyond 180 minutes in length) it appears the use of this screen with either a low or full screen would allow those protected to perform without the need for personal hearing protection.

Compared to previous investigations the screens have been shown to be at least as effective as the individual wrap-around absorptive screens used in orchestras (reported as 8 dB one metre from the sound source using pink noise [6]) and more effective than various configurations of single Perspex sheets reported by Libera and Mace [5] with few of the difficulties noted when either of these individual screens are utilised.

The screens have been also reported to be effective beyond the orchestra pit when a drum kit is placed within or close to the orchestra, to protect musicians from loud cymbal crashes. In this instance the screens are deployed with the Perspex angled away from the drummer and the diffusive panels facing towards the drummer. Anecdotally, drummers employed with the orchestra have reported a preference for these screens over the simple perpendicular Perspex sheeting, citing a greater comfort, much less reflection of their own sound back to them and greater connection with the ensemble.

Additionally, the screens are regularly deployed around two meters behind the French horn section with the Perspex angled away from the musicians in venues where there is either absorptive material to the rear of the stage or a very deep stage. Horn players report greater ease of projection and less need to play at extreme volume with the screens in place. Horn players had also been using perpendicular Perspex sheeting for this purpose prior to the

Table 4. Sound exposure during opera and ballet performances, various screen positions. All data is presented in dBALEQ

| Position | No Screen (ballet) | Short Screen (ballet) | Long Screen (opera) |
|---|---|---|---|
| Trumpet 1 | 90.1 | 91.4 | 90.1 |
| Cello 6 | 89.1 | 85.4 | 82.9 |
| Difference between trumpet 1 and screened position | 1.0 | 6.0 | 7.2 |

Table 5. Measured exposure and time to reach allowable recommended noise exposure in front of acoustic screen based on exposure reported in Table 4

| | No Screen (ballet) | Short Screen (ballet) | Long Screen (opera) |
|---|---|---|---|
| Measured exposure in front of screen - cello (dBALEQ) | 89.1 | 85.4 | 82.9 |
| Time to reach recommended allowable noise exposure (minutes) | 186 | 438 | 780 |

introduction of the screens and report greater aural comfort when playing at high volume with the bespoke screens.

This study was limited in that only one orchestral configuration within a single venue is presented. As previously reported, orchestral sound exposure is highly variable according to repertoire, instrument played, venue and orchestral configuration [2]. As such the effect of placing the described screens in alternate positions within the orchestra is still a matter of speculation. It is possible, for example, that the placement of instruments producing significantly higher sound levels in front of these screens may exacerbate exposure levels to these musicians. This requires further investigation.

## CONCLUSIONS

The acoustic screens described and investigated in this study were developed in the field over several years. They are a flexible and useful addition to hearing conservation control measures within a typical orchestra when used in the context of a detailed hearing conservation strategy. They both increase comfort and significantly reduce risk for those in need of protection from louder instruments to the rear without increasing the risk or discomfort to players of these louder instruments who have amongst the highest risk of overexposure to sound of any in the orchestra.

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# ON ABSORPTION AND SCATTERING COEFFICIENT EFFECTS IN MODELLISATION SOFTWARE

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This paper presents the results provided by two simulation programs for a very simple model: a cube in which all sides have the same absorption but different global scattering coefficients. Effects on merit figures of the main acoustic parameters are shown for scattering changes for a single absorption coefficient. The results are helpful for understanding the role of absorption and scattering on the values of these parameters.

## INTRODUCTION

The modelling of a room, hall, or concert hall by an acoustic simulation program involves two main processes: firstly, the development of a geometric model using a set of surfaces; and secondly, the allocation of a material for each of the surfaces. The most recent programs indicate absorption and scattering coefficients in frequency bands. The scattering coefficients are interpreted as the percentage of energy not specularly reflected. Energy conservation is given by the relationship

$$(1 - s)(1 - \alpha) + \alpha + s(1 - \alpha) = 1 \quad (1)$$

where $s$ is a scattering coefficient (the fraction of reflected energy that is not reflected specularly); and $\alpha$ is an absorption coefficient (the fraction of incident energy that is not reflected). The programs that include scattering give reliable results, as was remarked in the First International Round Robin on Room Acoustical Computer Simulations [1]. The use of scattering coefficients in simulation programs is a simple way to make these programs realistic. The coefficients can also serve as a tool that enable users to adjust their model to the experimental data [2].

Two commercial programs were used in this study: CATT-Acoustic v.8 with The Universal Cone Tracer (TUCT); and Odeon v10. Although both programs are based on ray tracing, the scattering treatments differ in each program.

CATT-Acoustic uses cone-tracing algorithms [3,4]. Accordingly, scattering is frequency dependent; while direct sound and first-order specular reflection are deterministic. From the second reflection, specular and diffuse reflection is performed randomly. Thus for the coefficient $s = 0.5$ (in CATT-Acoustic a number between 0 and 100 is used, while in Odeon a number between 0 and 1 is used), half of the rays are specularly reflected and the other half are diffused [4].

Odeon uses a hybrid algorithm [5]. For early reflections it uses the mixed method of images and ray tracing. For late reflections it uses a special ray tracing method, with secondary sources that radiate energy from the wall surfaces. The parameter transition order (TO) (default 2) defines the transition from early to late reflection methods. The program also specifies the number of rays included in the early reflection method.

For reflections with an order that is lower than the transition order (TO), Odeon determines the source image and includes the corresponding reflection in the reflectogram (if visible to the source). The attenuation of that reflection is determined while taking into account distance, scattering, and air absorption. Odeon continues with the mixed method until the TO order. The late-rays are treated as secondary sources and emit a diffracted beam in accordance with the Lambert distribution. The result is the sum of the two beams with the weights '1-$s$' and '$s$' (see Figure 1).

In brief, the difference between the two programs is in the first and second order reflections (TO by default). In the CATT case (Algorithm 1) the first order reflections are treated stochastically, whereas Odeon uses the mixed method of image and ray tracing.

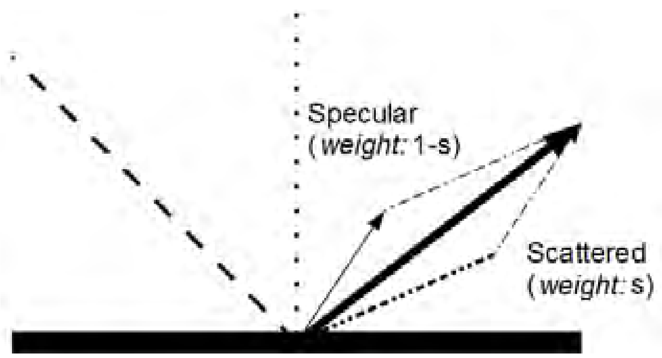



Figure 1. Energy conservation in Odeon algorithm [5]

## A SIMPLE GEOMETRY: 20 M CUBE

The cube is one of the simplest geometries that can be studied. The same absorption for all walls and frequency bands was given to minimise the model complexity when comparing the results of both programs. The scattering coefficient was uniformly distributed over all the surfaces and was varied for each simulation. A similar approach was found in the literature [6,7]. A deeper study on a complex room can be found in [8]. It must be underlined that this work does not discuss how each program works. As previously mentioned, the intention is only to compare the results provided by both programs for the cube model as this may serve as a valuable guide for program users.

The comparison of simulations was performed for the following absorptions: 10, 20, 25 and 30 (these figures cover the cases relevant for practice as average absorption coefficients are not usually larger than 30). For each mean absorption considered, the following scattering coefficients were used: 0, 20, 40, 60, 80 and 100.

Figure 2 shows the CATT-TUCT window that indicates the software options. The default options are used for the calculations. The auto measure is used for the ray number. Similarly, auto setting was chosen for the impulse response length. The air absorption option is also marked.

In Odeon, the default options used were provided by the ‘engineering’ button. The program asks the user to select the impulse response duration. It is set at 5000 ms, which represents the time used by CATT-TUCT as a default option (see Figure 3).

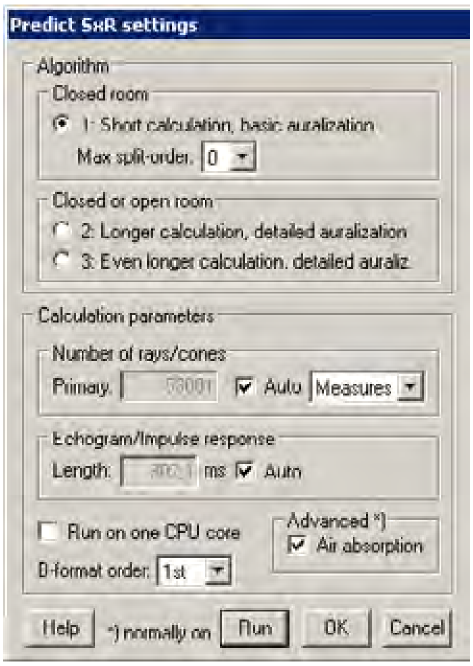


Figure 2. CATT-TUCT default options

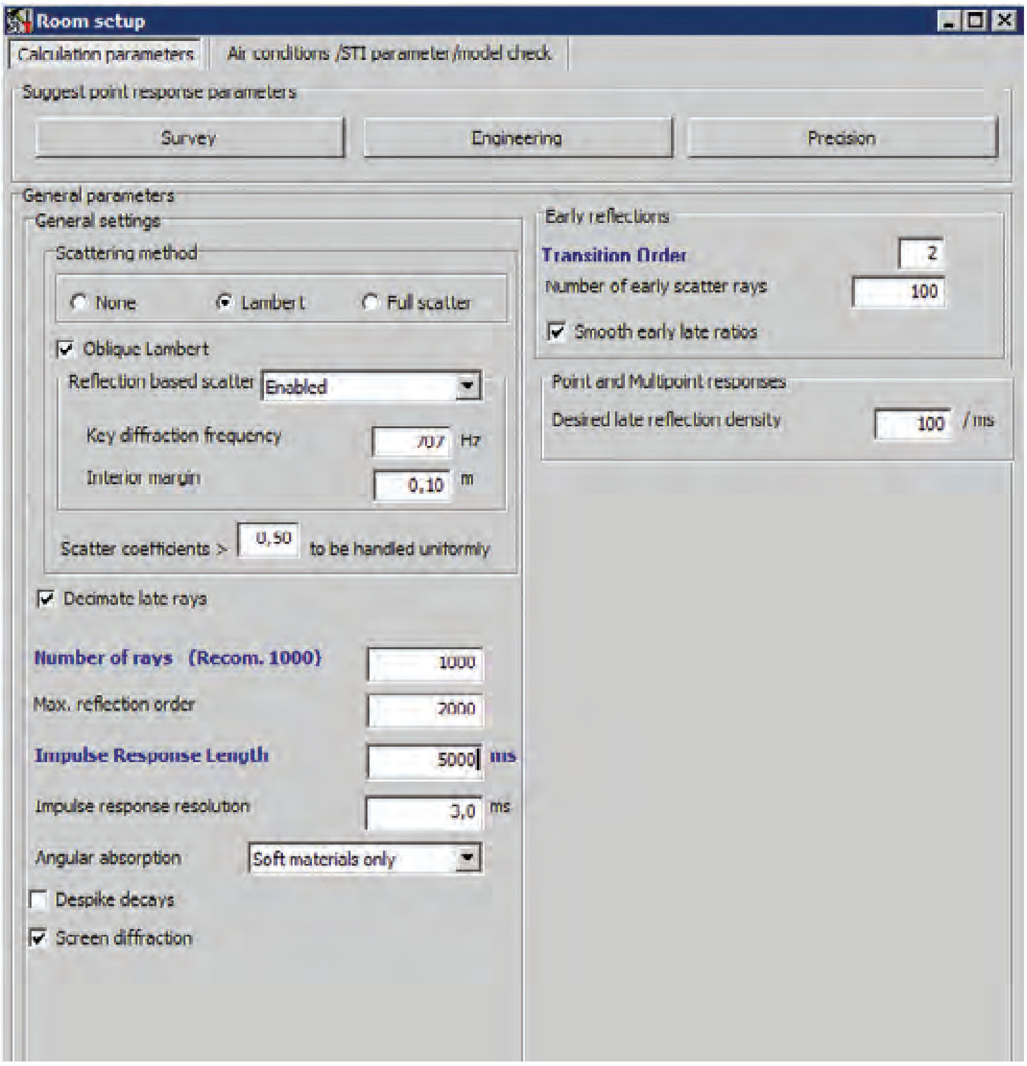


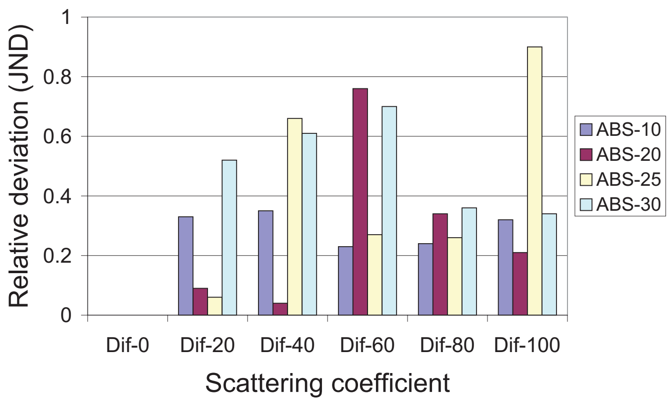
Figure 3. Odeon default options

**Stability of the solutions**

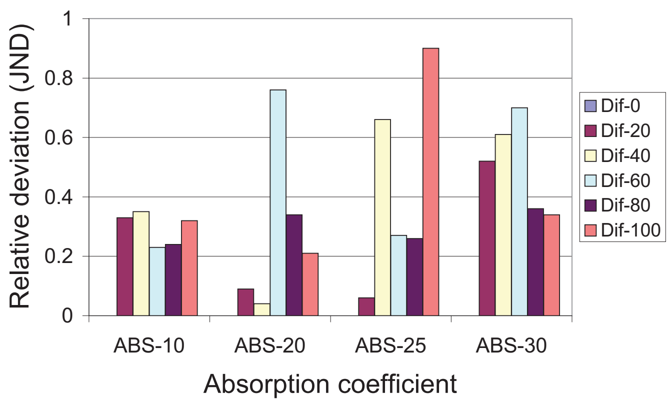
The following must be taken into account when choosing a program to calculate a prediction:

- Odeon provides a non-probabilistic solution. To repeat a calculation we have to 'trick' the program by some modification of the model. Otherwise, the program informs us that the task has already been done.
- CATT-Acoustic provides a non-deterministic solution if a non-zero scattering coefficient is entered. Therefore, it is necessary to study the stability of the solution.

To study the CATT-Acoustic stability, two calculations were performed for each variant of the model. Figure 4 shows the average variation in all frequencies for RT30 in units of relative deviation and taking the corresponding Just Noticeable Difference (JND) (5%) into account [9]. It can be seen that the effect of scattering does not produce significantly different results (i.e. for both calculations there is a difference of more than 1 JND in the RT30 prediction). This means that when working with the default options in CATT-Acoustic, the convergence of the solution and the number of rays and truncation time the program uses are guaranteed for the studied cases (usually averaged absorptions).




(a)




(b)

Figure 4. Variation of the CATT-Acoustic stability solution for RT30 for (a) various absorption coefficients for constant scattering coefficients and (b) various scattering coefficients for constant absorption coefficients, measured by relative deviation to JND

**Effect of scattering coefficient in each program**

For each program, the effect of the scattering coefficient with fixed absorption was studied when predicting different acoustic parameters. The values of each parameter were compared with the value obtained for a scattering coefficient '0' [6]. To avoid the commercial misuse of the results, we have omitted the name of the program. Programs 1 and 2 are used without identifying which is which. Average absorption coefficients are usually not larger than 30. This holds for large performance spaces, as well as for living rooms and classrooms.

