Acoustics Australia

Australian Acoustical Society

Volume 41 Number 3 December 2013

SignalCalc ACE

QUATTRO

ACE-QUATTRO has:

- Standard FFT, Synthetic 1/3 Octave standard,
- Quality Control, Point & Direction,
- Sound Power, Sound Intensity,
- Acoustic Intensity, Human Vibration,
- Sound Quality, Demodulation,
- Disk Record & Playback Analysis,
- Frequency Domain Compensation,
- Real Time Octave Analysis, Event Capture,
- 4-32 Inputs (Abacus), 2-8 Outputs + Tachy,
- ActiveX command & control macro programs,
- Single to Multi-Plane Balancing ...and more

2, 3 or 4 Inputs
0, 1 or 2 Outputs
1 Tachy + Trigger
Fast USB 2 for
Data & Power.

BSWA TECH

MEASUREMENT MICROPHONES.

microphones & accessories for Australian & NZ.

High quality, calibrated electret prepolarized microphones to class 1 & 2 & preamplifiers.

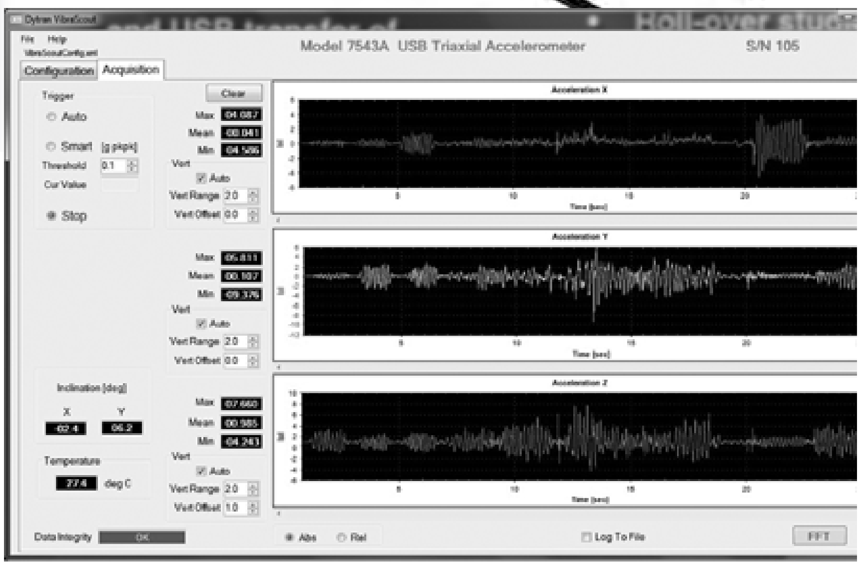
Some of the Accessories available:

- IEPE power sources, - portable sound level calibrators,
- USB powered 2 channel measurement soundcard
- a 12 speaker dodecahedral sound source,
- compact light weight 2 channel power amplifiers,
- a self contained Tapping Machine sound source for foot fall measurements.
- impedance tubes providing 125 Hz to 3200 Hz range,
- Outdoor noise monitoring, terminal and software
- Small Reverberation chamber,
- Artificial Ear, Mouth & Head

VibraScout USB DC Triaxial

Vibration Measurement System

from KINGDOM PTY LTD.



VibraScout system is an intelligent sensor including a USB digital Triaxial MEMS accelerometer, a microprocessor, a USB 4.6 m 4 pin cable, Data Acquisition software and Post Processor software. Just add your PC to get power and data exchange from the PC bus. VibraScout provides real-time 3 axis acceleration, static inclination, auto & smart triggering, RT Temperature recording, data logging with export to ascii, jpg, TDMS binary and conversion to PSD & FFT. Frequency span Zero to 1100 Hz.
Highly suitable for NVH, Vibration, calibration, rotating and more.

For more information phone: **02 9975 3272**
or visit: ***www.kingdom.com.au***

KINGDOM PTY LTD

AAS 07/10

Acoustics Australia

Acoustics Australia General Business
(subscriptions, extra copies, back issues, advertising, etc.)
Mrs Leigh Wallbank
P O Box 70
OYSTER BAY NSW 2225
Tel (02) 9528 4362
Fax (02) 9589 0547
wallbank@zipworld.com.au

Acoustics Australia All Editorial Matters
(articles, reports, news, book reviews, new products, etc)
The Editor, Acoustics Australia
c/o Nicole Kessissoglou
School of Mechanical and Manufacturing Engineering
University of New South Wales
Sydney 2052 Australia
+61 401 070 843 (mobile)
AcousticsAustralia@acoustics.asn.au
www.acoustics.asn.au

Australian Acoustical Society Enquiries see page 250

Acoustics Australia is published by the Australian Acoustical Society
(A.B.N. 28 000 712 658)
ISSN 0814-6039

Printed by
Cliff Lewis Printing
91-93 Parraweena Rd,
CARINGBAH NSW 2229
Tel (02) 9525 6588
Fax (02) 9524 8712
email: matt@clp.com.au
ISSN 0814-6039

Vol. 41, No. 3 December 2013

LETTERS

PAPERS

TECHNICAL NOTES

REGULAR ITEMS

Cover design: Helena Brusic

MESSAGE FROM THE PRESIDENT

Hello everyone,

As we approach the close of this year, everyone seems to be very busy. We have had a good year across the Society with some very good technical meetings and a very successful Annual Conference in Victor Harbor! Congratulations to Peter Heinze and his team on a great effort.

While talking about Peter, I want to take this opportunity to recognise Peter's efforts as President and then Past President over this past year and I want to thank him for his valuable support and input over this last year. And following our recent Federal Council meeting, it is with great pleasure that we welcome Tracy Gowen as VP this year. Tracy will become your President at next year's November Federal Council meeting. Congratulations Tracy and we look forward to your leadership and working with you.

At the Conference banquet, which was a very nice event, I had the pleasure of announcing the elevation of Peter Heinze (SA), David Mee (QLD) and Gillian Adams (QLD) to Fellows of the Society. Congratulations on a well-deserved recognition. We also announced the Education Grants for this year and the CSR Bradford Excellence in Acoustics Winner for 2013, Jonathan Cooper (Resonate Acoustics) and colleagues (Tom Evans & Dr Dick Petersen) for their paper on "Detailed tonality assessment procedure for a wind farm". The President's Prize for best paper at the Conference was awarded to William Roberston.

At the recent Federal Council meeting in Victor Harbor, your representatives decided on some very important initiatives for the Society. It was agreed to fund a major website upgrade with a goal to be ready by April 2014.

With regard to our Acoustics Australia journal, Council was advised that Nicole Kessissoglou, our Editor for the last four years, was resigning. Council thanks Nicole for her term as editor and for raising the quality of our publication. Marion Burgess has taken over as Interim Editor until we find a replacement. We are looking at alternative options for publication of AA and the results of the survey which are discussed in this issue were used to inform the way forward. The last hard copy of the journal as we know it will be the April 2014 issue. Otherwise, look forward to our new electronic edition.

Another exciting initiative is the AAS Research Grant. Council allocated $100,000 over 3 years, funded by proportional contributions from Divisions. The funding is to be matched at least 50:50 with third parties and we will call for submissions in March/April 2014. We thank Matthew Stead in particular for his input to the Research Grant process development.

Council also decided to increase our membership fees for Member and Fellow grade to $150 inc GST for the year 2014-2015. This is an increase of 15.4% and the first increase since 2007. This will assist in all the initiatives that we currently have including the new website which will add greater ease of use and improved functionality and allow our support of research initiatives that will benefit acoustics in Australia. The other grades will also increase proportionally.

And yes, Internoise 2014 is less than 1 year away. Please go to the website www.internoise2014.org and check it out.

I would like to take this opportunity to wish everyone best wishes for the Season, enjoy the break and come back refreshed. Let us hope that 2014 is a happy and successful year for us all.

Norm Broner
President AAS and President Internoise 2014

FAREWELL FROM THE EDITOR

I am sad to write that this is my last issue as editor. I'm very fond of the journal and over the last four years have had the pleasure of watching it expand gracefully with age (like most of us). The journal has recently grown such that it is no longer possible to list the short titles of the contents on the front cover!

I'd like to thank the Society for the editorship. It was (and still is) a daunting task. It was truly an honour and a privilege to run the journal independently. I'd also like to take this opportunity to thank the very many wonderful contributors to the journal, particularly the authors and reviewers (I am sure there are a number of reviewers who are relieved I won't be knocking at their door yet again). I'd like to thank Joe Wolfe, Peter Heinze, Norm Broner, Marion Burgess, Tracy Gowen, Pam Gunn, Leigh Wallbank, Louise Fraenkel, the news reporters from the various divisions and my husband Max Stanton, who has had to put up with my obsession with the journal for the last few years.

A recent survey on the journal has revealed that the majority of readers find articles of interest in each issue (see page 224 for a summary of the survey). I hope you continue to contribute to the journal, with your interest, letters, articles, news items, and any feedback. All the best!

Nicole Kessissoglou

SAVTek Sales+Rentals

SAVTek Pty Ltd
Sound and Vibration Technology

Phone: (07) 3300 0363 | www.savtek.com.au

LETTER TO THE EDITOR

Peter Alway, PO Box 2021, Boronia Park, NSW 2111
palwayacoustics@gmail.com

WIND FARM NOISE

For some time I have been following the technical political and medical difficulties associated with wind farms. The excellent technical note by Neville Fletcher [1] made me rethink the problem.

A few weeks ago Ray Hartog, a long standing Acoustical Consultant, took me up to Newcastle to look at a small, solitary wind turbine on Kooragang Island. This unit is small by today's standards being only 73 metres high and generating only 0.6 megawatts, but I felt that viewing a single unit would be instructive. The turbine, when we were there, was rotating at 28 rpm with a blade pass frequency of around 90 rpm. As we were very close we could hear the swish and measure with some inaccuracy the very low frequencies. While we didn't record levels we confirmed that we were able to register the fundamental frequencies of the turbine.

At the end of Neville's article he implied that 180 degree out of phase sound generation may reduce the transmitted noise. Usually with noise cancellation the position of the speaker and the polar plot of its signal are critical. However, at such low frequencies the speaker's polar plot should be circular and so make the effort more possible. Fixing a problem at its source, if that is possible, is usually the preferable thing to do.

On assuming a band pass frequency of around 1 Hz, Professor Fletcher drew attention to the coincidence of that low frequency with human pulse rate, walking pace and rhythm. This exemplifies the possible involvement of pressure pulses as wavelength at such low frequencies seems an unlikely direct cause of distress.

To my knowledge the reviewed literature is bereft of examples of people affected by wind farm noise, so I think it is reasonable to suggest that few are affected. Having said that, if even a small number are affected we should find the mechanism by which it occurs and then find a way to minimise the effect.

When I was in charge of the NAL test facilities, I once took a group of adult students into the anechoic room. After a short time one lady said that she felt pressure in her ears. I pointed out that as the large door to the room was open the pressure outside and inside was the same. With that the lady shouted at the top of her voice "WHY WON'T YOU BELIEVE ME?".

I have emphasised this last statement because today, more than ever, perception is reality. To some extent it always was, but now it is generally difficult to call on the authority of, say, a physicist or an engineer to support an argument, as opinion is regarded to be of greater importance. There is hardly any effective reaction against the State Government's science cutbacks or the Federal Government's 'quadrimate' (1 in 4) of CSIRO's staff.

I suspect that, for some people who live in quiet rural areas, their tympanic membrane may stretch to give maximum sensitivity searching for aberrant sound that they perceive may cause problems, in our case wind farm noise. I think this searching for a signal is exactly what happened to the lady in the anechoic room. This I believe can be a very uncomfortable feeling.

We could test this by placing the subject in an area where they can see some turbines, then play pleasant music at a reasonably loud level, say 80 dB SPL and see if, after say 30 minutes, the general distress they previously experienced subsides.

While I think a large proportion of the problems experienced may be explained by this ear discomfort, Stephen Cooper, a very experienced Acoustic Consultant, tells me that he has at least one subject who can tell when the vanes on the local generator are moving and when they are still. So it appears that there may be more than one cause and, therefore, there may be more to the story.

I would like to give another suggestion of how we could progress further. There is a very small possibility that, in some people, the low frequency beat from the generators could pull the heart or breathing rate into synchronisation. With the thunderous lack of any other explanation I suggest a low risk method of testing to hopefully eliminate this theory: Attach a CO2 sensor near the nose and/or a movement detector on the chest as well as attaching a portable ECG monitor to the susceptible subject. Each instrument should be fitted with a transmitter. The outputs could then be compared with the electrical output of a monitoring SLM that is recording the turbine.

Peter Alway
MAAS (retired)

ACKNOWLEDGEMENTS

Thank you Den for being available to discuss with me my mad and a little less mad ideas. All responsibility for these dubious ideas is mine.

REFERENCES

[1] N. Fletcher, "Musical rhythm, vibrato and wind turbine noise", *Acoustics Australia* **41**(2), 174-175 (2013)

CHARACTERISATION OF MULLOWAY *ARGYROSOMUS JAPONICUS* ADVERTISEMENT SOUNDS

Miles J.G. Parsons[1], Robert D. McCauley[1] and Michael C. Mackie[2]
[1]Centre for Marine Science and Technology, Curtin University, Western Australia
[2]Department of Fisheries, Government of Western Australia
m.parsons@cmst.curtin.edu.au

Increasingly, fishes are reported as using acoustic variations in calls for different environmental and social contexts. However, to understand call functions and their associated behaviours it is first necessary to separate and characterise the species call types. During the Austral summer, mulloway (*Argyrosomus japonicus*), a vocal sciaenid, aggregates to spawn in the lower regions of the Swan River, Western Australia. *In situ A. japonicus* calls recorded here exhibited call spectral peak frequencies between 175 and 350 Hz and pulse repetition rate of 59 Hz. These swimbladder driven calls were categorised into; short grunts of 1-6 pulses ('Bup'), more predominant as the aggregation forms and separates; long grunts comprising 11-32 pulses ('Baarp'), most prominent in the hours after sunset; and a series of short calls comprising 1-5 pulses ('Thup') that increase sharply in call rate over a period of tens of seconds. This last category was observed only once or twice each evening. The second category was divided into several types of call where a single audible tone can also be broken into two or more parts, often preceded by one or more short 'Bups' (for example, 'Bup-bup-baarp').

INTRODUCTION

Many species of fish are soniferous, producing sound in a variety of contexts, most commonly spawning [1-3]. The waters of Western Australia are home to many types of fish calls and choruses, some of which may be associated with spawning [4-6]. Passive listening to a chorus of aggregating fish can greatly improve a biologist's ability to delimit spawning areas for conservation of essential fish habitat and identify movement patterns of the callers without creating behavioural bias [7-12]. However, to understand the timing and spatial extent of spawning behaviour, it is necessary to characterise the functions of calls produced during the reproductive period and identify the mobility of the fish over the calling period. This is because fish reproduction (and vocalisation) can comprise a complex array of behaviours that are associated with spawning, for example competition or courtship, but the may be spatially and/or temporally separated from the act itself [13-15].

Sciaenidae is a very vocal family of fish known as croakers or drummers [16,17]. Often only the males of the species possess the specialised 'sonic' muscles used to vibrate the swimbladder and produce sounds for which the family is renowned and in many cases competing males call repetitively, either individually or in a group, to attract a female with which they can spawn [18]. Mulloway (*Argyrosomus japonicus*) have been shown to produce sounds during spawning [19] and while both male and female *A. japonicus* possess sonic muscles, in previous studies the males produced almost all of the advertisement related sounds [20].

During the Austral summer, mature *A. japonicus* form spawning aggregations in Mosman Bay, Swan River (Figure 1), where catch data from studies during the 2004-5 and 2005-6 spawning seasons reported a mean total length of 101 cm [21]. Many of the fish captured in those studies were close to spawning maturity (discharged milt upon capture) or had very recently spawned, confirming times of spawning [22].

The aims of the study detailed here were to describe *in situ* vocalisations of *A. japonicus* in Mosman Bay, produced at times when spawning is known to occur in the area. The study also investigated whether different types of call and their occurrence throughout an evening spawning cycle could be discriminated by the observer.

METHODS

Passive acoustic recordings were taken in Mosman Bay over 37 evenings between November and March, during the 2006, 2007 and 2008 spawning seasons, from 17:00 hrs (prior to sunset) to 01:00 hrs. In Mosman Bay, the river banks descend rapidly to a 21 m deep channel comprising a sand/silt substrate of low acoustic reflectance (Figure 1) [23]. A few artificial reefs and several depressions are present, some of which reach 22 m depth at high tide. During recordings the water temperature in the bay ranged between 18 and 26° C.

Acoustic data were acquired using omni-directional HTI-90U (Hi-Tech Inc., MS, USA) hydrophones connected to Centre for Marine Science and Technology (CMST) – Defence Science and Technology Organisation (DSTO) developed sea-noise loggers located on the riverbed. Highpass (50 Hz) and lowpass (1500 Hz) filters were applied at various stages of data processing to remove noise. Spectrograms were produced using a 1024 or 2048 point Hanning window with 0.7 overlap. For analysis, the start of each call was taken as the first detected amplitude peak in the call pressure waveform and referred to as the Call Initiation Peak (CIP). The end of a call was noted as the point at which the final pulse decayed below background noise. The following characteristics of each call were recorded: call duration, pulse period, number of pulses in a call, pulse repetition rate (PRR) and spectral peak frequency. Where calls were speculated to originate from the same source the time between calls was noted.