Figure 5 shows the results for Program 1. To compare the differences obtained in this software, the same scale as the axis of deviation was used. Here are the comments for each parameter:

- EDT: for studied absorptions (coefficient ≤ 30), variations with the scattering coefficient do not exceed 2 JND.
- RT30: this parameter remains fairly stable with scattering variation for studied absorptions.
- SPL: remains stable in all studied cases (1 dB has been considered as JND).
- C80: remains stable in all studied cases (1 dB has been considered as JND).
- D50: shows great dependence on the scattering coefficient.
- Ts: No great variations shown.

In Figure 6, the results of Program 2 are presented. The same axis of deviation scale was used to compare the differences obtained with this software. The variation in parameters is generally greater than in Program 1. The comments for each parameter are given below:

- EDT: variations are close to 5 JND.
- RT30: this parameter remains fairly stable with the variation of scattering for studied absorption.
- SPL: variations of around 5 JND were observed for the studied absorptions (1 dB is considered as 1 JND).
- C80: remains stable in all studied cases (around 2 JND).
- D50: shows large variations of around 20 JND.
- Ts: shows variations of around 5 JND.

**Comparing the programs**

In Figure 7 the results obtained in both programs were compared. Average relative deviations of each value obtained in Program 1 with respect to the value obtained in Program 2 were calculated. The comments for each parameter are detailed below:

- EDT: differences do not generally exceed 4 JND for each absorption and scattering.
- RT30: differences do not generally exceed 2 JND for absorption and scattering used.
- C80: average differences obtained for each program did not exceed 4 JND.
- D50: large deviations between obtained values in each program are shown.
- Ts: differences generally do not exceed 4 JND for absorption and scattering.

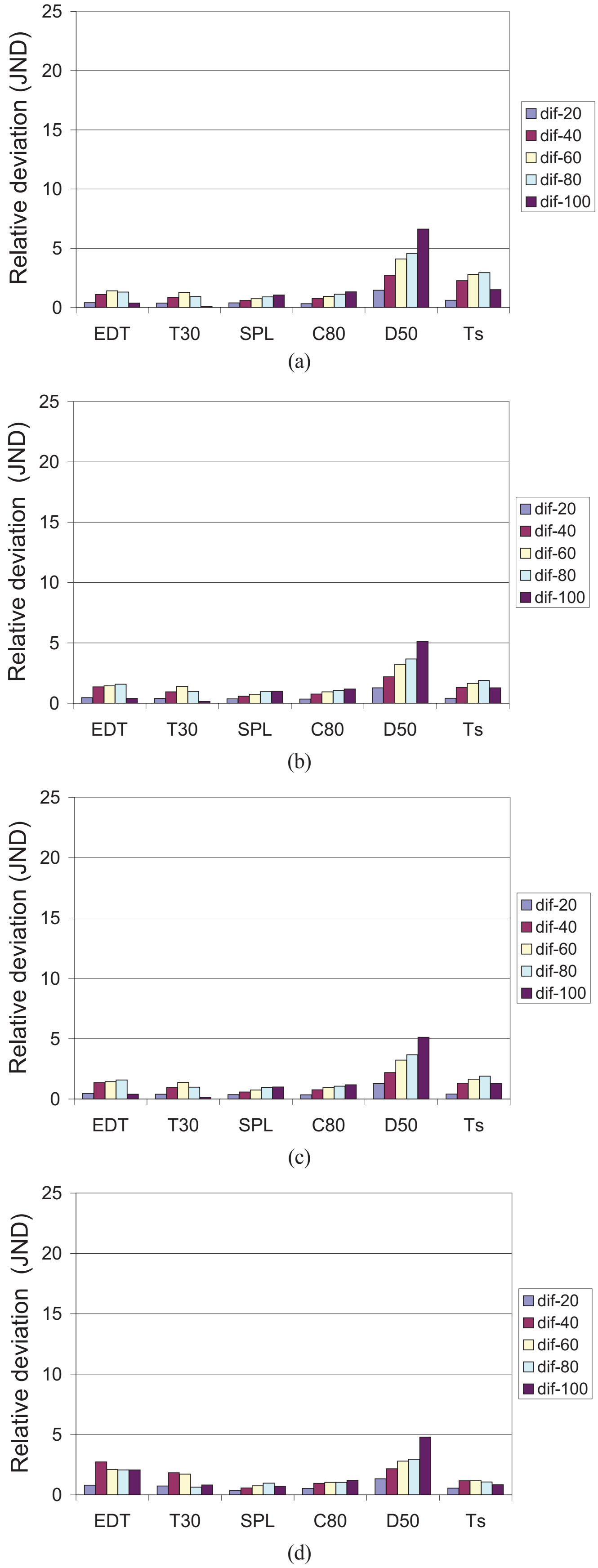



Figure 5. Parameter variation with the scattering coefficient for a given absorption (Program 1): (a) ABS-10, (b) ABS-20, (c) ABS-25, (d) ABS-30

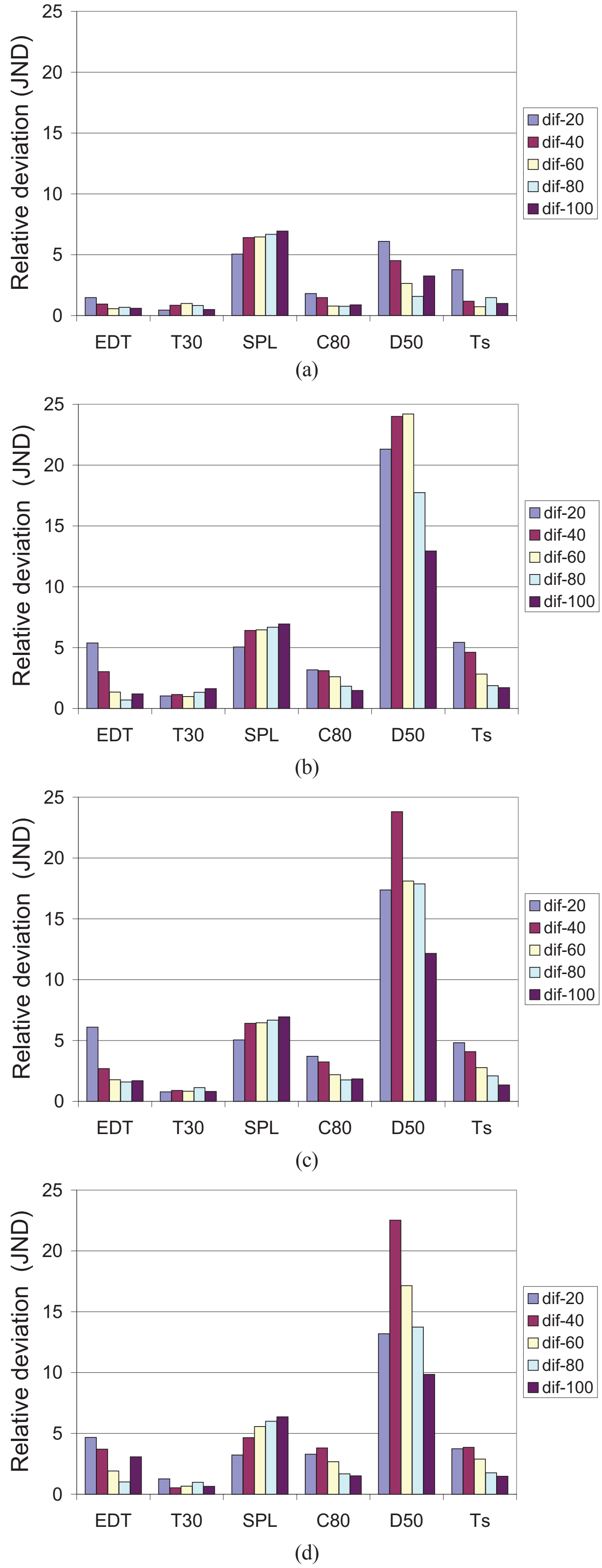



Figure 6. Parameter variation with the scattering coefficient for a given absorption (Program 2): (a) ABS-10, (b) ABS-20, (c) ABS-25, (d) ABS-30

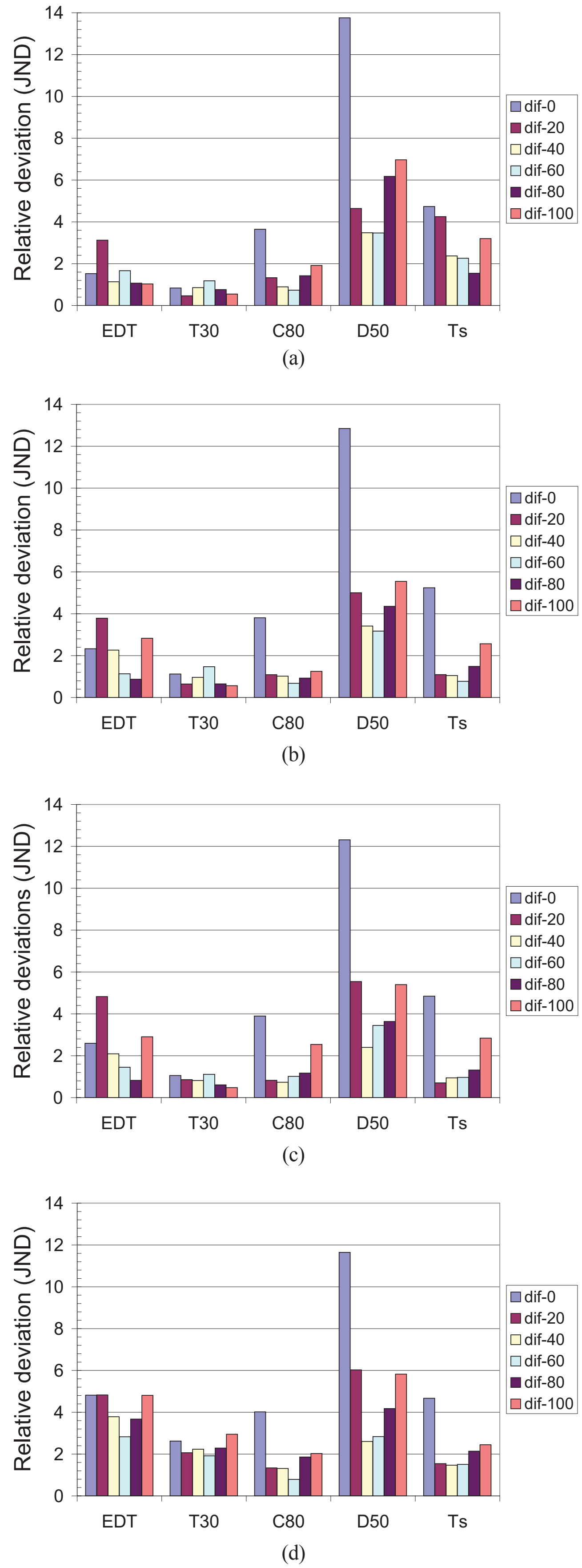



Figure 7. Relative deviation of JND between Programs 1 and 2: (a) ABS-10, (b) ABS-20, (c) ABS-25, (d) ABS-30

## SUMMARY

Studying the simplest geometry (with special acoustic characteristics) revealed differences in the simulation programs (Odeon and CATT-Acoustic) when working with the default options. In a simple approach these differences seem to be justified due to the different algorithms used by the programs. The authors' intention is not to discuss the causes of these differences, nor to discuss if one of the programs is better than the other. Future research should analyse these differences more deeply in real predictions. This case study, without being generalised to other models, leads to the following conclusions:

- When working with the default options both programs guarantee the convergence of the RT solution with the number of rays and truncation time that the program uses by default.
- Program 1 shows fewer variations than Program 2 with scattering coefficient changes. D50 is the most sensitive parameter to scatter variations.
- Although in general, the parameters obtained do not show great variations between programs, results show that D50 values are program dependent. Moreover, Ts and EDT sometimes show variation near 4 JND. These facts show that the differing algorithms and scattering treatments in each program produce considerable differences in early reflections.

## ACKNOWLEDGEMENTS

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# AN EXPERIMENTAL STUDY ON THE SOUND ABSORPTION OF THREE-DIMENSIONAL MPP SPACE SOUND ABSORBERS: RECTANGULAR MPP SPACE SOUND ABSORBER (RMSA)

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A microperforated panel (MPP) is usually placed with a rigid-back wall to form a Helmholtz resonator with its hole and the air-back cavity. However, the authors have so far proposed an MPP space sound absorber without any backing structure. In the previous studies, as a basic form of such an MPP space absorber, multiple-leaf MPP structures without a back wall were proposed, and were theoretically and experimentally examined. In order to provide more unrestricted usage and designs for an MPP space absorber, the authors have also proposed a three-dimensional MPP space absorber, called a cylindrical MPP space absorber (CMSA). The CMSA was shown to exhibit resonance peak absorption and additional low frequency absorption. In this paper, another alternative of a three-dimensional MPP space absorber, a rectangular MPP space absorber (RMSA) is proposed. Its sound absorption performance is discussed using experimentally measured results. The results show sound absorption characteristics similar to a CMSA, and an RMSA can be effectively used if properly designed.

## INTRODUCTION

A microperforated panel (MPP) is one of the most promising alternatives among so-called "next-generation sound absorbing materials". The use of an MPP solves problems associated with porous absorbing materials such as low durability, hygiene and low recyclability. An MPP is usually made of a thin panel or film (less than 1 mm thick) with submillimetre perforations with perforation ratio of less than 1%. The acoustic resistance and reactance suitable for sound absorbing materials is realised, and hence an MPP offers better sound absorption performance than ordinary perforation panels with larger perforations. An MPP was first proposed by Maa [1] in the 1970s. Maa developed the theory and design principle of an MPP as well as validated its effectiveness [2-4]. Many researchers have since presented studies on its application for various purposes [5-8].

The basic usage of an MPP is to place it in front of a rigid-back wall with an air-back cavity in-between. Helmholtz resonators are then formed with the holes of the MPP and the air-back cavity. The authors proposed a double-leaf structure of MPPs with an air-cavity in-between without a rigid-back wall, which is called a double-leaf MPP space sound absorber (DLMPP) [9,10], as well as a similar structure with three MPPs which is a triple-leaf MPP space sound absorber (TLMPP) [11]. The authors also proposed a space sound absorber with an MPP and a permeable membrane without any backing structure [12]. The sound absorption characteristics and the effectiveness of these sound absorbing structures were examined.

The DLMPP, TLMPP and MPP-membrane space absorbers are all in the form of a panel-like structure, which can be used as a sound absorbing panel or partition. However, this restricts the usage of these space sound absorbing structures due to its flat panel-like shape in some cases in actual rooms or buildings. To overcome these limitations, the authors proposed a light-weight three-dimensional MPP space sound absorber, which can be easily hung from the ceiling or put more freely on the floor. A cylindrical MPP space sound absorber (CMSA) which is made of an MPP shaped in a cylindrical shape was examined. The acoustic performance of the CMSA was experimentally studied [14]. CMSAs demonstrated moderate sound absorption performance which is rather similar to a DLMPP, and are considered to be a useful alternative for a sound absorber in rooms and buildings. A CMSA is also a good consideration as a sound absorption treatment in architectural design because it can be made of a transparent material as well as can be coloured.

Three-dimensional MPP space absorbers are a good sound absorption treatment if they can be made in various shapes. In this study, an alternative design corresponding to a rectangular pole (box-like shape with an air cavity inside) is constructed with MPPs, which is called a rectangular MPP space sound absorber (RMSA). The sound absorption characteristics of RMSAs are experimentally measured and their acoustic performance is discussed.

## EXPERIMENTAL PROCEDURE

The measurement of the random incidence sound absorption coefficient in a reverberation chamber was carried out based on JIS A 1409 (ISO 354 compatible). The measurements of the reverberation time were made for two sound source positions and at five microphone positions. The reverberation chamber used is of volume 513 $m^3$ and surface area 382 $m^2$. To obtain the absorption coefficient, the following equation based on the Sabine's formula is used:

$$\alpha = 55.3\ \frac{V}{cS}\left(\frac{1}{T_2} - \frac{1}{T_1}\right) \tag{1}$$

where $\alpha$ is the absorption coefficient of the specimen, $c$ is the speed of sound, $V$ is the volume, $S$ is the surface area of the reverberation chamber, and $T_1$, $T_2$ correspond to the reverberation times without and with the specimens, respectively.