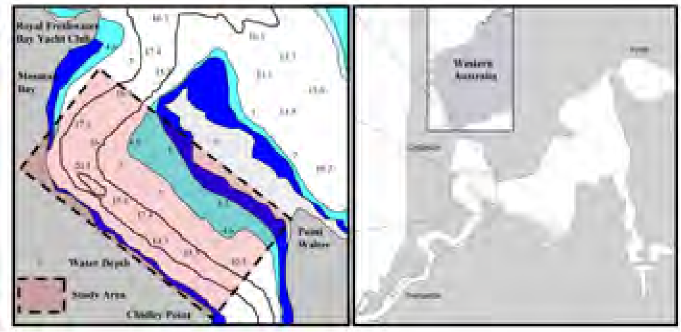


Figure 1. Map of the Mosman Bay study site and location within the Swan River, Western Australia

RESULTS

The light levels and turbidity at the time of *A. japonicus* calling restricts visibility to less than 2 m, thus video confirmation of calling was not possible. Lack of sexual dimorphism inhibited determination of sex of the calling fish. Anecdotal evidence from diver interactions with calling mulloway (including authors) confirmed them as the source of calls recorded in this study. Distress calls of *A. japonicus*, similar to the calls described here, have been reported anecdotally by fishers though they have not been recorded at this site.

Each evening, numerous *A. japonicus* calls were recorded with periods of low- and high-density calling. During low-density calling individual calls could be discriminated from each other and background noise (Figure 2). Calls were divided into three predominant categories, defined by the acoustic features and timing. Each call type comprised trains of swimbladder pulses of varying characteristics (Figure 2c, Table 1) and displayed sidebands of amplitude modulation typical of such sounds [24]. Between 19:30 and 23:00 many calls were masked by louder calls from other, closer fish and could not be counted. Due to interference, overlap between calls, or low signal-to-noise ratio, there were a number of calls where it

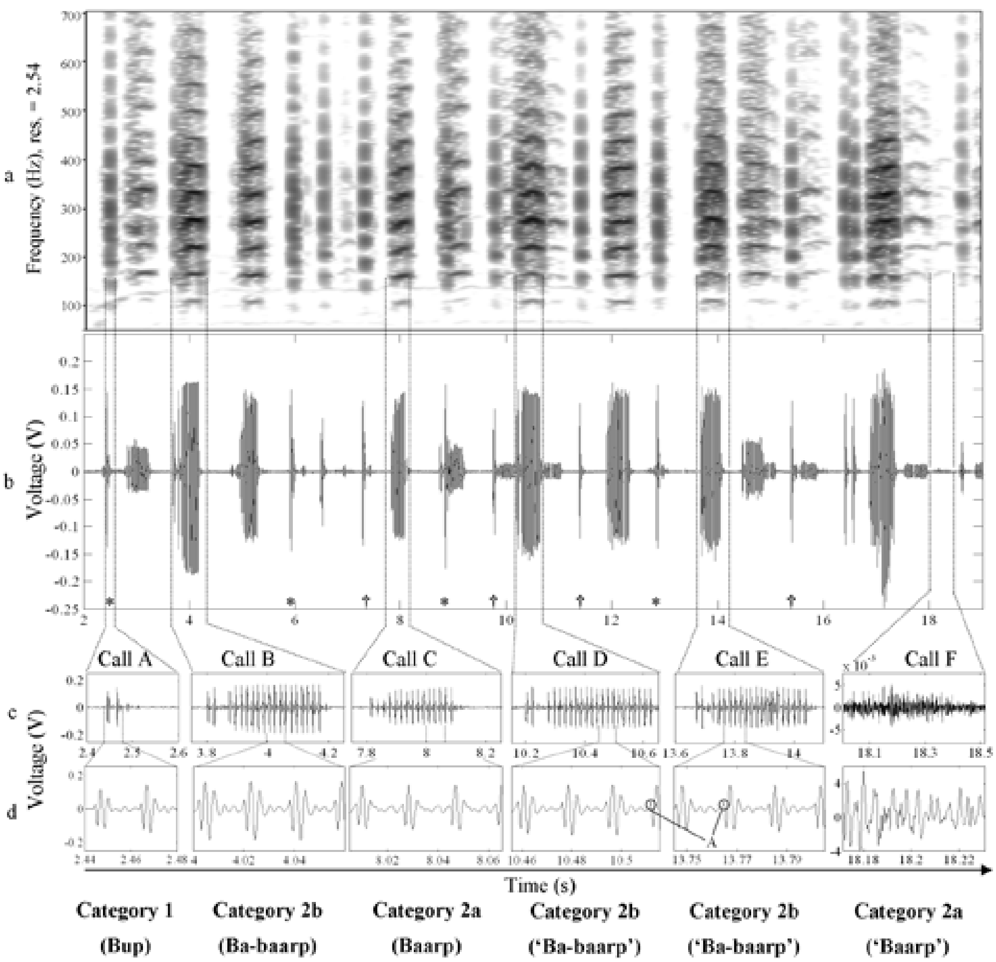


Figure 2. Spectrogram (a) and waveforms (b) from 17 seconds of Mosman Bay *A. japonicus* calling, recorded at 4 m depth in 19 m of flat water at 19:35, 17th January, 2007. Expansions of six selected call waveforms highlighting the entire calls (c) and sets of swimbladder pulses (d) are shown. Call F highlights an audible call of low signal-to-noise where waveform structure is distorted by noise. * and † denote examples of suspected repetitive Category 1 calls from individual fish.

Table 1. Example acoustic characteristics of all *A. japonicus* calls on the 5th March 2008 taken from the first minute of each hour between 17:30 and 23:31. Sunset occurred at 19:43

Time	Call Type	Number calls (no. analysed)	Call duration (s) x 10-1 (max, min)	Pulse number (max, min)	Modulation frequency (Hz) (max, min)	Spectral peak frequencies (Hz)
Total	1	509 (140)	0.56 ±0.25 (1.58, 0.26)	2.8 ±0.9 (6, 2)	52.6 ±10.9 (79.7, 36.2)	251
	2a	498 (170)	3.66 ±0.76 (5.27, 1.7)	21.6 ±4.5 (32, 9)	60.0 ±2.6 (63.8, 48.6)	250
	2b	81 (28)	3.94 ±0.68 (5.27, 2.68)	20.6 ±3.7 (30, 15)	52.4 ±3.9 (58.1, 43.00)	245
	2c	24 (12)	4.15 ±0.46 (4.54, 2.74)	22.75 ±4.0 (26, 18)	54.75 ±6.7 (62.6, 47.7)	275
	3	1 series (31)	0.22 ±0.12 (0.04, 0.09)	2.1 ± (4,1,1.09)	91.3 ±10.3 (114.2, 74.3) (22 measured)	260

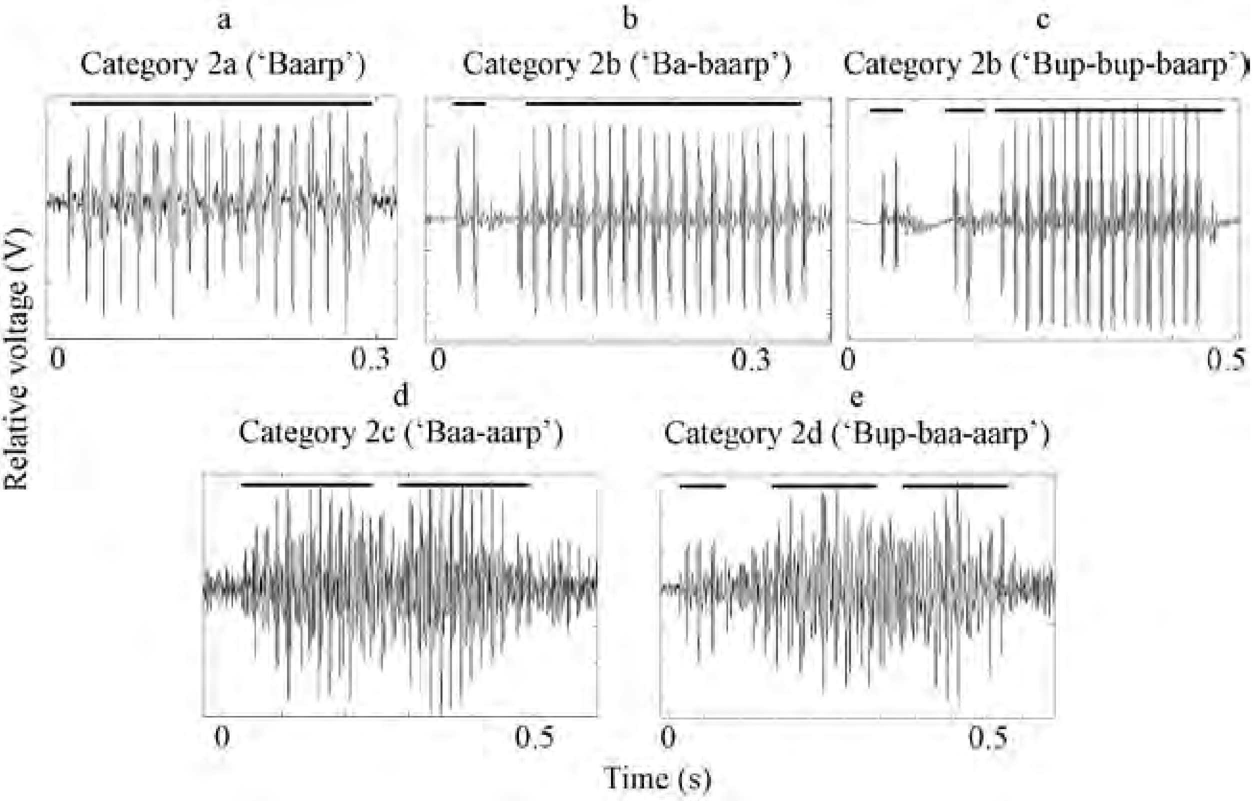


Figure 3. Waveforms of various detected Cat. 2 calls. Black lines shown above each waveform provide an impression of the audible periods of tone structure for each call type

was not possible to discern some acoustic characteristics (such as number of pulses or pulse duration) despite the call being distinguishable to the human ear. Across all call types spectral peak frequencies between approximately 175 and 350 Hz were observed, with sidebands of amplitude modulation at regular intervals (55.1 ±9.87 Hz, n = 350).

The majority of calls recorded were classified into two significantly different categories (Welch's t-test), depending on the number of pulses and duration of the call. To the ear Cat. 1 short calls (Figure 2c, Call A) sounded like a "Bup" and comprised 2.8 ± 0.92, n =140 pulses at a mean PRR of 52.6 Hz (Table 1). These signals were classed as an individual call if no further call, deemed to be from the same individual, followed within a second.

Cat. 2 calls were significantly longer than Cat. 1, comprising between 9 and 32 pulses (Table 1; Figure 2, Calls B-F). This category of calls comprised successive swimbladder pulses at sufficient PRR to be discerned by the listener a single audible tone (*pers. obs.*). However, this tone was often broken into constituent parts by a short cessation of pulses within the train (Figure 3, where the audible part of each call is marked with a black line). The gap in the acoustic tone most commonly occurred after the initial two swimbladder pulses and lasted between one and three pulse periods (Figure 3b). However, the position of this gap within the pulse train was found to vary. As a result, Cat. 2 calls were classified into five different types. Cat. 2a was a single audible tone, unbroken by pulse cessation ('Baarp'; Figure 2c Call C and Figure 3a). If the tone was preceded by one or more of the two pulse 'Bups' it was classed as Cat. 2b ('Bup-baarp' or 'Bup-bup-baarp'; Figure 2c

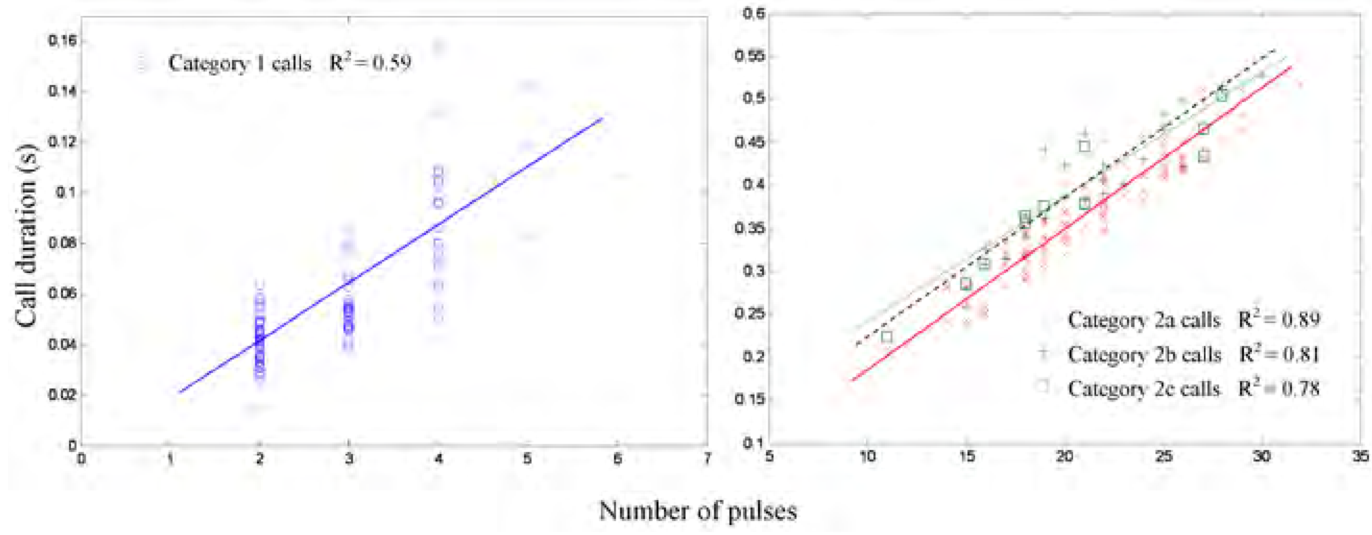


Figure 4. Distribution of calls as a function of numbers of pulses within the call (a), together with the relationship between the number of pulses and the duration of the call for Cat. 1 (b) and Cat. 2 (c) calls. Correlation coefficients of Cat. 1(○), 2a (x), 2b (+) and 2c (□) calls were r^2=0.59, 0.81, 0.81, 0.78, respectively.

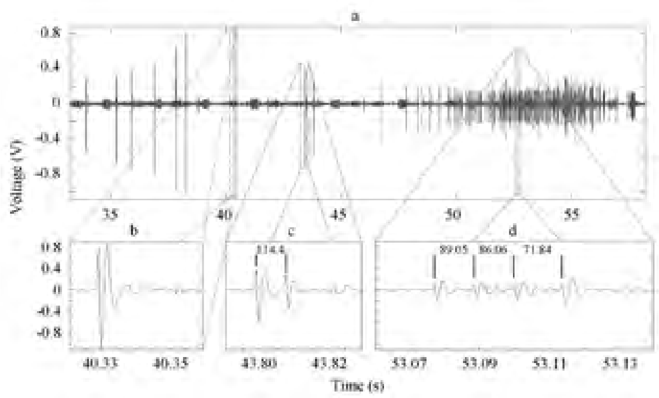


Figure 5. Waveforms of a series of Category 3 calls (a) recorded on the 8th March, 2008 at 19:57 post sunset. Expansions of single (b), double (c) and quadruple (d) pulse calls within this category are also shown with pulse repetition rates highlighted above (Hz). As with all Category 3 calls the PRR of the multiple pulses in (d) decreased through the call (i.e. the spacing between pulses increased)

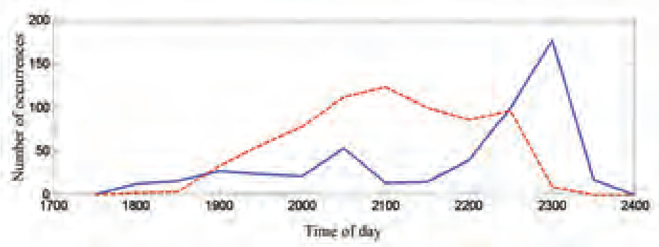


Figure 6. Number of occurrences for each call in the first minute of every half hour of an example evening spawning cycle from 17:00 to 24:00. Category 1 (continuous line) and Category 2 (dashed line) are shown, however, an unknown number of calls could not be counted between 19:30 and 23:31 due to call overlap

Calls B, D and E, and Figure 3c and c). Cat. 2c calls contained a break later in the call ('Baa-aarp'; Figure 3c). Finally, Cat. 2d calls contained a number of different parts characterised by two of more points of cessation within the call ('Bup-baa-aaarp'; Figure 3c).

In general, recorded mean peak-to-peak amplitudes of the first cycle in the pressure waveforms of Cat. 2 calls were 30-50% greater than those of Cat. 1 calls. This observation did not account for caller position and therefore signal propagation to the hydrophone, although a random distribution of Cat. 1 and Cat. 2 caller ranges was assumed. Additionally, it was observed that in many cases the first one and often two initial pulses of the long calls were of lower detected amplitude than the successive pulses (Figure 3c). The distribution of calls as a function of the number of pulses within a call illustrates separation between short Cat. 1 and long Cat. 2 calls (Figure 4a). There was a distinct relationship between the number of pulses in a call and the call duration (Figure 4b and c) in both categories.

During the hour prior and post sunset, series of calls were often recorded which could not be classed as Cat. 1 or 2 calls, and so were deemed of a third category (Figure 5). This call category was less frequent than the others, observed only once or twice in an evening, throughout the spawning season. These Cat. 3 calls each comprised 1-5 pulses at PRRs of 91.3 ±10.3 Hz (max = 114.2, min = 74.3, n =22), significantly higher than those of Cat. 1 and 2 calls (Table 1 and Figure 5c and d). The calls began with seconds between each call and increased in rate to a maximum with several multiple pulse calls per s (Figure 5a, at approximately 55 s).

Evening calling cycles (within the hydrophone detection range) typically began approximately 2 hrs before sunset with few Cat. 1 calls from a small number of distant individuals (Figure 6), although on occasion these were recorded up to 4 hrs before sunset. As calls became recorded at increasingly closer range from the hydrophone they became of sufficient signal-to-noise ratio to analyse acoustic characteristics (Table 1). With time the number of Cat. 1 calls increased, along with the number of callers (Figure 6). By comparing waveform amplitude, shape and spectral peak frequency and localisation data [11] it was possible to discriminate between some callers and note individual repetitive calling (Figure 2, marks * and †). At times of low calling density this discrimination allowed a mean estimate of repetitive calling rates of 3.6 ± 0.85 s (n = 17) for Cat. 1 calls.