The test specimens are two types: number 1 is with square cross section of 0.25 m. Number 2 is with square cross section of 0.5 m. Both are void (with air cavity) inside and of 1 m high: They are constructed with MPP leaves of 0.25 m x 1.0 m for number 1, and 0.5 m x 1.0 m for number 2. The MPPs are attached to a wooden frame of square pole to form a parallelepiped shape. The MPP used is of 0.5 mm hole diameter, 0.5 mm thickness, 0.785% perforation ratio and 0.6 kg/$m^2$ surface density, and made of transparent polycarbonate. A photograph of specimen number 1 is shown in Figure 1. The specimens set in the reverberation chamber are shown in Figure 2. In the both cases (numbers 1 and 2), the measurement was made with the top open ends either kept open or closed by removable covers.

In previous studies on a cylindrical MPP space sound absorber (CMSA), preliminary experiments were made to check the effect of area. In the present study, preliminary measurements were also made with 2, 4 and 6 specimens (which are separated to each other by 1 m distance, placed on the central area of the floor of the chamber) to observe if the absorption power per one specimen changes with the number of specimens. However, as in the case of a CMSA, the absorption power per one specimen did not change when the number of specimens was changed. As in the CMSA cases, the sound absorption power obtained in the experiment is normalised by the total of the surface area of the specimens so that the value equivalent to the sound absorption coefficient is obtained. All measured data in this work are with six specimens. The absorption coefficient corresponds to the sound absorption power per unit surface area of the specimen by normalising the measured results with the total surface area of the six specimens. Hence in the case of number 1, the specimen surface area is 1 $m^2$ and the sound absorption power per one specimen is equivalent to the sound absorption coefficient. In the case of number 2, the specimen surface area is 2 $m^2$ and the measured value normalised by 2 is equivalent to the sound absorption coefficient.

Figure 1. A photograph of the specimen of RMSA used in the experiment (number 1 with a removable cover on the top end)

Figure 2. The specimens of rectangular MPP space sound absorbers (RMSA) set in a reverberation chamber. In this photo the top ends of the specimens are closed with removable covers

## EXPERIMENTAL RESULTS

### Case number 1

The measurement results for number 1 are shown in Figure 3. In the graph both the open end and closed end cases are shown for comparison. As a general feature, the RMSA shows a resonance absorption peak (in this case at around 500 Hz). At low frequencies around 200 Hz the absorption coefficient still keeps a constant value around 0.2, which is similar to CMSA, DLMPP and other space absorbers with permeable materials. This is caused by the acoustic resistance of the MPP and particularly for permeable structures, as inferred in the study on CMSA [14]. The effect of the cover appears around the resonance peak: the peak is more significant when the top end is closed by a cover. This is also a similar feature to a CMSA. Therefore, the sound absorption mechanism of the RMSA is considered to be similar to that of a CMSA.

**Case number 2**

The measurement results for number 2 are shown in Figure 4. In the graph both the open end and closed end cases are shown for comparison. On the contrary, in this case a significant peak does not appear. This is attributed to the fact that the cavity inside becomes too large to cause a resonance: the resonance peak does not significantly appear even in the case of a typical single-leaf MPP with a rigid-back wall [13]. This phenomenon has also been discussed in the case of a CMSA with larger dimensions [14]. When the top end is closed by a cover the absorption coefficient becomes somewhat higher, and a broad peak-like feature appears. This is also similar to CMSA cases. The RMSA does not show a significant resonance peak when the size becomes too large, which is similar to a CMSA. Therefore, the size of the RMSA is critical in the design of a three-dimensional MPP space sound absorber, that is, the size should not be too large.

**Other considerations**

In this section, further discussion on a rectangular MPP space sound absorber (RMSA) is given in what follows. Firstly, it is useful to discuss the difference between the RMSA and the common MPP sound absorber with a rigid-back wall. Although the RMSA shows a resonance peak absorption which is similar to the common MPP absorber with a back wall, the peak is lower. This feature also occurs in the case of a cylindrical MPP space sound absorber (CMSA) and is attributed to the fact that the boundaries that form the cavity are not rigid. On the other hand, the additional low frequency absorption, which is not produced in the common MPP absorber, occurs in both a RMSA and a CMSA. Therefore, although the peak for a RMSA and a CMSA is lower than that for the common MPP, these absorbers can cover a wider frequency range.

The rectangular and cylindrical MPP space sound absorbers show very similar sound absorption characteristics as described previously. Both absorbers have lower resonance peak absorption as well as additional low frequency absorption. However, the peak value tends to be higher in the CMSA case compared to the RMSA case. The reason why the RMSA peak is lower than CMSA is subject to discussion in the future study, although one possible reason is because of the difference in the angle of incidence to the sound absorber. In the cylindrical case (CMSA), even though the sound is incident randomly, the absorber can be regarded to behave as though the sound has normal incidence. However in the rectangular case (RMSA), the incident wave has a certain angle to the surface. Since the MPP shows the most efficient absorption in the normal incidence case, this fact makes the peak for the RMSA lower than for the CMSA.

It is also useful to consider the effect of the floor on the sound absorption by a RMSA. In the present work the measurements were only made with specimens placed on the floor. However, in the previous study on the CMSA, the measurements were conducted with specimens set apart from the floor [14]. According to Ref. [14], when a CMSA is set apart from the floor, though the difference is very small, the peak absorption becomes slightly higher, especially in the case when the ends are closed by the cover. The reason of this feature is not clarified, however, the same effect can be expected in the present RMSA cases.

Finally, it is of interest whether it is possible to use an alternative material to construct a similar sound absorbing

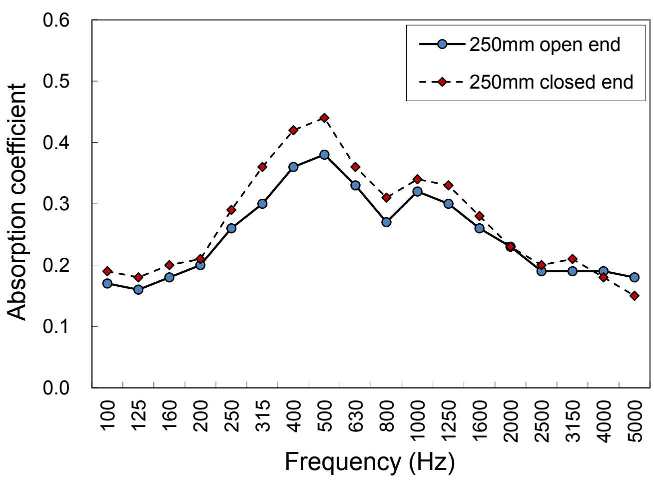


Figure 3. Measured results of the rectangular MPP space absorbers (RMSAs) of 250 mm x 250 mm square section. Solid line: open top ends; Dashed line: closed top ends with covers

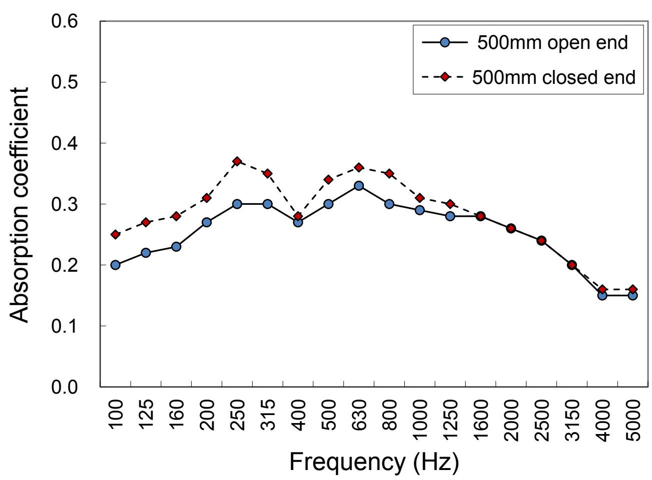


Figure 4. Measured results of the rectangular MPP space absorbers (RMSAs) of 500 mm x 500 mm square section. Solid line: open top ends; Dashed line: closed top ends with covers

system. According to previous studies on membrane-type space absorbers [15, 16], a permeable membrane can be used to produce a space absorber similar to a CMSA or RMSA: in that case no resonance peak will appear but almost flat sound absorption characteristics may be obtained due to its acoustic flow resistance. On the other hand, if an impermeable membrane is used, it is not expected to obtain sound absorptivity: a resonance peak of membrane-type absorption could be produced if the cavity depth is appropriate [17]. However, if the cavity is too large, the resonance will be very weak, and it is difficult to make an efficient sound absorber.

## CONCLUDING REMARKS

To develop a more flexible design, easy-to-use space sound absorber with MPPs, a rectangular MPP space sound absorber (RMSA) is proposed as an alternative to a three-dimensional MPP space sound absorber. The sound absorption characteristics of the RMSA are measured experimentally in a reverberation chamber and the results are discussed.

The RMSA shows characteristics similar to a cylindrical MPP space sound absorber (CMSA) or a double-leaf MPP space sound absorber (DLMPP) and other space sound absorbers with MPPs: it exhibits a resonance absorption peak and additional low-frequency absorption due to the acoustic resistance. Although the sound absorption coefficient is not very high as it covers a wide frequency range, RMSAs are useful as an alternative sound absorption treatment to control the acoustic environment in rooms and buildings.

In order to effectively use RMSAs (as well as CMSAs), it should be noted that the size should not be too large. When the size is too large (0.5 m in this study), the resonance absorption peak does not appear and the absorber becomes less efficient. The absorber is more efficient when its ends are closed by covers.

In order to design and predict the sound absorption performance of an RMSA more efficiently, it is necessary to theoretically predict its acoustic performance. Furthermore, other three-dimensional shapes should be considered for wider variation of designs. These will be discussed in future work.

## ACKNOWLEDGEMENTS

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# THE DIVISION OF THE PERCEPTUAL VOWEL PLANE FOR DIFFERENT ACCENTS OF ENGLISH AND THE CHARACTERISTIC SEPARATION REQUIRED TO DISTINGUISH VOWELS

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The results of an on-line study of vowel recognition by English speakers are analysed. A relatively unused region of the perceptual vowel plane is identified at about ($F2$, $F1$) = (1800 Hz, 350 Hz). The rest of the plane is divided among vowels in ways that differ somewhat for different countries and regions thereof. Vowel length is used in several cases to help distinguish vowels whose distributions overlap substantially in ($F2$, $F1$). When the fundamental frequency is higher, the values of $F1$ and $F2$ are also higher, though much less than proportionally. This is consistent with the observation that women's vocal tracts are usually shorter than men's. The characteristic separations required to distinguish vowels in the ($F2$, $F1$) plane were 115 Hz and 292 Hz in the $F1$ and $F2$ directions respectively, with similar values in different countries.

## INTRODUCTION

This paper analyses results collected by an on-line study of vowel recognition. Its aims are to compare accents of English speakers in different provinces and countries by identifying the regions of the perceptual vowel plane that correspond to a given vowel. It also aims to quantify how far an intended vowel may be displaced on the vowel plane from its mean position before it ceases to be recognised. These aims could, in principle, be achieved in a laboratory study. On a large scale, however, such a study would be laborious and expensive. The advantage of this on-line study is that it is automated and that, following its launch five years ago, it has had large scale and wide-ranging international participation.

The method of the study was reported in detail by Ghonim et al. [1], where some preliminary results were reported. Briefly, the survey has a large set of synthesised sounds of the form h[vowel]d, chosen because nearly all such combinations are real English words. On-line volunteers listen to a synthesised sound and choose, from a list on their screen, the h[vowel]d word they think the sound most resembled, or else judge it unrecognisable. Their choice and the parameters used to synthesise that vowel are then recorded in a database. They then progress to the next sound. At the start of a session, each respondent gives information about their native language, their regions of birth and residence, their gender, age and some other details about their linguistic history and environments.

Much of the phonemic information in the vowels of English is contained in the first two formant frequencies, $F1$ and $F2$. These formants are broad peaks in the spectral envelope produced by the first two resonances in the vocal tract [2,3].

The study by Ghonim et al. [1] uses a synthesis method developed by the authors for the purpose [1]. It samples the vowel plane in 50 Hz steps between the boundaries shown in Figure 1. Two other parameters are varied: the vowels can have two different lengths ($t$ = 120 and 260 ms, hereafter 'short' and 'long') and two different initial fundamental frequencies ($f_0$ = 126 and 260 Hz, hereafter 'low' and 'high'). The number of sounds identified by each subject depends on their good will and patience. However, over all subjects, points in the space ($F2$, $F1$, $t$, $f_0$) are presented in a pseudo-random order so that each point has a similar number of occurrences.

The present paper analyses the results from this study, shows how the perceptual ($F2$, $F1$) plane is divided among vowels and unrecognised regions, and how this division depends on vowel length and $f_0$. It then uses the data to determine how the chance of identifying a sound as having a particular vowel varies as a function of the distance from the sound having the mean values ($\overline{F2}$, $\overline{F1}$) for that vowel. Using this function for each vowel, the characteristic distances on the perceptual ($F2$, $F1$) plane that are required to distinguish different vowels are calculated. The vowel plane is usually plotted as ($F2$, $F1$) with the direction of the conventional axes reversed; this is to preserve a similarity to the phoneticians' plot of mouth opening versus position of the tongue constriction. This tradition has been followed in this work.

## RESULTS

### The data set

40.5% of respondents used headphones and 59.5% loudspeakers. The frequency range of $F1$ often lies in a range over which the gain of radiating loudspeakers varies strongly with frequency, so it was of interest to see whether this made a difference to results. Averaged over all vowels, the shift in mean frequency of ($F2$, $F1$) from headphones to loudspeakers was (5.5 Hz, –7.6 Hz) for the survey population. This shift is insignificant in comparison with the sampling interval on the ($F2$, $F1$) plane (±50 Hz) and consequently headphone and loudspeaker data are pooled in all the subsequent analysis.

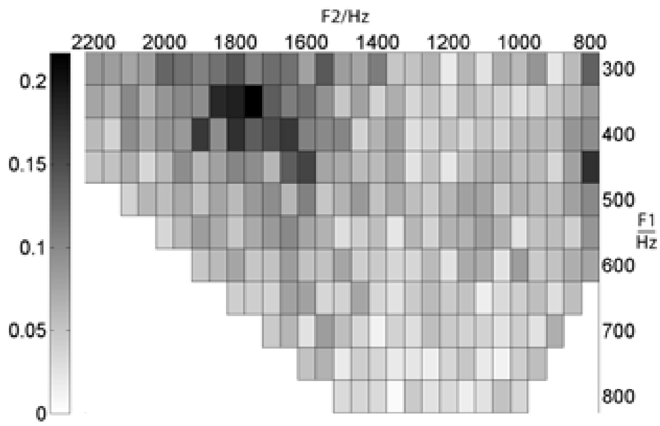



Figure 1. Distribution of unrecognised sounds on the perceptual ($F2$, $F1$) plane as a fraction of all choices by all respondents. The grey scale on the left indicates the fraction of sounds that were not recognised as any word.

**Unrecognised sounds**

The grey scale in Figure 1 shows the fraction of sounds that were not recognised as any of the listed words. Over the whole parameter space and for all respondents, the fraction of sounds that were not recognised as any of the words is 6.5%. In one area of the plane, near (1800 Hz, 350 Hz) or between 'heed' and 'who'd' in US, Australian or UK English, the proportion rises to 15-20%, suggesting that this area of the plane is not so much used in the accents of English most represented in this study, which are American, Australian and British. Other local areas of low recognition occur on the right, at very low values of $F2$.

Figure 2 shows the percentage of tokens over all of the parameter space that were recognised as each of the listed words by respondents born in the US (202 male respondents, 193 female), Australia (54 male, 49 female) and the UK (49 male, 18 female). They are grouped into words with monophthongs without the letter *r*, words containing the letter *r*, words that are often pronounced with diphthongs and those which were unrecognised. In what follows the effects of diphthongs and the letter *r* are discussed.