The number of Cat. 2 calls increased as sunset approached, with types 2a, 2b, 2c and 2d in order of occurrence (Table 1, Figure 6) and repetitive calling was determined at 3.72 ± 0.65 s between Cat. 2 calls. The peak in call numbers occurred approximately an hour after sunset and during this period predominantly Cat. 2 calls were observed (Figure 6). Whether Cat. 1 calls were not emitted at this time or were masked by Cat. 2 calls could not be confirmed. Cat. 2 calls then became less frequent and Cat. 1 calls were heard again, in greater numbers than before (Table 1, Figure 6). Cat. 1 calling intervals at this time ranged between approximately 1.8 and 3.1 s (calling rates of each individual reduced in rate as the evening progressed). Several hours after sunset the Cat. 2 calls had all but disappeared leaving a few callers emitting Cat. 1 calls of comparatively low received SPLs, typically between the hours of 22:00 and 00:00, until all calls ceased.

DISCUSSION

The *in situ* recordings demonstrated that Mosman Bay *A. japonicus* have a greater variety of vocalisation linked to times of spawning than previously thought [25,26]. In addition, a greater variety of calls were recorded here than similar studies in Taiwan [19,20], possibility illustrating the behavioural changes in geographically separated populations. This is a large repertoire, similar to the Atlantic croaker *Micropogonius undulates* [27], compared with that of other species [8,16,28,29]. *A. japonicus* produce sounds via multiple contractions of sonic muscles, exciting the swimbladder in a train of pulses [19]. In contrast to many soniferous Sciaenidae, such as the weakfish *Cynoscion nebulosus* [8], *A. japonicus* PRRs are greater, such that the produced sound can be a singular tone, rather than a series of knocks, similar to *Argyrosomus regius* [30].

Assuming call source levels of different, but similar sized fish are comparable [27], the difference in detected waveform amplitudes show that individual fish are separated by a minimum distance. Consistency in this separation highlights the low density of calling fish in the recording area and corroborates the suggestion of individual calling territories for *A. japonicus* in the wild [26]. This separation also supports a proposal of pair spawning in Mosman Bay, rather than group spawning where an indistinguishable (dense) chorus would be more prominent, similar to that of other species [16,30,31]. Thus while callers are exhibiting repetitive calling behaviour from stationary, or near stationary, locations it is possible to observe the different fish within the detection range of the hydrophone. A continuous chorus does form in Mosman Bay, during peak calling, however, the high source levels of *A. japonicus* calls [26], compared with those of other fish [26,32] means that fish from greater ranges contribute significantly to the overall sound pressure levels in the chorus. They would therefore still be able to call from separate locations and still form a chorus from an aggregation that is spread over a considerable area.

Lagadere and Mariani [30] observed that *A. regius* short calls are of lower intensity than the long calls, similar to the Cat. 1 calls here, compared with Cat. 2 calls. However, in many Cat. 2 calls the initial pulses were also of lower amplitude, raising the question of whether the sonic muscles take time to attain the tension required to generate amplitudes exhibited by later pulses of the long calls. Further study, including analysis of muscle tension during contraction is planned to elucidate this.

ACKNOWLEDGEMENTS

The authors would like to acknowledge the Australian Government, via the Fisheries Research and Development Corporation (FRDC) for funding of this work. Mal Perry and Frank Thomas are acknowledged for time and effort exercised during the course of the project's fieldwork. The Department of Fisheries, Western Australia has provided logistical support during the course of this study.

REFERENCES

[1] H.E. Winn, "The biological significance of fish sounds", in *Marine bioacoustics*, edited by W.N. Tavolga, Pergamon Press, New York, 1964, pp. 213-231

[2] M.L. Fine, H.E. Winn and B.L. Olla, "Communication in fishes", in *How animals communicate*, edited by T.A. Sebeok, Indiana University Press, Bloomington, 1977, pp. 472-518

[3] J. Höglund and R.V. Alatalo, *Leks*, Princeton University Press, USA, 1995, pp. 49-147

[4] M.J.G. Parsons, R.D. McCauley and F. Thomas, "The sounds of fish off Cape Naturaliste, Western Australia", *Acoustics Australia* **41**(1), 58-64 (2013)

[5] M.J.G. Parsons, P. Lewis, S.L. Longbottom, R.D. McCauley and D.V. Fairclough, "Sound production by the West Australian dhufish (*Glaucosoma hebraicum*)", *Journal of the Acoustical Society of America* **134**(4), 2701-2709 (2013)

[6] R.D. McCauley, "Fish choruses from the Kimberley, seasonal and lunar links as determined by long-term sea-noise monitoring", *Proceedings of Acoustics 2012 Fremantle*, Western Australia, 21-23 November 2012

[7] A.D. Hawkins, "The use of passive acoustics to identify a haddock spawning area", *Proceedings of an International Workshop on the Applications of Passive Acoustics in Fisheries*, Dedham, Massachusetts, 8-10 April 2002, pp. 43-47

[8] J.J. Luczkovich, H.J. Daniel III, M.W. Sprague, S.E. Johnson, R.C. Pullinger, T. Jenkins and M. Hutchinson, *Characterization of critical spawning habitats of weakfish, spotted seatrout and red drum in Pamlico Sound using hydrophone surveys*, North Carolina Department of Environment and Natural Resources, Division of Marine Fisheries, Morehead City, Final Report F-62-2, 1999

[9] J.J. Luczkovich, H. J. Daniel III, M. Hutchinson, T. Jenkins, S. E. Johnson, R.C. Pullinger and M.W. Sprague, "Sounds of sex and death in the sea: Bottlenose dolphin whistles suppress mating choruses of silver perch" *Bioacoustics* **10**, 323-34 (2000)

[10] R.A. Rountree, R.G. Gilmore, C.A. Goudey, A.D. Hawkins, J.J. Luczkovich, and D.A. Mann, "Listening to fish: Applications of passive acoustics to fisheries science", *Fisheries* **31**, 433-446 (2006)

[11] M.J.G. Parsons, R.D. McCauley, M.C. Mackie, P.J. Siwabessy and A.J. Duncan, "Localisation of an individual mulloway (*Argyrosomus japonicus*) within a spawning aggregation and their behavioural characteristics throughout a diel spawning period", *ICES Journal of Marine Science* **66**, 1007-1014 (2009)

[12] M.J.G. Parsons, R.D. McCauley, M.C. Mackie, P.J. Siwabessy and A.J. Duncan, "A comparison of techniques for ranging close range mulloway (Argyrosomus japonicus) calls using a single hydrophone", *Acoustics Australia* **38**(3), 145-151 (2010)

[13] F. Engen and I. Folstad, "Cod courtship song: A song at the expense of dance?", *Canadian Journal of Zoology* **77**, 542–550 (1999)

[14] A.D. Hawkins and M.C.P. Amorim, "Spawning sounds of the male haddock, *Melanogrammus aeglefinus*", *Environmental Biology of Fishes* **59**, 29-41 (2000)

[15] J.T. Nordeide and I. Folstad, "Is cod lekking or a promiscuous group spawner?", *Fish and Fisheries* **1**, 90-93 (2000)

[16] H.K. Mok and R.G. Gilmore, "Analysis of sound production in estuarine aggregations of *Pogonias cromis*, *Bardiella chrysoura*, and *Cynoscion nebulosus* (Sciaenidae)", *Bulletin of the Institute of Zoology, Academia Sinica* (Taipei) **22**, 157-186 (1983)

[17] M.P. Fish and W.H. Mowbray, *Sounds of Western North Atlantic fishes*, The Johns Hopkins Press, Baltimore, 1970

[18] M.D. Hauser, *The Evolution of Communication*, MIT Press, Cambridge, MA, 1996, pp. 111-145

[19] J.P. Ueng and B.Q. Huang, "Circadian rhythm of sound production in Japanese croaker, Argyrosomus japonicas", *Journal of the Fisheries Society of Taiwan* **25**, 287-294 (1998)

[20] J.P. Ueng, B.Q. Huang and H.K. Mok, "Sexual differences in the spawning sounds of the Japanese Croaker *Argyrosomus japonicus* (Sciaenidae)", *Zoological Studies* **46**, 103-110 (2007)

[21] B.M. Farmer, D.J.W. French, I.C. Potter, S.A. Hesp and N.G. Hall, "Determination of biological parameters for managing the fisheries for Mulloway and Silver Trevally in Western Australia", *Fisheries Research and Development Corporation Report, FRDC Project* 2002/004 (2005)

[22] B.M. Farmer, *Comparisons of the biological and genetic characteristics of the Mulloway Argyrosomus japonicus (Sciaenidae) in different regions of Western Australia*, PhD Thesis, Murdoch University, Australia, 2008

[23] F.B. Jensen, W.A. Kuperman, M.B. Porter and H. Schmidt, *Computational Ocean Acoustics*, Springer-Verlag, New York, 2000, pp 36-49

[24] W.A. Watkins, "The harmonic interval: Fact or artefact in spectral analysis of pulse trains", in *Marine bioacoustics*, edited by W.N. Tavolga, Vol. 2, Pergamon Press, New York, 1967, pp. 15-42

[25] M.J.G. Parsons, R.D. McCauley and M.C. Mackie, "Spawning sounds of the mulloway (*Argyrosomus japonicus*)", *Proceedings of Acoustics 2006*, Christchurch, New Zealand, 20-22 November 2006, pp. 401-406

[26] M.J.G. Parsons, R.D. McCauley, M. Mackie, and A.J. Duncan, "In situ source levels of mulloway (*Argyrosomus japonicus*) calls", *Journal of the Acoustical Society of America* **132**, 3559-3568 (2012)

[27] D.P. Gannon, "Acoustic behavior of Atlantic croaker, *Micropogonius undulates* (Sciaenidae)", *Copeia* **1**, 193-204 (2007)

[28] M.A. Connaughton, M.H. Taylor and M.L. Fine, "Effects of fish size and temperature on weakfish disturbance calls: Implications for the mechanism of sound generation", *Journal of Experimental Biology* **203**, 1503-1512 (2000)

[29] J. Nilsson, *Acoustic behaviour of spawning cod (Gadus morhua L.)*, Candidatus scientiarum Thesis, University of Bergen, Norway, 2004

[30] J.P. Lagardere and A. Mariani, "Spawning sounds in meagre Argyrosomus regius recorded in the Gironde Estuary, France", *Journal of Fish Biology* **69**, 1697-1708 (2006)

[31] M.W. Sprague and J.J. Luczkovich, "Quantifying species-specific contributions to the overall sound level", P*roceedings of an International Workshop on the Applications of Passive Acoustics in Fisheries*, Dedham, Massachusetts, 8-10 April 2002, pp. 147-153

[32] M.A. Connaughton, M.L. Fine and M.H. Taylor, "Weakfish sonic muscle: Influence of size, temperature and season", *Journal of Experimental Biology* **205**, 2183-2188 (2002)

BEM SIMULATIONS OF DIFFRACTION-OPTIMIZED GEOMETRICAL NOISE BARRIERS, WITH A FOCUS ON TUNABILITY

Sara Gasparoni[1], Paul Reiter[1,2], Reinhard Wehr[1], Marco Conter[1] and Manfred Haider[1]
[1] AIT, Austrian Institute of Technology, Vienna, Austria
[2] Technical University of Vienna, Vienna, Austria
sara.gasparoni@ait.ac.at

Traffic noise is an increasingly important problem with the increase in traffic volume. To counteract this, noise barriers are the most used traffic-noise-abatement tool. In an attempt to reduce the amount of material, and thus the costs for the construction of noise barriers, it is of interest to reduce the height of the barriers. One possibility to reduce the height is to use absorbing materials. This is a good solution but the porosity of these materials makes them very sensitive to clogging by dirt and changes their absorbance and their performance with time. In this paper, non-standard geometrical forms of noise barriers with added devices are investigated. The boundary element method is used to investigate the insertion loss produced by these noise barriers. This method is also used to propose tunable barriers that could adapt to the changing noise spectrum.

INTRODUCTION

Traffic noise is increasing with the increase in traffic volume. There are different methods that can be used to reduce traffic noise, for example traffic noise could be reduced at the source by producing low-noise asphalts or low-noise tyres, or at the receiver by using sound-absorbing materials for buildings. Noise barriers are a common way to shield residential areas from traffic noise, mainly because they can be built *ad hoc* after the problem of traffic noise has shown up.

While the efficiency in noise absorption can be evaluated experimentally with reverberation room and situ methods [1-3], simulations are a helpful and cost-saving tool in predicting and planning new noise barrier solutions. The boundary element method (BEM) has long been used in the simulation of noise barriers, as it proves to be an effective method whose results are compatible both with analytical solutions and experimental results [4]. Non-standard shapes of the barrier tops have been used to obtain a better performance of the barrier [5-10].

This paper uses a BEM simulation to study non-standard barrier shapes with a particular focus on the formation of a virtual soft plane for some frequencies. Destructive diffraction from the top edge of the barrier is used in order to optimize the shielding effect of the barrier. After verifying the effectiveness of alternative shapes, the possibility of tuning the barrier in frequency is examined. This is an interesting novel theme to be explored as it may give the opportunity to adapt the barrier to a changing noise environment.

BARRIER NUMERICAL MODEL

Perfectly reflecting (acoustically hard) materials are not considered the best choice for noise barriers, as they generate many unwanted reflections. Absorbing materials seem much more appropriate, but their impedance changes rapidly with time in the presence of dirt, which is in the case of highways. Different geometrical shapes have been used to obtain a specific input impedance for improved performance [5-7].

Consider the fork-shaped barrier shown in Figure 1(b). If the surfaces are rigid, the specific input impedance at the open side can be approximated by [5]

$$Z_{in} = i \cot(kd) \qquad (1)$$

where d is the depth of the fork and k is the wavenumber. According to this equation, at frequencies f_n with $k_n d = (2n+1)\pi/2$, the impedance is zero. This means that for the range close to those frequencies, the fork element plays the role of a soft plane with complete absorbance. The condition of a soft plane can never be fully realized with the use of absorbent materials, which makes the geometric solution a useful alternative. As this impedance is only dependent on the geometry, the problem of the time-variance of the absorbent materials is practically solved. On the other hand, this solution is efficient only for some frequencies.

This problem can be dealt with using a barrier whose channels have different lengths, using the fork gradient shown in Figure 1(c). This corresponds to using a strip of absorbent material whose impedance changes gradually along the length of the material. The idea is similar to chirped mirrors in optics, made out of different layers that can filter different wavelengths.

For the current investigation, a 2D BEM analysis has been performed, assuming the invariance of the system on the *y*-axis. OpenBEM, an open-source software developed in the Matlab environment by the University of Southern Denmark [11], has been used. OpenBEM solves the Helmholtz equation using a direct collocation approach.

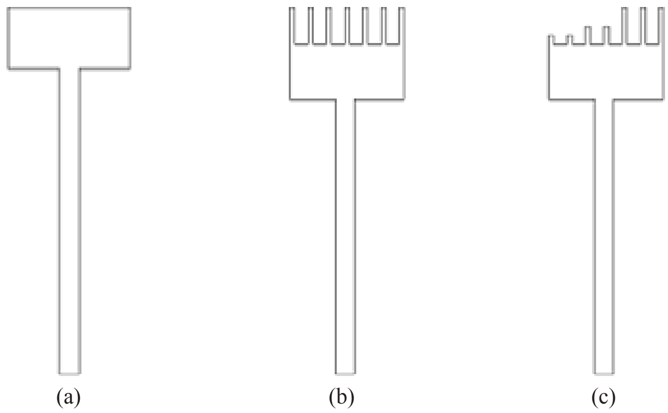


Figure 1. Different barriers used for the simulation corresponding to (a) T-shape, (b) fork shape, (c) fork gradient

The set-up for the simulation is shown in Figure 2. In the numerical model, the ground is assumed to be perfectly reflecting. The source is placed on the ground, in order to prevent unwanted reflections, and at 8 m distance from the barrier. On the other side of the barrier, 9 microphones are placed in a regular grid structure, at the different heights of 0, 1.5 m and 3 m from the ground, and at the distances 20, 50, 100 m from the barrier. Simulations are performed at the middle frequencies of the one-third octave bands. Simulations are initially performed without barrier, then with the three different barriers shown in Figure 1 and with a normal barrier of the same height and without added devices. Similar simulations have also been performed in the presence of absorbent materials.

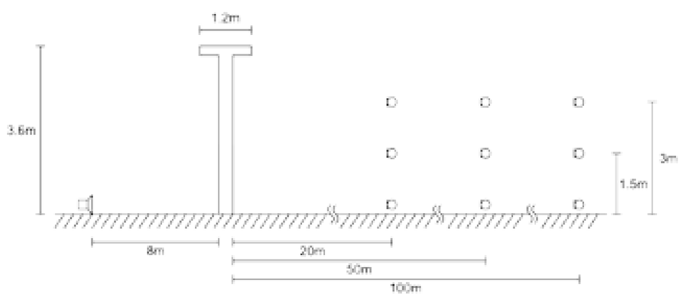


Figure 2. The simplified set-up of the BEM simulations

It is possible to add material properties in the OpenBEM software. The method used in the simulations follows the semi-empirical law of Delany and Bazley [12]. Allard and Champoux [13] derived the following empirical formulae

$$k = \left(\tfrac{\omega}{c}\right)[1 + 0.0978X^{-0.7} + i0.189X^{-0.595}] \quad (2)$$

$$Z_c = 1 + 0.00571X^{-0.754} + i0.087X^{-0.732} \quad (3)$$

where $X = \rho_0 f / R_s$, ρ_0 is the density of air, R_s is the flow resistivity, f is the frequency and Z_c is the normal surface impedance [13]. This semi-empirical model, drawn from the best fits of a large number of impedance tube measurements, is valid for $0.01 < X < 1.0$. In the simulations, a flow resistivity of 30000 Ns/m^4 was used, which is a good description for mineral wool applied to the noise barrier and well within the values of validity of Eqs. (2) and (3).

The insertion loss (*IL*) in dB is calculated as an average value for the 9 points, according to the formula

$$IL = -10\log_{10}(R) \quad (4)$$

$$R = \frac{1}{n}\sum_{i=1}^{n}\left(\frac{p_i}{\bar{p}_i}\right)^2 \quad (5)$$

where p_i represents the pressure on the i^{th} microphone and $\bar{p}_i$ the pressure on the i^{th} microphone position of the configuration without a barrier.