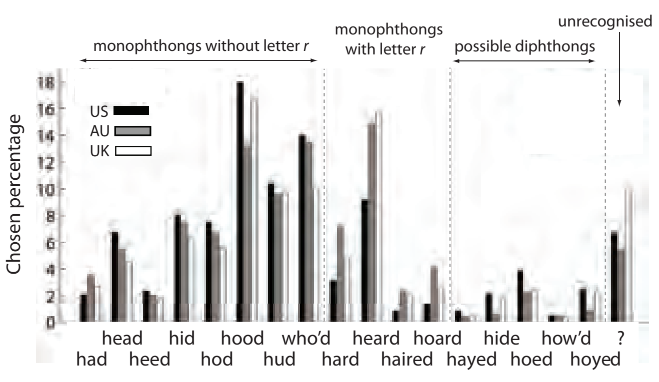



Figure 2. Percentage of words chosen by respondents born in US, AU and UK. '?' indicates that the vowel was unrecognised.

**Diphthongs and *r***

The sound samples did not include any words synthesised with a diphthong. Nevertheless, the survey allows respondents to chose words that would, in most Australian speech, be pronounced as h[diphthong]d: *hayed*, *hide*, *hoed*, *how'd*, *hoyed*. They were included because it is conceivable that some of these words might be pronounced as monophthongs in some accents. In practice, few respondents chose these words: over all sounds, *hayed* (0.8%, 0.3%), *hide* (2.1%, 0.6%), *hoed* (3.9%, 2.2%), *how'd* (0.5%, 0.5%), *hoyed* (2.5%, 0.9%) where the two values are respectively for respondents born in the US and Australia. These results suggest that more Americans than Australians recognise these words as monophthongs.

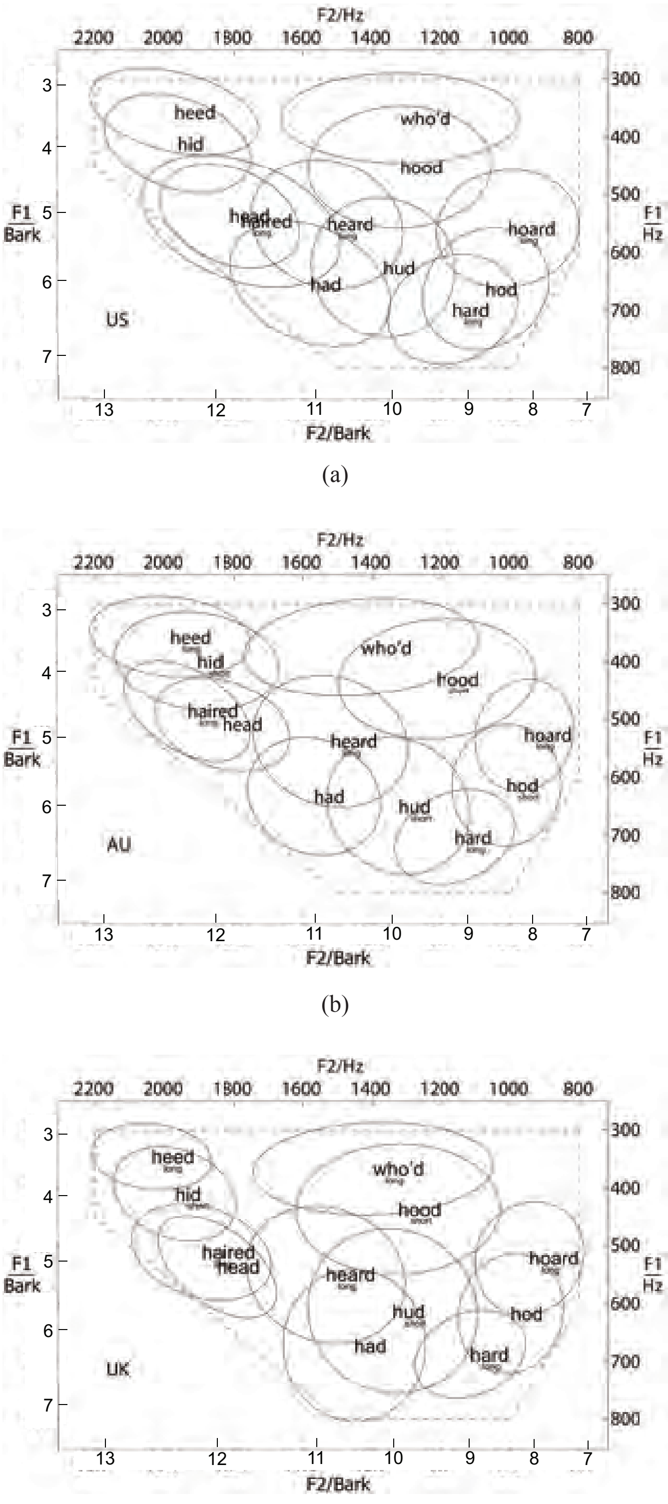
The sound samples did not include any rhotic *r* sounds. Nevertheless, the survey allows respondents to choose the words *haired*, *hard*, *heard* and *hoard*. In Australian English, and in some other varieties, these words are often pronounced without the rhotic *r* as h[vowel]d, but this is less frequent in the US. Figure 2 shows that each of these words was chosen by a higher proportion of respondents born in Australia than the proportion of residents born in the US. The proportions of Australians who chose these words when the vowel was long and short were: *haired* (98.1%, 1.9%), *hard* (93.4%, 6.6%), *heard* (92.1%, 7.9%) and *hoard* (96.8%, 3.2%). This is consistent with the observation that *r* in Australian English has the effect of lengthening the preceding vowel [4].

**Distribution of vowels in different regions**

Figure 3 shows the distribution of vowels on the perceptual ($F2$, $F1$) plane for respondents born in the US, Australia and the UK. For each group of respondents, the centre of each ellipse shows the mean values, the direction of the major axis is the line of regression and the semi-axes show the standard deviations in that direction and the direction at right angles to the line of regression. The gap between 'heed' and 'who'd' (mentioned above in the context of unrecognised vowels) is less noticeable in the Australian than in the American or UK data.

'Short' or 'long' printed below one of the words in Figure 3 means that more than 75% of the selections of that word were from the short or long sound samples, respectively. (On the average, each respondent should have received equal numbers of short and long sounds). This difference explains the overlap of some of the vowels: in the Australian and UK data, the distinction between *heed* and *hid* is largely made by vowel length, rather than position in the perceptual ($F2$, $F1$) plane. It is also important in distinctions between *hud* and *hard*, *hod* and *hoard*, and *head* and *haired*. This effect is smaller in the US data.

If we consider only the words that do not contain r and that are not possible diphthongs, Figure 2 shows that the words that are least chosen by both US and Australia respondents are *had* (2.1%, 3.5% respectively) and *heed* (2.4%, 2.0%). For *heed*, the alternative choice in that region is either *hid* or 'unrecognised'. For *had*, there is no nearby peak in the 'unrecognised' choice, but there is competition for much of that region of vowel space from several other vowels. Conversely, Figure 3 shows that there are few vowels at the top right of the plane, so *hood* and




(a)

(b)

(c)

Figure 3. Vowel distribution and standard deviation ellipses for (a) US, (b) Australian and (c) UK respondents to this survey. The dashed line shows the limit of the perceptual ($F2$, $F1$) plane sampled in these two dimensions.

*who'd* have high values in Figure 2. The word with the neutral vowel, *heard*, is also chosen frequently.

Figure 4 compares the results of New South Wales (38 respondents) and Queensland (18 respondents). For these populations, seven of the vowels showed differences significant at the 95% level (Figure 4). Averaged over all vowels, $F1$ was larger for NSW by 17 Hz, and $F2$ by 32 Hz. Both $F1$ and $F2$ increase with increasing mouth aperture so, on its own, this suggests that Queenslanders, on average, open their mouths less widely than New South Welshmen. However, the Queensland means are usually closer to the edges of the vowel plane, suggesting that Queenslanders use more of the vowel plane and thus have larger differences between vowels. It should be remembered, however, that 32 Hz is still smaller than the separation between harmonics in this study.

Among the three US states with the largest number of respondents – California (47), New York (33) and Ohio (20) – the differences were smaller than those between New South Wales and Queensland. At the 95% level, significant differences were found for only three vowels between California and Ohio (*had*, *hod*, *who'd*), three vowels between New York and Ohio (*had*, *heed*, *who'd*) and two between California and New York (*had*, *heed*).

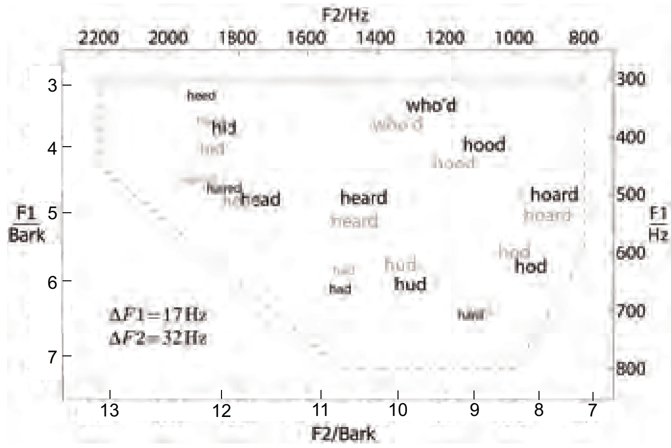



Figure 4. Shift in mean formant frequencies from Queensland (black) to New South Wales (grey). Those in large font are significantly different at the 95% level. For each word, $F1$ lies on the mid-line of the word, and $F2$ immediately to the left of the word.

## Vowel length

In several cases, vowels whose ellipses overlap significantly when plotted as in Figure 3 were, in part, distinguished by vowel length. Table 1 plots, for each word and each of the US, UK and Australia data, the fraction of choices that were long vowels. Thus *hard* was usually chosen when the sound sample had a long vowel, which distinguished it from *hud* and *hod*, which are nearby on the vowel plane for all these countries. *heed* and *hid* are distinguished by length in all these countries, though the difference is slightly less in the US.

We looked for patterns in the displacement on the perceptual vowel plane between the long and short versions of the same chosen word. $F1$ increases with mouth aperture

Table 1. The percentage of choices that were for a long vowel, for each word and for each of three countries. Bold font highlights values above 75% or less than 25%, that is, words that are classed as long or short in Figure 3.

| %long | had | head | heed | hid | hod | hood | hud | who'd | hard | heard | haired | hoard |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| US | 61.0 | 41.9 | 74.5 | 27.9 | 54.3 | 29.1 | 26.2 | **72.4** | **87.6** | **90.5** | **80.0** | **87.7** |
| AU | 45.7 | 31.3 | **80.9** | **15.8** | **18.4** | **15.4** | **5.3** | 63.6 | **93.4** | **92.1** | **98.1** | **96.8** |
| UK | 55.9 | 29.5 | **88.2** | **11.7** | 28.1 | **15.1** | **5.4** | **79.7** | **98.9** | **98.0** | **95.9** | **98.0** |

and so, to a lesser extent, does $F2$. Perhaps sustained vowels give the speaker more time to open the mouth. If so, one would anticipate the longer vowels to be displaced down and to the left on the perceptual plane. There was no such effect, nor any other consistent pattern in the US, UK and Australian data, and the average shift for these pairs was only several Hz. These results differ from an earlier perceptual study [5], where shifts in $F1$ and $F2$ were recorded for vowel lengths similar to those studied here.

**Dependence on $f_0$**

The reason for including high and low $f_0$ was to simulate the difference between male and female voices. Acoustic measurements of the vocal tract resonances of young Australian men [6] and women [7] showed that the resonant frequencies used by women for a given vowel are typically higher than those used by men, which is traditionally explained by observing that women, on average, have shorter vocal tracts than men. Positive shifts on the perceptual ($F2$, $F1$) plane for synthetic vowels have been reported previously [8,9] so one might expect a similar result for formants in this perceptual study. Figure 5 shows the displacements of the vowels (low to high) for the Australian data. The displacements are positive in $F1$ and $F2$, as expected.

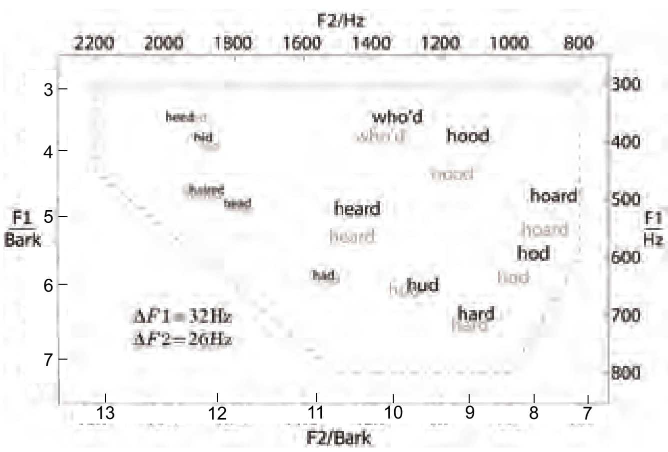



Figure 5. Shift in mean formant frequencies from the 'male voice' ($f_0$ = 126 Hz), printed in black to the 'female voice' (260 Hz, grey) in the Australian data. A large font indicates that the difference is significant at the 95% level.

**Characteristic displacements for vowel recognition**

A sound with values of $F1$ and $F2$ corresponding to the centre of one of the ellipses in Figure 3 has a high chance of being recognised as containing the vowel indicated on that ellipse: it has the mean values of $F1$ and $F2$ for that vowel identified by all respondents from that country. For sounds displaced significantly from that point, the chance of being thus identified falls. How far can a vowel 'stray' on the vowel plane before it ceases to be recognised? To answer this, the chance of being thus recognised as a function of distance on the vowel plane from its mean value was plotted. Distance on the perceptual vowel plane could be measured in Hz, but this would over-represent displacements in the $F2$ direction, because $F2$ is distributed over a larger range of frequencies. In a previous paper [10], a non-dimensional displacement $d$ on the vowel plane was defined. The Pythagorean distance between two points $a$ and $b$ on the plane was scaled by the standard deviations $\sigma_{F1}$ and $\sigma_{F2}$ of all vowels in the $F1$ and $F2$ directions to give the dimensionless separation:

$$d = \sqrt{\frac{(F1_b - F1_a)^2}{\sigma_{F1}^2} + \frac{(F2_b - F2_a)^2}{\sigma_{F2}^2}} \quad (1)$$

where $\sigma_{Fi}$ is the standard deviation in $F$i over all vowels, which in this case is $\sigma_{F1}$ = 147 Hz and $\sigma_{F2}$ = 374 Hz. So, for a particular vowel $v$, whose mean value on the recognition plane occurs at ($\overline{F2}$, $\overline{F1}$), the fraction $f_v$ of vowels recognised as $v$ is plotted as a function of the radial distance $d$ from ($\overline{F2}$, $\overline{F1}$).

The ellipses in Figure 3 show that the spread of vowel recognition is large and that there is considerable overlap. It is therefore interesting to ask how much of this spread is due to variation among respondents and how much to variation in the choices made by each individual respondent.

Figure 6 shows $f(d)$ for the respondents born in Australia. It also shows $f(d)$ for one Australian-born respondent who had a relatively large number of sample responses, and thus gave reasonably good statistics. At $d$ = 0, the rate of recognition by the single subject was about 60% while for the population it was about 25%. Of course, the plot shows that, even for one subject, a vowel occupies a finite area on the plane. For a large population, which may have and be familiar with different accents, the distribution for each vowel is larger than for an individual.

Dowd et al. [10] fitted both exponential ($a_0 e^{-d/\lambda}$) and Gaussian functions ($b_0 e^{-d^2/2\sigma^2}$) to $f(d)$, so we fit those functions here, to give two characteristic, non-dimensional distances, $\lambda$ and $\sigma$ respectively. In the present study, the Gaussian appears to be a rather better fit (Figure 6). For the Australian individual and the Australian sample data, the values of $a_0$ are respectively 0.63 and 0.27, $b_0$ are 0.56 and 0.23, values of $\lambda$ are 0.94 and 1.00, while those of $\sigma$ are 0.62 and 0.74 respectively. These values of $\sigma$ correspond to 86 and 229 Hz in the $F1$ and $F2$ directions respectively for the individual, and 109 and 277 Hz for the population. The population values are surprisingly similar to those of [10], who reported 105 and 279 Hz. Direct comparison between them is not advised, however: in the Dowd et al. [10] study, we used acoustic measurements of the tract resonances, not formants, we used real human speech, not synthesis, and the language studied was French, not English.