RESULTS

The results of the insertion loss for the various barrier designs corresponding to a straight barrier, T-shape, fork shape and fork gradient barrier are shown in Figure 3. The improvement of the *IL* using the fork shape and fork gradient barriers compared to the straight and T-shaped barriers can be seen in Figure 3. At some frequencies, an improvement of up to 10 dB for the fork shape can be found. A decrease in *IL* at around 500 Hz for the fork shape can be observed and corresponds to the maximum of the impedance. According to Eq. (1), the maximum *IL* is expected when $kd = \pi/2$, which for d = 400 mm, occurs at the frequency f = 210 Hz. The results for the fork shaped barrier also shows peak *IL* between 600 and 800 Hz.

The use of the fork gradient barrier, where channels of different depths are used, presents an input impedance that changes along the length. It represents a considerable improvement in the insertion loss as the attenuation is better distributed along the considered frequency range, as shown in Figure 3. The *IL* at 500 Hz has increased by around 5 dB for the fork gradient barrier compared to the fork shaped barrier.

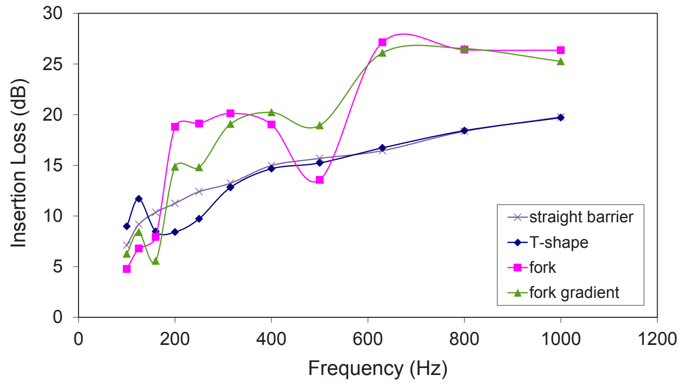


Figure 3. Insertion loss for the various barrier designs corresponding to a T-shape, fork-shape, fork gradient and a straight barrier

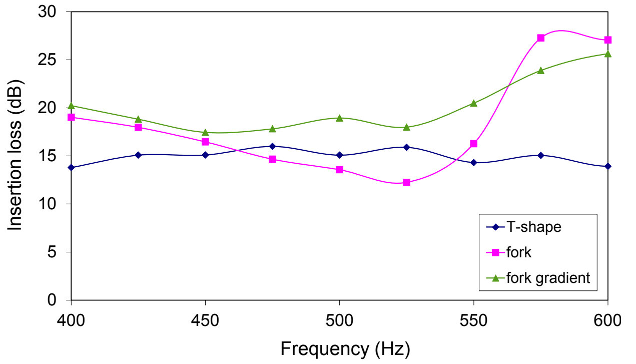
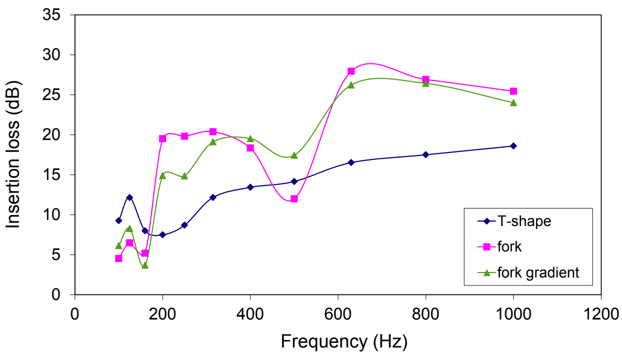
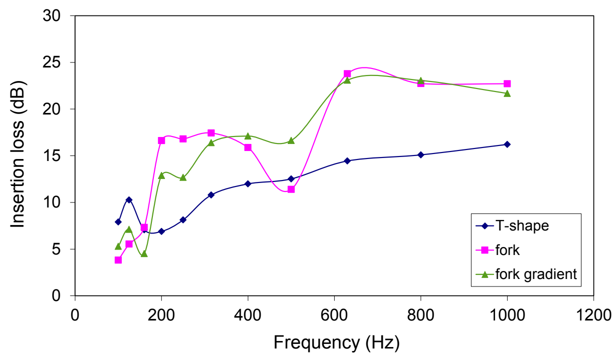


Figure 4. Insertion loss for a narrow band frequency range from 400 to 600 Hz



(a)



(b)

Figure 5. Insertion loss at two different receiver points on the ground at (a) 20 m and (b) 100 m from the barrier

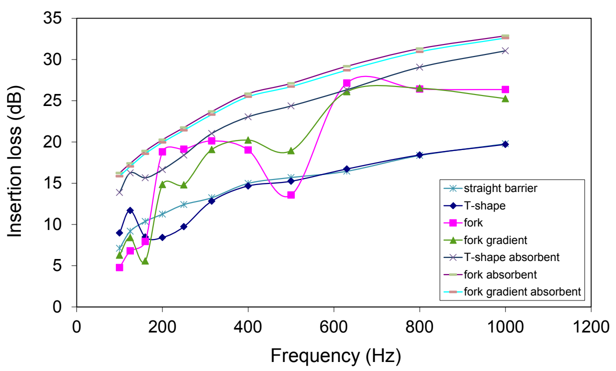
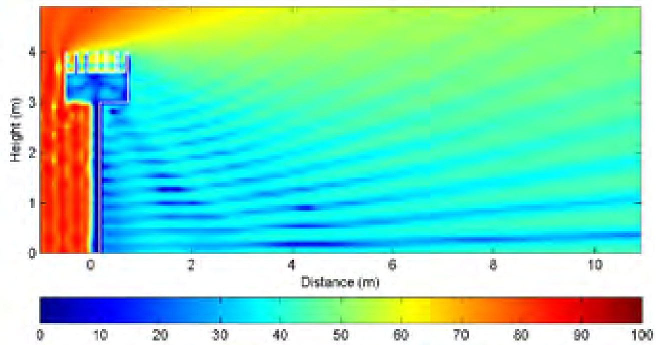
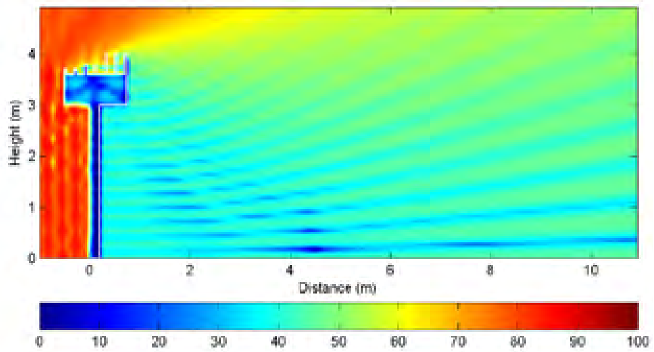


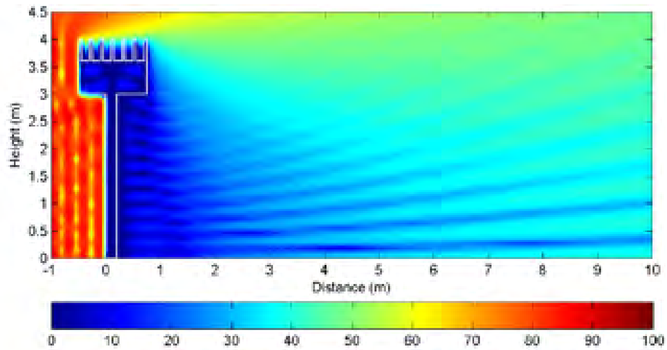
Figure 6. Insertion loss for the various barrier designs with and without mineral wool



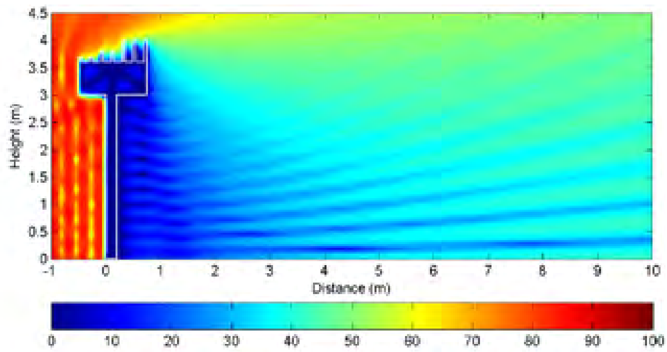
(a)



(b)



(c)



(d)

Figure 7. Sound pressure level at a frequency of 630Hz for (a) the fork barrier without absorbent material, (b) the fork gradient barrier without absorbent material, (c) the fork barrier with absorbent material, and (d) the fork gradient barrier with absorbent material

It is of interest to observe the insertion loss for the various barrier designs in a narrower frequency range corresponding to frequencies between 400 Hz and 600 Hz. In order to achieve better accuracy, simulations have been performed at increase frequency resolution. From Figure 4 it is clear that the *IL* related to the fork barrier is smooth and the dip in *IL* is broad. The same applies to a higher degree for the fork gradient barrier.