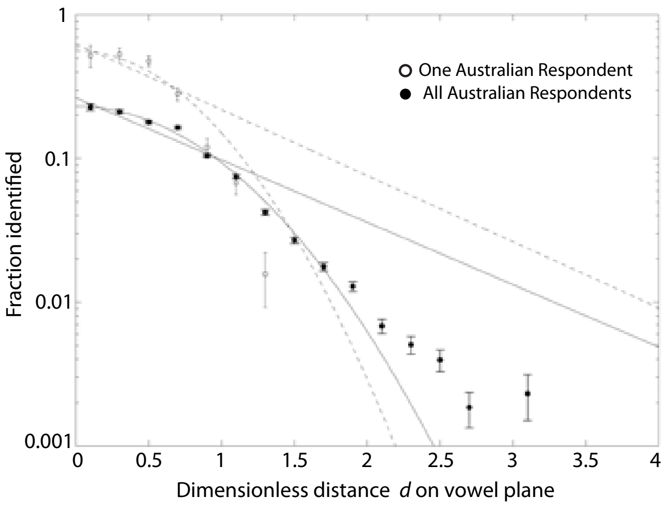



Figure 6. The fraction of sounds identified as a having a particular vowel plotted as a function of the dimensionless Pythagorean distance ($d$) from the mean position for that vowel. The data are averaged for all vowels. Solid lines and black points are for the Australian population. Dashed lines and grey points are for one Australian respondent with a large data set. The straight and curved lines indicate the results of an exponential or Gaussian fit respectively to the data. Error-weighted fits are used, hence points with large values of $d$ that were chosen infrequently do not contribute strongly to the fit.

The values of $\sigma$ from the Gaussian fits are listed in Table 2 for all respondents and for the five countries having the greatest numbers of respondents (Australia, US, UK, Canada and France). There is little variation among these.

Table 2. The characteristic distance required to distinguish vowels (Gaussian model). $\sigma$ is the dimensionless separation defined by equation (1) and $\sigma_1$, $\sigma_2$ the separations in $F1$ and $F2$ respectively.

| Population | AU | US | UK | CA | FR | ALL |
|---|---|---|---|---|---|---|
| $\sigma$ | 0.74 | 0.79 | 0.76 | 0.70 | 0.79 | 0.78 |
| $\sigma_1$/Hz | 109 | 116 | 112 | 103 | 116 | 115 |
| $\sigma_2$/Hz | 277 | 295 | 284 | 262 | 295 | 292 |

**Future use**

One possible use of the data gathered by this survey might be voice synthesis that is tailored for different regions. To obtain finer detail, it would be necessary to advertise the survey in the required geographical regions.

The survey [11] has run for only a few years, so it is too early to look for evidence of vowel drift with time. It would be interesting, however, to study changes on a time scale of decades, as suggested by Mannell [12]. The authors are prepared to make data available to other researchers, subject to conditions that include the anonymity of the data being met.

## CONCLUSIONS

This survey quantifies the vowel plane for several countries and regions thereof. A relatively unused region of the vowel plane is identified at about ($F2$, $F1$) = (1800 Hz, 350 Hz). In several cases, vowel length helps distinguish vowels that overlap on the plane. The values of $F1$ and $F2$ rise slightly when the fundamental rises from typical women's to men's range. Using a Gaussian model for vowel distribution, the characteristic separations required to distinguish vowels in the ($F2$, $F1$) plane were respectively 115 Hz and 292 Hz in the $F1$ and $F2$ directions.

## ACKNOWLEDGEMENTS

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# SHOCK WAVES AND THE SOUND OF A HAND-CLAP — A SIMPLE MODEL

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The aerodynamics of the impact between two human hands in a hand-clap is examined, in particular in relation to the hand profile which may be either nearly complementary between the two hands, giving a nominally flat impact, or else domed so that there is a significant enclosed volume. It is shown that shock waves are generated in nearly all hand-claps, with the addition of a Helmholtz-type resonance in the case of domed impacts. As can be judged by simple listening, a flat clap produces broad-band sound that typically extends to about 10 kHz while the spectrum of a domed clap usually has a subsidiary maximum somewhere below 1 kHz and then declines with frequency more rapidly than does the flat clap.

## INTRODUCTION

While the Zen koan of 'the sound of one hand clapping' aims to encourage meditation, the practical human two-hand-clap is an emotive communication gesture used in many gatherings such as concerts or lectures. A brief experimental study shows that the sound can vary from a dull thud through a low-frequency pulse to a sharp high-frequency snap, selection between these sounds being controlled by the shape of the hands on impact. A simple semi-quantitative examination of the impact dynamics shows that, for those configurations of the hands that produce a loud sharp sound, the generation of shock waves is involved. It is the purpose of the present brief paper to examine this process.

There is little in the way of previous studies to refer to, despite the ubiquity of clapping. A 1987 study by Repp [1] examined the sound spectra for different hand configurations, but concentrated on perceptual psychophysics rather than on physical acoustics. A later study by Hargather et al. [2] noted the presence of shock waves in some handclap sounds and examined them by schlieren photography, but again did not investigate the underlying physics.

## VARIETIES OF IMPACT

While the surface profile of the human hands is rather complex, a quick self-experiment shows that, in order to produce a loud sound as is generally desired, the two hands are oriented so that the more protuberant part of the profile — the ridge below the fingers or the fingers themselves — is brought into collision with the recessed palm of the other hand. Because the flesh of the hand is softly elastic, the collision is generally terminated by complete contact over the impact area, though it is possible to arch the hand so that some enclosed cavity remains. The flesh of the hands is also sufficiently damped by its cellular structure that acoustic vibration of the solid structures can be ignored, so that sound production is entirely due to the enclosed air.

Since the hands are not flat, there are several possible simplified geometries of this enclosed air, as shown in Figure 1.

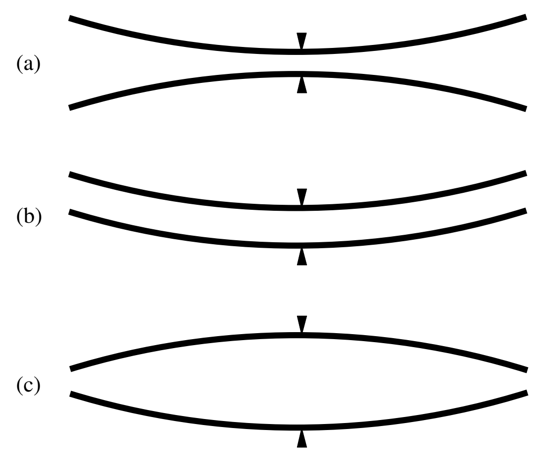


Figure 1. Three simplified geometries for impact of two hands during a clap. (a) very little sound produced, (b) a sharp clap sound with no resonance, (c) a sharp loud clap with a low-frequency resonance. There may be vents left at the edges.

If the hands are brought together where both surfaces are convex, as shown in Figure 1(a), then experiment shows that almost no sound is generated. If the relative positions and orientations of the hands are chosen so that the surface shapes are complementary, as in Figure 1(b), then the sound is a sharp high-frequency snap with no audible resonant component. If, however, the hands are cupped as in Figure 1(c), then there is a sharp sound with emphasized low-frequency components, the central frequency of which can be altered by adjusting the curvature of the hands, and thus the enclosed cavity volume, and also perhaps the geometry of the exit opening.

## DYNAMICS OF A FLAT IMPACT

While the hand surfaces are generally somewhat curved, as idealized in Figure 1, any matching curvature of the two hands, as in Figure 1(b), has very little acoustic effect. An initial simplified model can therefore consider the impact of two planar surfaces in order to approximate the actual sound from a sharp clap. This model can then be modified to incorporate an enclosed volume that does not collapse under the force of the impact. These two possibilities will be considered in order.

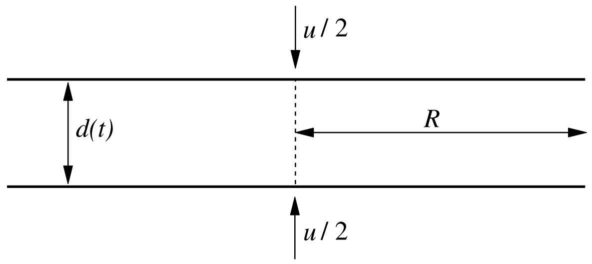


Figure 2. Simplified geometry for a planar-impact clap. Two circular planes of radius $R$ are approaching each other at speed $u$.

Consider the simple case of two rigid circular plates of radius $R$ moving towards each other along their common axis with velocity $u$ as shown in Figure 2. Ignoring compression for the moment, when the spacing between the plates is $d$ the air between them is expelled from the cylindrical ring around their edges at a speed $v$ where

$$v = \frac{\pi R^2 u}{2\pi R d} = \frac{Ru}{2d}. \tag{1}$$

If the initial spacing between the plates at time $t = 0$ is $d_0$, then this can be written

$$v = \frac{Ru}{2(d_0 - ut)} \tag{2}$$

which reaches the speed $c$ of sound in air after a time

$$t_0 = \frac{d_0}{u} - \frac{R}{2c}. \tag{3}$$

For an exit speed $v$ significantly less than the speed of sound $c$, the excess pressure $\Delta p$ between the plates can be described by the Bernoulli equation

$$\Delta p = \frac{\rho v^2}{2} = \frac{\rho R^2 u^2}{8d^2} = \frac{\rho R^2 u^2}{8(d_0 - ut)^2}. \tag{4}$$

For simplicity we assume this equation to apply up to the speed of sound. Using the assumed data in Table 1 gives a pressure of 120 Pa when the disc spacing is 1 mm, rising to 12 kPa for a spacing of 0.1 mm. Such a narrow uniform spacing is unrealistic, particularly for a handclap, and one might consider a configuration in which only a fraction $\beta$ of the vent perimeter remained open near the conclusion of the clap. This would raise the exit velocity by a factor $1/\beta$ and so raise the internal pressure by a factor $1/\beta^2$ which could easily be greater than 10 and increase with time. The pressure in the cavity between the plates thus rises increasingly rapidly with time, as also does the exit airspeed until it reaches the speed of sound.

When the exit speed $v$ reaches the speed of sound $c$, not all the compressed air escapes and the excess pressure $\Delta p$ between the discs increases more rapidly. The rate of decrease of the cavity volume is $\pi R^2 u$ and the rate of escape around the edges is $2\pi Rcd$. If $p_0$ is normal atmospheric pressure and $\gamma \approx 1.4$ the adiabatic constant for air, then the rate of pressure rise in the cavity is approximately

$$\frac{d\Delta p_2(t)}{dt} \approx \gamma p_0 \left( \frac{\pi R^2 u - 2\pi Rcd}{\pi R^2 d} \right) \tag{5}$$

where the subscript 2 has been used for this approximation. Using the result in equation (2) then gives

$$\frac{d\Delta p_2}{dt} \approx \gamma p_0 \left( \frac{u}{d_0 - ut} - \frac{2c}{R} \right). \tag{6}$$

Integrating this expression over the time range from $t_0$ to $t$ gives

$$\Delta p_2(t) \approx \gamma p_0 \log \left( \frac{d_0 - ut_0}{d_0 - ut} \right) - \frac{2\gamma p_0 c(t - t_0)}{R}. \tag{7}$$

This equation (7) has $\Delta p_2 = 0$ at the changeover point $t = t_0$, which is clearly not correct, but a more realistic solution is that

$$\begin{aligned} \Delta p(t > t_0) &= \Delta p_1(t_0) + \Delta p_2(t) \\ &\approx \Delta p_1(t_0) + \gamma p_0 \log \left( \frac{d_0 - ut_0}{d_0 - ut} \right) - \\ &\quad \frac{2\gamma p_0 c(t - t_0)}{R} \end{aligned} \tag{8}$$

for $t > t_0$. This gives a smooth transition between the subsonic and supersonic regimes at $t = t_0$. The prediction of the combined equations (4) and (8) is plotted in Figure 3 for the arbitrary but realistic set of parameter values shown in Table 1.

Table 1. Model parameters

| | | |
|---|---|---|
| Cavity radius | $R$ | 3 cm |
| Initial spacing | $d_0$ | 1 cm |
| Impact speed | $u$ | 1 m/s |
| Closing time | $d_0/u$ | 10 ms |
| Supersonic onset time | $t_0$ | 9.96 ms |
| Duration of shock wave | | 0.04 ms |

When this prediction is compared with a sound recording using real human hands, however, a new phenomenon appears, as shown in Figure 4. The smooth increase in pressure in the initial part of the graph agrees at least qualitatively with equation (4), and there is then a sudden change to a plateau of duration about 1 ms and then a broadband oscillatory signal with maximum frequency around 10 kHz and with a duration of about 5 ms. A possible explanation for this behavior is that the plateau is the duration of a simple shock wave exit and the subsequent oscillation is due to propagation of shock waves across the rather irregular geometry of the space between the hands. Depending upon the geometry of the closing hands, there are normally a few passages and apertures of about 5 mm height upon impact, and for a closing speed of about 1 m/s this implies an oscillation duration of about 5 ms, which is in approximate agreement with the measurements in Figure 4. From this figure the peak spacing in the waveform is about

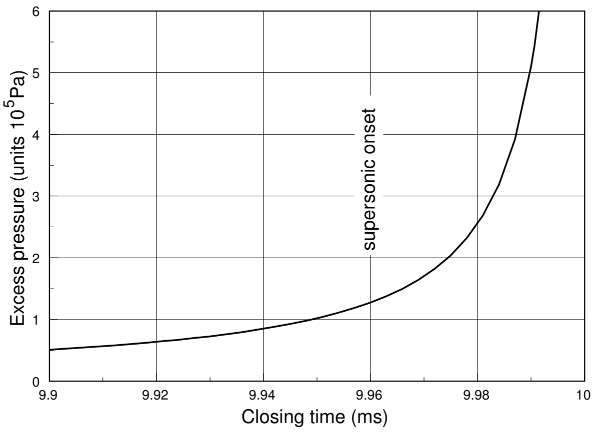


Figure 3. Increase in pressure between flat hands for the parameter values in Table 1. Normal atmospheric pressure is about $10^5$ Pa.

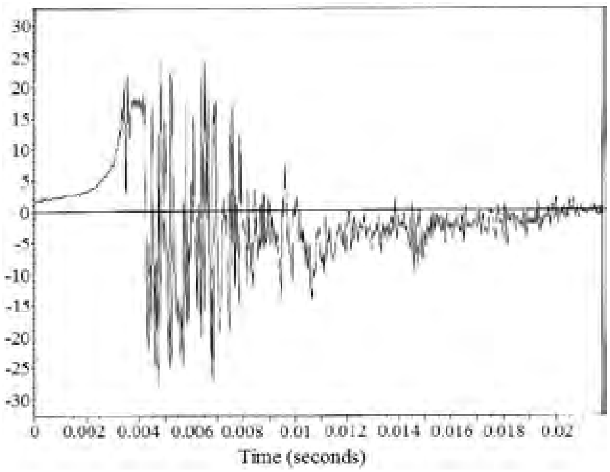


Figure 4. Waveform for a flat clap between two hands. The amplitude scale is arbitrary but the scan duration is 22 ms.

1 mm which corresponds to a time duration of about 0.2 ms. Since the irregular shallow cavity between the hands is typically about 6 cm in diameter, consideration of a circular wave approximation [3] suggests resonance frequencies of about 4, 10, 15, ... kHz, each of these being greatly broadened by the irregular shape of the cavity. Spectral aspects of the flat impact will be considered after discussion of the dynamics of a cupped impact in the next section.

## DYNAMICS OF A CUPPED IMPACT

When we consider the impact of two cupped hands, as in Figure 1(c), the geometry of the impact is of greater significance. Generally the two hands will make an initial impact that leaves an opening of moderate size in one place, so that shock waves are not generated at this stage. The geometry is then essentially that of a Helmholtz resonator with volume $V$ vented through an opening with cross section $S$ and effective wall thickness $l$. This has a resonance frequency

$$f = \frac{c}{2\pi}\left(\frac{S}{Vl}\right)^{1/2} \tag{9}$$

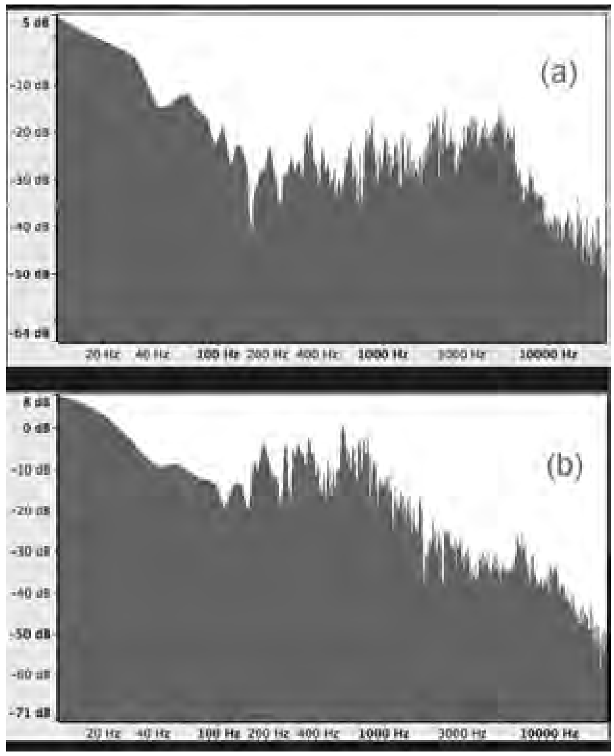


Figure 5. (a) Spectrum of the sound of a nearly-flat clap. (b) Spectrum of the sound of a cupped clap with the same hands. The sound pressure reference level is arbitrary and the frequency scale is logarithmic from 10 Hz to 20 kHz. Note that the level range in (b) is different from that in (a).

where $c$ is the speed of sound in air. As the hands move closer together and compress their contact areas, the height of both the enclosed volume and the opening decrease about linearly with time. The airflow through the aperture has a constant value, however, so that it ultimately reaches supersonic speed and generates a shock wave. If the aperture does not close, however, no such shock is generated and there is simply a radiated pulse at the resonance frequency of the vented cavity.

In a typical case the enclosed volume $V$ is about 100 cm$^3$, the total aperture cross-section $S$ about 1 cm$^2$ and the aperture length $l$ about 1 cm, giving a resonance frequency of about 500 Hz. The initial impact velocity may be higher than 1 ms$^{-1}$, but this slows once contact is made, allowing rather more than 0.01 s during which several oscillation cycles can occur before the final shock impulse. Because of the short duration of this resonant pulse, it is expected to produce a rather broad-band sound centered on this frequency.

## SPECTRAL ANALYSIS

Measurements of the sound of nominally flat and cupped natural handclaps were made in an anechoic environment. Because of the variability of hand profile and impact position there is large variability from one clap to another but one was nearly flat and the other had a deep cup. The measured spectra are shown in Figure 5. Both spectra show a decline of about 20 dB for a frequency increase of a factor 10, so that the sound amplitude varies about as $1/f$. It is clear that the main difference is that the nearly-flat hand-clap has a subsidiary intensity maximum in the range 1–10 kHz while the deep-cupped clap has a subsidiary maximum in the range 0.1–1 kHz. This is about what is expected from what is heard of the sound and confirms the general explanation presented above.

## CONCLUSIONS

The overall conclusion of the study is that shock waves produced by the rapid impact of two human hands appear to play a prominent part in the emitted sound. Because of the irregular shape and soft texture of the hands such shock waves appear to be generated to some extent in most cases, but their magnitude is much greater in the impact of a pair of nominally flat hands where the impact area is large rather than in cupped hands where there is a much smaller impact area. The extent to which shock waves are present in the sound of impacting cupped hands depends greatly upon their shape, which can be controlled by the person clapping. As is expected, a flat handclap has a large proportion of radiated sound in the frequency range 1–10 kHz while a cupped handclap has this maximum in the range 0.1–2 kHz, again under the control of the person clapping.

## ACKNOWLEDGEMENTS

This brief study of handclaps was provoked by my supervision of a University student project on Aboriginal clapsticks carried out by David Johnston. Handclaps are actually not significant in Aboriginal music or dance however, because the impacting objects are usually cylindrical clapsticks or more often boomerangs, the surfaces of which are convex as in Figure 1(a). The sound in these cases then results mainly from vibration of the two objects involved in the clap impact. I am also grateful to John Smith for help with the sound spectra and to Joe Wolfe for helpful comments and suggestions.

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Technical Note

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# NOISE FROM OTHER PEOPLES' HEADSETS

**Warwick Williams and Tom Harper**
**National Acoustic Laboratories, Sydney**

Noise leaking from under the headsets of personal stereo users is sometimes found annoying by those in the immediate vicinity and can be the subject of complaints. Measurements presented here demonstrate that the associated noise levels should be minimal in nature and not pose a significant source of difficulty for most people.

## INTRODUCTION

As acousticians at some stage you will have no doubt been indignantly informed that many users of personal stereo players (PSPs) have the levels of their headphones set at extremely loud values. We are frequently told that this noise is not only damaging the hearing of the user of the PSP but also annoying others by disturbing their acoustic amenity. The National Acoustic Laboratories (NAL) recently had the opportunity to measure such 'leakage' noise from sixteen circumaural and supra-aural headsets.

## METHOD AND RESULTS

Eight circumaural and eight supra-aural devices were assessed to provide an indication of typical levels that can be expected to be heard by adjacent listeners (i.e. leakage) and the insertion loss provided by the devices.

Circumaural devices sit on the side of the head and encompass the pinna as in the case of an ear-muff. Supra-aural devices sit on the surface of the pinna and thus provide less attenuation of background noise when compared to circumaural devices while allowing more of the signal to escape. Insertion loss and leakage measurements were taken using an acoustic test fixture (ATF) specified in ISO 4869-3: 2007 [1] (Figure 1). This insertion loss measurement is similar to the $SLC_{80}$ but not equivalent.

To test device sound leakage the equivalent continuous A-weighted free-field sound pressure level output of the earphones was set to an LAeq of around 90 dB in accordance with the procedures outlined in AS/NZS 1269.1:2005 [2]. The actual signal input level for an output of 90 dB had previously been determined using a GRAS Ear and Cheek Simulator Type 43AG. The sound 'leaking' from under the devices was measured using a B&K C-Frame Portable Pulse 3560 spectrum analyser at a microphone distance of 300 mm. All noise sources used were pink noise and systems were calibrated appropriately.

Both leakage and insertion loss results are presented in Table 1. The ATF essentially acts as an 'artificial head' and for this exercise it was fitted with an artificial pinna to accommodate supra-aural devices.

Figure 1. Photo of the acoustic test fixture as per ISO 4869-3:2007 [1], showing mounting of the circumaural headset under test

Table 1. Summary of test results for insertion loss, and leakage at 300 mm referenced to an equivalent at ear $L_{Aeq}$ of 90 dB free field

| Device code | Type | Leakage @ 300 mm ($L_{Aeq}$ ref = 90 dB) | Insertion loss (dB) |
|---|---|---|---|
| 01BEA | Supra-aural | 49 | 9.0 |
| 02BOS | Supra-aural | 50 | 8.0 |
| 03BOW | Supra-aural | 52 | 1.7 |
| 04FAN | Supra-aural | 45 | 6.6 |
| 05GRA | Supra-aural | 65 | 2.0 |
| 06HAR | Circumaural | 31 | 12.0 |
| 07MON | Circumaural | 41 | 9.1 |
| 08PHI | Supra-aural | 43 | 9.5 |
| 09PIO | Supra-aural | 39 | 8.5 |
| 10SEN | Circumaural | 66 | 3.2 |
| 11SKU | Circumaural | 42 | 6.6 |
| 12SMS | Circumaural | 36 | 5.3 |
| 13SOL | Supra-aural | 48 | 4.9 |
| 14SON | Circumaural | 41 | 10.0 |
| 15ULT | Circumaural | 41 | 14.8 |
| XXSON | Circumaural | 41 | 10.1 |

## DISCUSSION

A summary of the test result statistics is presented in Table 2. As can be seen from the summary statistics the noise levels escaping from the above devices are not particularly significant in terms of hearing health. This does not mean that these levels may not cause some annoyance to adjacent individuals. In particular those levels at the higher end of the range, around 65 dB, may be noticeable in a very quiet environment. Most levels would be readily masked when travelling on public transport.

The reference value of 90 dB was selected as a worst case scenario and to ensure that equivalent measurements could be comparable for all devices. Previous studies show that typical $L_{Aeq}$ values average around the 80 to 85 dB level, much less than the 90 dB selected [3,4]. It should also be noted that the measured insertion loss for the devices does not govern the attenuation of the leaking noise. The attenuation of the devices is comparatively low as would be expected as they are not intended to act as a conventional ear-muff. Typically circumaural devices provide greater attenuation but this is not generalisable as some circumaural devices are more acoustically transparent than others when that may be designed to permit the wearer to monitor their acoustic environmental.

The limitations of this study are that music tracks themselves were not used as the sources of the noise and that some individuals may, and do, select much higher noise levels when listening to their favourite music. However the results measured should give a reasonable order of magnitude of the levels to expect when PSPs are in use.

Unfortunately the testing only included circumaural and supra-aural devices but no in-ear devices, such as ear-buds or inserts. However even these limited results are interesting as they provide an indication of the maximum noise levels that can be expected to be heard by adjacent listeners particularly those travelling on public transport.

Table 2. Summary of device test descriptive statistics for insertion loss, and leakage at 300 mm referenced to an equivalent at ear $L_{Aeq}$ of 90 dB free field

| Device type | Average leakage @ 300 mm $L_{Aeq}$ (rel) 90 dB at source (range) | Insertion loss dB (range) |
|---|---|---|
| All devices (16 devices) | 45.5 (σ = 9.3) (65.7 - 31.3) | 7.6 (σ = 3.6) (14.8 - 1.7) |
| Supra-aural devices (8 devices) | 48.7 (σ = 7.6) (64.6 - 39.1) | 6.3 (σ = 3.1) (9.0 - 1.7) |
| Circumaural devices (8 devices) | 42.3 (σ = 10.1) (65.7 - 31.3) | 8.9 (σ = 3.7) (14.8 - 3.2) |

## CONCLUSION

The leakage of sound from headset use associated with personal stereo players should have minimal impact on those in close proximity to the user. This cannot of course guarantee that some people will not be affected and chose to complain about the use of personal stereos and the noise they may produce.

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# ACOUSTIC REDESIGN OF BRISBANE CITY HALL AUDITORIUM

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Prompted by a need for urgent structural repairs, a $215 million project was initiated by Brisbane City Council in 2009 to restore Australia's largest town hall. AECOM was commissioned through the architects to provide acoustic advice for the overall building, including Council Chambers, staff offices, and a new rooftop gallery for the Museum of Brisbane. AECOM was also engaged to provide specialist architectural acoustic design for restoration of the main Auditorium.

## A BRIEF ACOUSTIC HISTORY

The size and geometry of the Auditorium at Brisbane City Hall creates an imposing space and distinctive ambience of grandeur. However, the large size of the space, its circular form, and geometry of the domed ceiling all contributed to acoustic issues that have affected events and activities taking place in the Auditorium since its original opening 83 years ago.

Previous refurbishments of the Auditorium had attempted to address some acoustic deficiencies, primarily through introduction of acoustic absorption. In the 1970s the solid dome ceiling was replaced with expanded vermiculite, applied to chicken-wire on a timber frame. In the 1980's large fabric-faced wall and ceiling absorber panels were applied liberally throughout the auditorium. While such treatments were clearly well-intentioned modifications to control the issues of focusing and poor intelligibility, these treatments had not addressed the underlying room geometry, and as a result never truly tamed the problems of focused sound.

Conditions for a variety of contemporary uses – including banquet dining, speaking events, trade shows and amplified music concerts – were still compromised. Indeed, with absorptive treatments only, the acoustic character of the auditorium had been substantially altered and the reverberation time reduced to such an extent that the City Hall was unsatisfactory for orchestral or chamber music – and wholly unsuited to performances on the historic Henry Willis & Sons pipe organ. To support the ongoing viability of the auditorium both as a commercial venue, and as a community space, the renovation plans necessarily had to address these acoustic issues.

## FEATURES OF THE ACOUSTIC REDESIGN

The old vermiculite dome facing has gone, replaced with transondent membrane which replicates the dome shape visually (with subtle adjustment to the geometry – as shown in Figure 1), while concealing acoustical reflector arrays and allowing the architects and specialist lighting designers to provide theatre systems and integrated lighting displays. This system incorporates two layers of lightweight and micro-perforated stretched membranes. A concealed ceiling reflector array was then designed to meet the exacting structural constraints of the historical building structure. Even very small increases in weight, multiplied across dozens of repeating elements would affect the ability of the building structure to support temporary event rigging systems.

The outer dome was restored and treated with a sound-deadening composite foam lining, incorporating a fire-resistant facing and an embedded limp-mass layer. This treatment provided the necessary balance of sound insulation and absorption whilst being relatively lightweight.

Figure 1. The newly installed dome, incorporating heritage central lantern

Another critical design feature was the introduction of new acoustic diffusers to replace the existing wall panels, as shown in Figures 2 and 3. These were developed specifically to combat acoustic deficiencies in the base auditorium geometry. The diffuser panels are moulded in glass-reinforced plaster, recessed up to 600 mm into the auditorium walls to minimise visual impact. The design also allowed for modular construction, aiding manufacture, delivery and installation on site.

Figure 2. Installed acoustic diffuser panels and displacement air grilles, with architectural facings

Figure 3. Acoustic diffuser panel with integral variable absorption, architectural facing and heritage light fitting

Variable acoustic control has been incorporated into the space through automated acoustic banners to provide subtle control over reverberant conditions in the space, allowing conditions to be matched to a variety of uses from meetings and exhibitions to organ recitals. The banners and diffusor panels have been concealed with architectural facings to integrate with heritage details.

Figure 4. Prototype panel testing at RMIT

The panel designs were extensively tested prior to manufacture via 3D acoustic ray tracing. Prior to installation full-scale prototypes were constructed and tested in the reverberation chamber at RMIT in Melbourne to verify absorptive properties, as shown in Figure 4.

Additional measurements of the directional diffusion coefficient were conducted at full-scale, in a temporary testing facility established specifically for the tests at Jands' factory in Sydney. This testing applied the newly published standard for testing of directional diffusion coefficients [1].

## RESTORATION RESULT

Figure 5 shows the acoustic result for the auditorium is an improved reverberation time – extended by over one second – much more consistent with the room's original grandeur, and enabling the Henry Willis organ to be featured. The auditorium also enjoys variable acoustics for fine-tuning of the space according to the type of event being held.

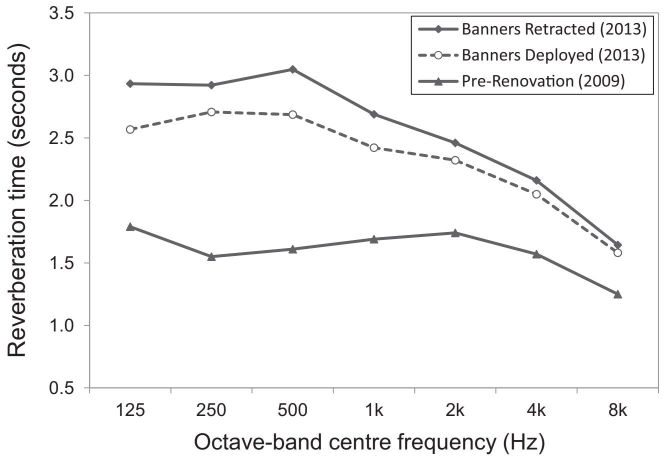


Figure 5. Reverberation times (unoccupied) pre- and post- renovation works

## RESTORATION FACTS

New building elements critical to creating the new acoustic conditions in Brisbane City Hall's main auditorium include:

- Dome array consisting of 70 acoustic reflector panels, concealed above the visible dome face
- New visible surface for the dome made from a transondent tensile membrane
- Modified geometry and radius for the visible face of the ceiling dome
- Restored external copper dome has had a multi-layer limp-mass lining added to provide absorption within the dome and boost the transmission-loss
- 16 custom profiled 2D acoustic diffuser panels, moulded from glass reinforced plaster and recessed into the auditorium walls at the stalls level
- 18 custom profiled 3D acoustic diffuser panels, moulded from glass reinforced plaster and recessed into the auditorium walls above the balcony
- Variable acoustics provided by 46 operable acoustic banners integrated with new architectural elements
- Programmed control system providing automated setting of acoustic banners for different events
- New low-noise displacement air-conditioning system
- New seating throughout
- Original hardwood flooring reinstated
- Original plaster surfaces and detailing restored.

## ACKNOWLEDGEMENTS

AECOM acknowledges the work of Brisbane City Council, Tanner Kibble Denton Architects (in particular Megan Jones and Scott MacArthur), and Jands (Peter Grisard and Chris Clegg), contributing to the acoustic success of this significant public project.

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# MUSICAL RHYTHM, VIBRATO AND WIND TURBINE NOISE

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The nature and origin of frequency and amplitude modulation in music and other related human activities is examined and particular neural sensitivity is identified in the frequency range 1 to 10 Hz. It is surmised that amplitude modulation as the vanes pass the tower may be a major factor in wind turbine annoyance, and suggested that an electro-acoustic system to reduce this modulation might be effective in reducing noise annoyance.

## INTRODUCTION

Electrical power is of increasing importance to the survival of human civilisation, and many of the energy sources used to produce it, such as coal, gas, oil, and even nuclear energy, have limited future availability. Everything therefore comes down to solar power in some form, particularly solar-electric, tidal, and wind power. Of these, wind power alone presents acoustic problems because of the noise produced by the rotating turbine blades. This noise is not very loud but it has led to problems for communities living several kilometers away, though the basis of this noise annoyance is far from clear.

The present short paper examines some aspects of wind turbine noise and relates them to particular acoustic and psychoacoustic features, some of which might perhaps be remediable. Some of these things have doubtless been discussed before in the published literature, but here I adopt the approach of famous philosopher Ludwig Wittgenstein, who wrote in his *Tractatus Logico Philosophicus* "I quote no sources. It is a matter of indifference to me whether what I have thought has been thought by others before me."

## RHYTHM

The notion of rhythm probably derives from the periodic nature of leg motion during walking, which has a repetition rate of about 1 Hz. This is also about the beat rate of the human heart and is a feature of sound that is readily perceived. From it comes the music of a march, which links together the leg motions of all those involved to create a communal feeling. While simple marches use just a percussion instrument such as a drum to control synchronism, musical marches also evolved more than a thousand years ago and are now commonly used in military ceremonies.

This is, of course, not the only human rhythm of importance, for we also have many dance rhythms that are widely used, again with a repetition frequency of about 1 Hz. While a march has a simple left-right-left-right... repetition, dances can have more complex structures. A waltz, for example, has a three-fold rhythm with three beats to a bar, though the actual motion is six-fold because of transfer between the two feet. Again, music is common for leading dances and the interaction between dancers can be either communal or two-fold. Musical dance compositions sometimes create an additional striking effect by a structure called a hemiola, in which the bar-length remains constant but instead of being divided into three beats it is divided into two longer beats for just one or two bars.

No more discussion of rhythm is needed here, the main point being that human perception of rhythm is very acute near a frequency of 1 Hz.

## VIBRATO

While music may be thought of as made up from combinations of simple tones, either sequentially to form a melody or simultaneously to form a harmony, or both, there are some psychophysical subtleties that turn out to be very important in the present context. Some instruments, such as the pipe organ, produce notes that are steady in sound after an initial brief transient, while in the piano the notes simply decay slowly after initial production. This is different, however, from the sound of a violin or cello, where the player purposely rocks the finger defining the string length backwards and forwards to produce a periodic small variation in pitch. The finger motion is typically less than about 1 cm for a string of length about 30 cm, so that the total pitch variation is about 1/30 which is about half a semitone and the repetition rate of this vibrato is about 5 Hz. For a solo instrument, this adds interest or "warmth" to the sound, while for a large group of string instruments playing together as in an orchestra, the sound becomes narrow-band noise because the variations are not synchronised. This gives a modern string orchestra a rather different sound from that of an orchestra from baroque times, where the string instruments had frets to fix the string lengths and there was no vibrato.

Wind instruments operate in a very different manner from strings, but the player has some control over pitch, loudness and timbre (spectral envelope) by varying blowing pressure, mouth shape, or other parameters. Here the performance is, however, ruled to some extent by tradition, so that some instruments are played without any vibrato while others us it constantly. The pitch variation is generally quite small – only about a tenth of a semitone – but there can be a considerable periodic variation in both loudness and timbre. As with other instruments, the rate of this variation is typically about 5 Hz.

When we come to consider human singers, however, things become much more extreme. While a few singers, such as choir boys, sing pure notes without any vibrato, most adult singers do use vibrato, some to extreme. Operatic sopranos, in particular, can have very wide frequency spread in their vocalisation, typically as much as, or even more than $\pm 1$ semitone. The amplitude and intensity of the vibrato is increased purposely in dramatic or emotional scenes in the opera.

A persuasive explanation of the role and importance of vibrato can be seen in the behaviour of neural rhythms in the human brain. In a normal brain in a relaxed state, neurons tend to oscillate in a state called the "alpha rhythm" which has a frequency in the range 8–12 Hz. This is surprisingly close to the typical 6 Hz frequency of vibrato in both instrumental playing and in singing, particularly emotional singing. It is therefore reasonable to surmise that there is a close relation between vibrato and the alpha rhythm of the human brain. Taking things to an extreme state, it has been shown that visual stimulus from flashing lights can actually cause epilepsy in susceptible individuals.

## WINDFARM NOISE

So how does all this relate to the effect of wind turbine noise on listeners who are not very close, so that the sound is barely audible? A typical generator in a wind turbine has a rotation speed of about 0.3 Hz and, since it normally has three vanes on the rotor, this means that the frequency at which vanes pass the support tower is about 1 Hz. Most of the sound of the generator is produced by air flow over the rotating vane, so that this will be modulated in amplitude at a frequency of about 1 Hz. While this is a bit low to interact strongly with the alpha rhythm of the brain, it will have higher frequency components because the space between the vanes is large compared with their width. Even without such interaction, an amplitude modulation frequency of order 1 Hz would be readily perceivable and could cause much more annoyance to a listening human than would a steady sound of the same amplitude. The fact that 1 Hz is about the frequency of the sound involved in other human activities, such as marching or dancing, also leads to the surmise that such a modulation frequency might be more readily noticed by the listener than other much lower or much higher frequencies.

So where does this lead us? Obviously a simple reduction in the wind turbine noise level would reduce its impact upon not-too-distant listeners, but such a further improvement would be difficult to make. Perhaps, however, the solution is not to reduce the total sound level but to reduce the level of the amplitude modulation caused by vanes crossing the supporting tower. It is to be hoped that an aero-mechanical solution to this problem might be found, but another possibility would be to sample the radiated sound field at a moderate distance from the tower and then to provide an electrical signal that could drive a loudspeaker system mounted on the tower and with an appropriate directional radiation pattern so as to reduce the level of the amplitude modulation.

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# CONCERT QUALITY SOUND FOR ANZ STADIUM

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**The P.A. People Pty Ltd, Rhodes NSW 2138**

A $3mil sound system upgrade to Sydney's ANZ Stadium gives concert-sound quality to the 83,500-capacity venue, setting a new standard for stadium sound in Australia and matching the best in the world. The multipurpose stadium was purpose built for the 2000 Summer Olympics and now hosts professional sporting events as well as being the first choice for global entertainment acts including U2, The Rolling Stones, Andre Rieu and AC/DC. The new sound system doubles the number of speaker boxes to 374 suspended in clusters. The system was first used at the State of Origin opener on 5th June 2013.

The system was installed by Sydney based contractors The P.A. People, who have a long-standing relationship with the venue. The company designed and installed the original sound system in 1999 for the 2000 Olympics, undertook the system's conversion in 2003 when the venue was renovated, and has maintained the system with a comprehensive service regime and provided PA system operators for every major event since.

When the original system reached the end of its service life, the company was contracted to install the new d&b V system speakers along with the substantial mechanical design and metal fabrication required. The system is based around the Stadium geometry with loudspeaker clusters suspended up to 45 metres high in the four quadrants, each served by its own amplifier room. Each of the quadrants was installed in a single week through May to enable the entire PA to be fully operational for weekend club matches. The four-way stereo design comprises 266 full-range line-array loudspeaker elements, 44 full-range loudspeakers and 64 sub-bass cabinets. The system was designed by Scott Willsallen of Auditoria, with d&b components supplied by National Audio Systems, and new hoists by Jands Theatre Projects.

One of the main design tasks for the new sound system at ANZ Stadium was the conception and design of the flying frames and brackets for the d&b loudspeaker hardware. While d&b provided 3D models of each of the speaker cluster elements, it fell to The P.A. People's engineering design and fabrication team to craft each of the clusters in such a way that they could be supported from just two points each under the catwalks at the Stadium.

The design process was complicated due to the requirement to rotate the main axis of each of the components on the majority of clusters to achieve a radial distribution pattern. Precise geometry and calculation of each of the speaker cluster and frame elements was required in order to maintain the balance of the cluster, with some clusters weighing almost a tonne and measuring almost seven metres in length.

In a site the size of ANZ Stadium everything is a long way apart, both horizontally and vertically. The P.A. People used a pair of temporary staging decks under each speaker location to facilitate the removal of the old speakers and reinstallation of each of the new clusters. Once the clusters arrived on site they were driven to the appropriate location and manhandled onto the platforms. From there, the speaker chain hoists were used to assist in assembling the clusters before they were lifted to their final resting place, shown in Figure 1.

The system was installed over a four-week period in May this year. During each weekly installation period, two teams installed the system in one quadrant with one team handling the loudspeaker and cluster installation, while the second team stripped and fitted out the amplifier room to support the new system. On each Friday night during the installation period, the entire system (comprising both new and old clusters) was used to support a local NRL match - just another challenge for the installation team.

Figure 1. ANZ Stadium sound system

# NEWS

### The next ICA president is...

Marion Burgess! A milestone is Australian acoustics history has been reached. At the General Assembly of the International Commission for Acoustics (ICA) held in June, Marion Burgess was elected as President and Jing Tian from China as the Vice President. This will be effective from 1st October 2013. This is the first time the President has been from Asia and also the first time a female has been elected President. Marion Burgess is a Research Officer at UNSW, Canberra. Marion is an editor of the Acoustics Australia journal and a Fellow of the Australian Acoustical Society.

### Release of the Rail Infrastructure Noise Guideline

A new guideline to help manage noise and vibration from rail infrastructure projects has been released by the NSW Environment Protection Authority (EPA). The EPA's *Rail Infrastructure Noise Guideline* (RING) aims to control noise and vibration levels from new heavy and light rail lines and rail traffic-generating developments like new coal mine and non-network rail lines.

The new guideline reflects advice received from transport, planning, infrastructure construction, rail agencies, acousticians and the broader community during the consultation period on the draft guideline in early 2012.

Key improvements are:

- Noise 'trigger levels' set for light rail systems and non-network rail lines increasingly being built by industry.
- Rail projects must seek to reduce rail traffic noise levels in a rail corridor down towards the "trigger levels" not just reduce the additional noise associated with the rail project.
- The guideline clarifies whether a land developer or rail track owner is responsible for noise mitigation when residential development encroaches on rail lines.
- There are no exemptions allowed for 'minor works'.

The *Rail infrastructure noise guideline* replaces the *Interim guideline for the assessment of noise from rail infrastructure projects* published in 2007.

Copies of the guideline can be downloaded from the EPA's website at http://www.environment.nsw.gov.au/noise/railnoise.htm

### NSW Noise Guide

The NSW Environment Protection Authority (EPA) has recently updated the "Noise Guide for Local Government". This is an important resource designed to assist agencies to deal efficiently and effectively with noise issues.

The entire Guide has recently been updated to align it with changes to the regulatory provisions, clarify the existing material or provide additional guidance. In particular, the areas related to vessels, aircraft and planning have been amended. Additional guidance has been provided in relation to dealing with barking dogs and the management of large open air music events. A number of technical areas have been clarified and references to government agencies responsible for managing noise have been updated.

The updated Guide may be accessed at: www.epa.nsw.gov.au/noise/nglg.htm

### Cheers for Ears

Cheers for Ears (CfE) is a school health program designed to educate and encourage healthy behaviours amongst young people to prevent noise induced hearing loss. The program is currently available for children in years 5-7 (10-12 year olds) at schools throughout Perth. Since 2010 more than 22,000 children from in excess of 150 schools throughout the Perth metropolitan area have participated. The program was the 2011 winner of the AMA(WA)/Healthway Healthier WA Award. The program is endorsed and supported by the Ear Science Institute Australia management team and Board. There are fact sheets and other information available from the website to supplement the program.

A recent initiative of the program is a mural aimed to visually represent hearing called a Living Wall. This aims also to present a new way to encourage care for hearing. The concept and background to the implementation of the Living Wall can be seen from http://youtu.be/eLBIJxVhlEQ

### Promotion of awareness of hearing loss

A worldwide competition for ideas to promote awareness of hearing loss and prevention was run by the Danish organisation Ida Institute. The 3 winners and 10 finalists can be found in the Big Ideas section at http://idainstitute.com/ideas_worth_hearing/

### Marshall Day Acoustics now in Perth

Marshall Day Acoustics has recently opened a new office in Perth, Western Australia. The opening of the Perth office brings the total number of Marshall Day locations to 12, with existing Australian offices in Melbourne, Sydney and Adelaide; four offices in New Zealand; and international offices in Shanghai, Hong Kong, London, and Ireland. The WA State Manager is Ben Wilson. Ben has returned to Marshall Day having spent the last few years working in the UK. Joining Ben in the Perth office is Peter Jago who will represent Marshall Day's successful theatre consulting arm, Entertech.

Contact details for the new office are Suite 1, 186 Hay Street, Subiaco WA 6008. Tel. (08) 9779 9700 perth@marshallday.com

### 2nd Birthday for Resonate Acoustics

Resonate Acoustics celebrates 2 years of operations in August 2013. We have had amazing support and now have offices in Adelaide, Melbourne, Sydney, as well as a new office in Brisbane (opened July 2013). To celebrate we have also recently launched a new website resonateacoustics.com. For further information please contact Matthew Stead at matthew.stead@resonateacoustics.com

### Regupol new national sales manager

Nishi Kant Grover has been appointed as National Sales Manager (Acoustics & Vibration) Regupol (Australia) Pty. Ltd. He recently returned from initial training with BSW in Germany and follow on meetings. Acoustic Engineers are invited to register on the new Regupol website, www.regupol-vibration-technology.com.au to take advantage of the latest case studies and receive product updates throughout the year. Enquires are welcome on (02) 4624 0050 or vibration@regupol.com.au

### Quiet Acoustics Award

Quiet Acoustics have taken out the Industries Choice Award at the Chamber of Minerals and Energy 2013 Safety & Health Innovation Conference, for their noise cancelling industrial panel.

# STANDARDS AUSTRALIA

### Human vibration

In 2012 the Australian Acoustical Society submission to Standards Australia for a direct text adoption of the two ISO standards on hand arm vibration was successful. In early 2013 the committee dealing with *Vibration and Shock Human Effects* was reformed with Colin Tickell as the Chair. Even as a direct text adoption there was work to ensure that the Standards Australia versions provided sufficient guidance so that they would be consistent with other relevant Australian Standards. As part of the same project, there was also work to incorporate the recent ISO amendments in an updated version of the Standards Australia version of the ISO whole body vibration standard. The outcome is that three standards have now been published which bring the Australian methods for measurement and evaluation of human vibration in line with International methods:

- AS ISO 5349-1-2013 Mechanical vibration - Measurement and evaluation of human exposure to hand-transmitted vibration-Part 1: General requirements

- AS ISO 5349-2-2013 Mechanical vibration - Measurement and evaluation of human exposure to hand-transmitted vibration-Part 2: Practical guidance for measurement at the workplace
- AS 2670.1-2001 Amendment No. 1 to Evaluation of human exposure to whole-body vibration-Part 1: General requirements

Safe Work Australia (www.safeworkaustralia.gov.au) has developed fact sheets relating to hand arm and whole body vibration and is currently working on codes of practice which will now be able to reference the relevant Australian Standards.

Another action from this reformed committee is the agreement to withdraw AS 2670.2 1990 Evaluation of human exposure to whole-body vibration Part 2: Continuous and shock-induced vibration in buildings (1 to 80 Hz), in December 2013. This is an out-dated standard and it is intended to replace this with ISO 2631-2, which is itself proposed for revision by the ISO Technical Committee TC108-SC4 from September 2013

### Draft revision of AS/NZS 1269.4

A draft revision of AS/NZS 1269.4 - Auditory Assessment is expected to be released for Public Comment in August 2013. Standards Australia is particularly interested in comments from any acousticians with practical experience in measuring and assessing background noise levels in audiometric testing booths and spaces, as changes to this section of the standard are proposed. See http://www.standards.org.au/Pages/default.aspx

## New Products

### CadnaR room design software

Datakustik are pleased to announce the release of version 2.1 of CadnaR together with a new software option “Audio”. CadnaR offers a complete and integrated solution for the investigation of noise and sound inside rooms. New calculation and configuration features of CadnaR are

- frequency range extended to 9 octaves (31.5 Hz to 8000 Hz with adjustable range);
- calculation of the room impulse response at receiver points;
- several room acoustical parameters selectable;
- calculation up to the 500-th order;
- emission of particles per octave band (with energy-dependent number of particles);
- ability to include the directivity of point sources by the number of particles and time-dependent course of the particle emission displayed in 3D.

The CadnaR 2.1 Demo Version is available for download from the Datakustik website www.datakustik.com. Contact Rodney Phillips at Renzo Tonin & Associates for detailed information on (02) 8218 0500 or email rphillips@renzotonin.com.au

### Side Vented Condenser Microphone

PCB Piezotronics, Inc. announce the release of new Model 377A14 Side Vented 1/4" Condenser Microphone. The 377A14 microphone is designed for use in very high frequencies (100 kHz) and high amplitudes (174 decibels) applications. It features a side vented design providing a greater degree of accuracy when the microphone is flush mounted in a wall or tube where there are significant pressure differentials between the inside and outside of the tube or wall. Typical applications include testing of panels, airplane wing testing, HVAC, sound absorption in impedance tube tests, or general high amplitude or high frequency sound or noise measurements in cavities or small enclosures.

Model 378A14 is a Pressure Field microphone and preamplifier combination which combines the 377A14 microphone with a matching 426A05 preamplifier to help minimize leakage of external atmospheric pressure. This is available with TEDS (Transducer Electronic Data Sheets) programming for traceability of model, serial number, sensor location, sensitivity, and additional calibration information. These models are IEC 61094 compliant and are based on ICP® sensor technology.

Further information from Suheil Khandwalla, Thermo Fisher Scientific, (03) 9757 4429, suheil.khandwalla@thermofisher.com

### Radar gun for SoundBook

A radar gun can be used with the SoundBook, SAMURAI software or MATLAB to measure the speed, noise and/or vibration of passing trains or road traffic. Utilise the NoiseCam Option in SAMURAI and a web camera to capture a visual record and the speed of the vehicle and display the results in real time. Measurement data, including speed, may be exported and visually presented in Excel and other formats. The radar gun is designed to not stand out visually so as not to attract the attention of passing vehicles. Further information from Darryl Watkins at SAVTek, (07) 3300 0363 or dwatkins@savtek.com.au, www.savtek.com.au

## Divisional News

### NSW Division

On 26th June, Paul Maddock, a Senior Noise Officer at the NSW Environment Protection Authority, gave an overview of the EPA’s *Rail Infrastructure Noise Guideline* (RING), which was published in May. RING replaces the *Interim guideline for the assessment of noise from rail infrastructure projects* (IGANRIP, 2007). RING aims to streamline the approval processes for rail infrastructure projects while ensuring that potential noise and vibration impacts are assessed in a consistent way and minimised as far as possible. The Guideline reflects advice received from transport, planning, infrastructure construction, rail agencies, acousticians and the broader community during the consultation period on the draft guideline in early 2012.

On 16th May, Kim Henley, a specialist Technical Services Engineer with Orica Mining Services, gave a talk to the NSW Division on *Blasting in the urban environment*. Kim discussed that drilling and blasting for civil construction is a process that is sometimes unnecessarily excluded from consideration in tenders and project specifications. He showed that modern blasting technology has made breaking rock with explosives safer, more secure, and more productive than ever before. Kim’s talk also covered some of the practicalities of blasting in built up areas. He discussed the management of the risk associated with this type of blasting, in particular vibration, airblast and flyrock, and the environmental monitoring regime that is used to measure compliance.

Three years ago, Sydney was the host to the most prestigious acoustics event – the 20th International Congress on Acoustics (ICA 2010). In June 2013, the 21st International Congress on Acoustics was held in Montréal, Canada, and was attended by over 2,000 participants. The NSW Division allocated grants to assist with the travel costs for a number of young researchers to participate in this event.

### VIC Division

On the 17th June, the Victorian Division hosted a technical meeting at SKM’s offices in Melbourne in which measurement technologies were discussed. The invited speakers were Kung Nhim [Brüel & Kjær] and Miroslav Dosen [ETMC Technologies]. Kung presented first about remote noise monitoring with specific regard to B&K’s Noise Sentinel solution. Kung highlighted that the traditional approach of visiting sites to conduct investigations or to deploy long term noise monitors has its limitations. These challenges have been overcome in recent years with developments in remote monitoring systems. Such systems have truly come of age with the advent of 4G networks allowing large data volumes to be quickly uploaded. B&K’s system was visually splendid and seemed easy to use with the application delivered entirely over the internet. By appropriately configuring the system alerts, warnings can be generated when noise levels are excessive allowing for an almost immediate response. Additional functions such as air quality monitoring, NTi AUDIO

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audio capture for source identification and directional noise monitoring can also be incorporated into the solution.

Miroslav followed with a detailed overview of data acquisition systems and acoustic measurement methods. Miroslav highlighted the importance of understanding the full system rather than just the performance of individual components such as the microphone or preamplifier. He provided gems of wisdom including the recommendation to visually inspect the microphone diaphragm periodically as subtle damage or dust can modify its frequency response leading to potentially invalid results. Miroslav concluded with a video demonstration of Norsonic's acoustic camera applied to a train passby. The systems microphone array and video camera are contained within a circular carbon fibre disk equipped with two handles allowing for convenient and rapid set-up.

### WA Division

The WA division held two technical meetings in the month of July. Firstly, on the 3rd of July, a meeting was held to discuss the *State Planning Policy 5.4 Road and Rail Transport Noise and Freight Considerations in Land Use Planning*, which was attended by 18 members. This policy was gazetted in 2009 so the meeting was an opportunity for AAS members to discuss their experiences over the last 4 years and provide comments and recommendations for future revisions to the policy. John Macpherson, Principal Environmental Noise Officer with the Department of Environment Regulation, opened the discussion with a brief overview of the development of SPP5.4 and some experiences with its operation. This was followed by open discussion based on a series of questions prepared by Terry George of Lloyd George Acoustics. Following the meeting, Luke Zoontjens (Western Australian Division Chair) formulated a letter on behalf of the members to be sent to the Department of Planning.

The second technical meeting was held on 25th July at the Perth Arena. The Perth Arena is Western Australia's landmark home of live entertainment, music and sports, and is the first venue in WA to incorporate a retractable roof. The meeting began with an introduction from architects CCN, then members were given a tour through the main bowl, break out areas, team rooms and green rooms, and backstage areas. Of particular interest to members is the acoustically treated roof construction to reduce noise break-out when the roof is closed.

### QLD Division

On the 16th April, Queensland Division members visited the Kilargo factory and acoustic test facility at Banyo. Kilargo acoustic engineer Christophe Titry outlined the design features of their test facility which is used to conduct sound reduction testing of their door seal products. The test procedures and equipment used to conduct the testing was demonstrated for the benefit of those who attended. The manufacturing processes used to make different types of acoustic seals were also of immense interest.

On the 11th June Tom Hardy (Business Manager, Hufcor Sound Management) provided background information and a live demonstration of the sound conditioning system available from Hufcor. A number of different spectral compositions were played through the system to give members the opportunity to subjectively evaluate the intrusiveness of the sound. The demonstration provoked some lively discussion amongst the members regarding the most appropriate way to implement a sound conditioning system in practice.

## Future Conferences

### Inter-Noise 2013

Inter-Noise 2013 is the 42nd International Congress and Exposition on Noise Control Engineering, and will be held in Innsbruck, Austria, from 15-18 September 2013. Inter-Noise 2013 will be held at the Congress Center Innsbruck. The Congress is sponsored by the International Institute of Noise Control Engineering (I-INCE), and is being organized by the Austrian Noise Abatement Association. The theme of the congress is Noise Control for Quality of Life. For more information visit www.internoise13.org

### ICSV21 2014

The 21st International Congress on Sound and Vibration (ICSV21) will be held from 13-17 July 2014, in Beijing, China. The Congress is sponsored by the International Institute of Acoustics and Vibration (IIAV), and co-organized by the Acoustical Society of China (ASC) and the Institute of Acoustics, Chinese Academy of Sciences (IACAS). The theme of the Congress is *In Depth Sound and Vibration Research*, to concentrate on the physical insights into mechanisms of sound and vibration. Technical papers on this theme will be accepted and specially acknowledged. Other papers in all fields of sound and vibration will also be welcome. Companies are invited to take part in the ICSV21 exhibition and sponsorship. For further details visit www.icsv21.org

### Inter-Noise 2014

Inter-Noise 2014 is the 43rd International Congress and Exposition on Noise Control Engineering, and will be held in Melbourne from 16-19 November 2014. The venue will be the Melbourne Convention Centre on the banks of the Yarra River, with 12 rooms on the second floor available for the parallel technical sessions on Monday to Wednesday, while an expanded exhibition space will be located in the foyer. Morning and afternoon refreshments and a light lunch will be available in the middle of the exhibition area, permitting good interaction of delegates and exhibitors. Intending exhibitors should contact Norm Broner [NBroner@globalskm.com] about securing one of the prime display locations.

The Congress will commence on the afternoon of Sunday 16 November with a welcoming ceremony, followed by the first plenary lecture and light refreshments. We anticipate there will be four keynote lectures and two plenary talks during the Congress. Details of the speakers and their subject will be posted on our web-site, www.internoise2014.org once they are finalised. Already, over 80 people from many parts of Europe, UK, USA and Asia have agreed to chair or co-chair sessions during the Congress with, we are delighted to note, many Australian based researchers and consultants agreeing to assist. Topics include transport noise and vibration, low frequency effects on people and buildings, soundscapes in parks, wind turbine and gas-turbine noise, jet-noise and computational aero-acoustics, signal processing for active noise control, education and policy, light-weight building acoustics, low-noise tyres, virtual sources and underwater noise as well as psychoacoustics in noise evaluation and noise effects on humans, to name but a few.

### PRUAC 2013, Hangzhou

The Hangzhou Applied Acoustics Institute Research Institute (HAARI) and Zhejiang University are organising the 4th Pacific Rim Underwater Acoustics Conference in Hangzhou, China from 9-11 October 2013. PRUAC 2013 has the conference theme of "Underwater Acoustics and Ocean Dynamics". Every conference attendee will have the opportunity to participate in productive discussions on this specific theme. For more information see http://pruac.zju.edu.cn/index.htm

### ISMA 2014, Leuven, Belgium

The 26th International Noise and Vibration Engineering Conference, ISMA2014, will be held in Leuven (Belgium) from 15-17 September 2014. It will be organised in conjunction with the 5th edition of the International Conference on Uncertainty in Structural Dynamics - USD2014. Both conferences are organised by the division PMA of the KU Leuven. ISMA2014 follows the biennial international conferences on noise and vibration engineering, structural dynamics and modal testing. A single registration will grant access to both the ISMA and the USD conference. Information on the conference topics, as well as on the procedure for submitting abstracts are available from http://www.isma-isaac.be. The first important dates are 1st October 2013 for the start of online abstract submission and 15th January 2014 as deadline for submission of abstracts.

# Diary

## 2013

**26 – 28 August, Denver, USA**
Noise-Con 2013
http://www.inceusa.org/nc13

**27 – 30 August, Denver, USA**
Wind Turbine Noise 2013
http://www.windturbinenoise2013.org

**15 – 18 September, Innsbruck, Austria**
Inter-Noise 2013
http://www.internoise2013.com

**9 – 11 October, Hangzhou, China**
4th Pacific Rim Underwater Acoustics Conference
http://pruac.zju.edu.cn/index.htm

**17 – 20 October, New York, USA**
135th Audio Engineering Society (AES) Convention
http://www.aes.org/events/135/

**17 – 20 November, Victor Harbor, Australia**
Australian Acoustical Society annual conference
Acoustics 2013 Victor Harbor
http://www.acoustics.asn.au/joomla/acoustics-2013.html

## 2014

**1 – 5 June, Nara, Japan**
11th International Congress on Noise as a Public Health Problem (ICBEN 2014)
http://www.icben2014.com/

**6 –10 July, Beijing, China**
21st International Congress on Sound and Vibration (ICSV21)
http://www.iiav.org/index.php?va=congresses

**7 -12 September, Krakow, Poland**
Forum Acusticum 2014
http://www.fa2014.pl/

**16 – 19 November, Melbourne, Australia**
Inter-Noise 2014
http://www.internoise2014.org/

## 2015

**2 - 5 May, Singapore**
Wespac 2015
otsuru@oita-u.ac.jp

**10 – 15 May, Metz, France**
International Congress on Ultrasonics (2015 ICU)
http://www.me.gatech.edu/2015-ICU-Metz/

**31 May - 3 June, Maastricht, Netherlands**
Euronoise 2015
https://www.euracoustics.org/events/events-2015/euronoise-2015

**12 - 16 July, Brescia, Italy**
22nd International Congress on Sound and Vibration (ICSV22)
http://www.iiav.org/index.php?va=congresses

## 2016

**5-9 September, Buenos Aires, Argentina**
22nd International Congress on Acoustics (ICA 2016)
http://www.ica2016.org.ar/

*Meeting dates can change so please ensure you check the conference website: http://www.icacommission.org/calendar.html*

# Sustaining Members

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

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**industrialacoustics.com/australia**

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**www.environment.nsw.gov.au**

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Fax: (02) 8934 0001
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rdevereux@acran.com.au

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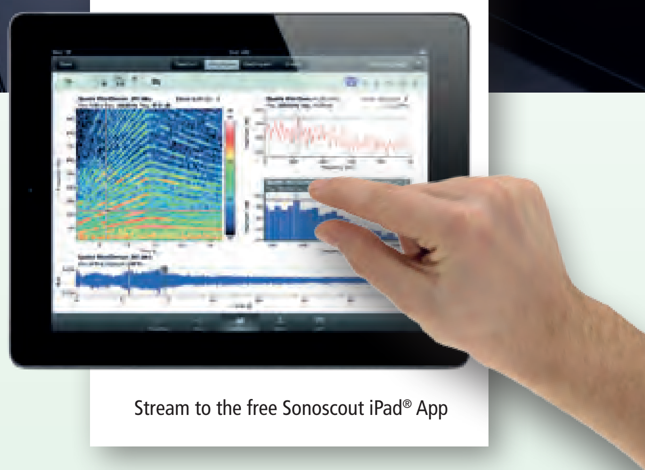

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