An average value of the insertion loss for the 9 microphones positions was calculated in order to represent an arbitrary point in the far field. In Figure 5, two graphs are shown that correspond to the points on the ground, one at 20 m and one at 100 m from the barrier. The average insertion loss given by Eqs. (4) and (5) is a good representation of the single location, as the behaviour is very similar to the results shown in Figure 3.

In Figure 6, simulation results are presented for the case where mineral wool is covering the T-shape, fork shape and fork gradient barriers. The presence of the mineral wool consistently improves the efficiency of the barrier, as can be seen by comparing the insertion loss of the T-shape barrier with and without the absorbent wool. For the fork shape and fork gradient barriers, applying the mineral wool results in an increase in insertion loss as well as a flattening of the insertion loss over the considered frequency range. Figure 6 shows that the best performance of the barrier occurs by both modifying the geometrical shape and applying the absorbent wool. Even if clogging deteriorates the absorbent properties of the mineral wool, the geometrical properties still remain for a long-lasting performance.

The sound pressure level (SPL) at a frequency of 630Hz is presented in Figure 7 for the fork shaped barrier and the fork gradient barrier, with and without the presence of absorbent material. On the edge of the barrier, the formation of the soft plane is clearly visible. According to Eq. (1), the soft plane is expected at about 630 Hz. In the case of absorbent barriers, the frequency behaviour is flat and no soft plane can be observed.

TUNABILITY

The geometrical profile of the top of the noise barrier can be used to tune the barrier. From Eq. (1), it is observed that changing the depth *d* gives rise to a shift in the frequencies where maximum absorption takes place. To investigate the effect of the depth, further simulations with the shapes illustrated in Figure 8 have been performed.

The first shape h_1 has a depth *d* of 100 mm. From Eq. (1), the first maximum absorbance for the top layer which corresponds to $Z_{in} = 0$ is expected at $kd = \pi/2$. This occurs at a frequency of f_0 = 840 Hz. By doubling the comb depth *d* for profile h_2, an absorption peak at 420 Hz occurs. Interestingly, the frequency f_0 which is a maximum of the absorbance for h_1 corresponds to a minimum of the absorbance for h_2, which means that for h_2 most of the radiation will be reflected. A doubling of the depth *d* results in opposite behaviour of the two barriers in the considered frequency range. At frequencies corresponding to peak *IL* for h_1 there is minimum *IL* for h_2 and vice-versa. Hence, the insertion loss spectrum for h_2 represents the mirrored image of the same spectrum of h_1. A similar situation is expected for h_2 and h_4, h_3 and h_6, for which a doubling of comb depth *d* also occurs.

In Figure 9, maximum insertion loss for barrier h_1

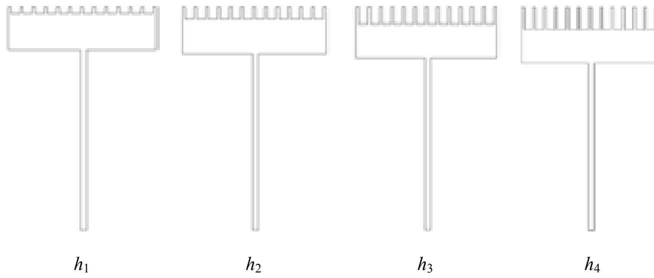


Figure 8. Fork shape barriers of different depths used for the tunability simulations

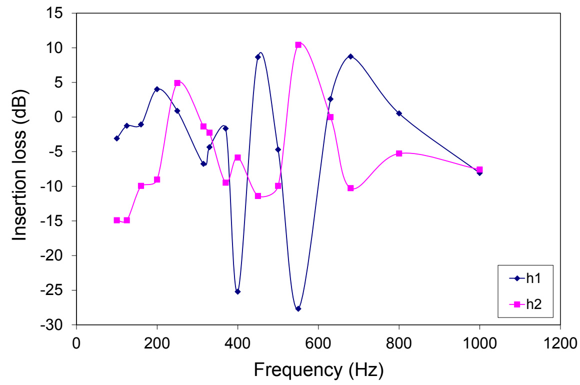


Figure 9. Insertion loss for barriers h_1 and h_2

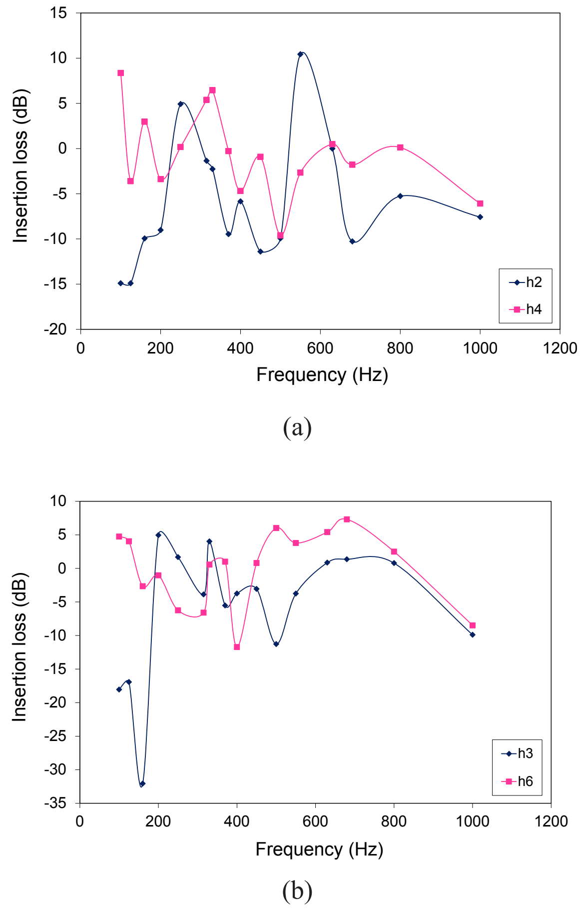


Figure 10. Insertion loss for barriers (a) h_2 and h_4 and (b) h_3 and h_6

occurs at around 680 Hz, while minimum insertion loss corresponding to $kd = \pi$ should occur at around 420 Hz. In reality the minimum is at about 550 Hz, showing that the Eq. (1) is only an approximation. A real part corresponding to $Z_{in} = i\cot(kd) + r_s$ is also present. As expected from the previous discussion, h_2 presents a mirrored behaviour compared to h_1. Similar behaviour for barriers h_2 and h_4 as well as for barriers h_3 and h_6 occurs, as shown in Figure 10. However, the effect is less clear, probably because the longer length of the channels imply more viscous effects [6].

CONCLUSIONS

Non-standard noise barriers for optimal far-field shielding have been investigated. A purely geometrical solution is not as prone to deterioration as absorbent barriers that tend to change their acoustic properties with time. The results of a two-dimensional BEM analysis for a fork shaped barrier and a fork gradient barrier are encouraging, as they present an increased insertion loss in the shadow zone of the barrier. A combination of a geometrically optimized shape with absorbent materials further increases the barrier performance.

By changing the depth of the channels in the fork shape barrier, the barrier can be tuned to further improve the barrier performance. If a mechanism is included into the barrier so that the height of the channels can be changed, this gives the possibility to change the spectral profile of the insertion loss. This could be useful in building sustainable barriers, in view of expected but not yet quantifiable shifts of the traffic noise spectrum in the future, as for example due to the increasing number of e-cars on the main transportation routes.

ACKNOWLEDGEMENT

The authors acknowledge Vicente Cutanda Henríquez for his help and support in working with OpenBEM.

REFERENCES

[1] J.P. Clairbois, J. Beaumont, M. Garai and G. Schupp, "A new in situ method for the acoustic performance of road traffic noise reducing devices", *Journal of the Acoustical Society of America* **103**, 2801 (1998)

[2] European Standard EN 1793-1 *Road traffic noise reducing devices – Test method for determining the acoustic performance, Part 1: Intrinsic characteristics of sound absorption*, 1997

[3] European Standard EN 1793-2 *Road traffic noise reducing devices – Test method for determining the acoustic performance, Part 2: Intrinsic characteristics of airborne sound insulation*, 1997

[4] R. Seznec, "Diffraction of sound around barriers: use of boundary elements technique". *Journal of Sound and Vibration* **73**, 195-209 (1980)

[5] K. Fujiwara, D. Hothersall and C. Kim, Noise barriers with reactive surfaces, *Applied Acoustics* **53**, 255-272 (1997)

[6] T. Ishizuka and K. Fujiwara, "Performance of noise barriers with various edge shapes and acoustical conditions", *Applied Acoustics* **65**, 125-141 (2004)

[7] S. Gasparoni, M. Haider, M. Conter, R. Wehr and S. Breuss, "BEM simulations of noise barriers", *Proceedings of the 39th International Congress on Noise Control Engineering, Inter-Noise 2010*, Lisbon, Portugal, 13-16 June 2010

[8] S. Gasparoni, M. Haider, M. Conter and R. Wehr, BEM simulations of diffraction-optimized noise barriers, *Boundary Elements and other Mesh reduction methods*, XXXIII, **52**, WIT press, New Forest, 2011

[9] M. Baulac, J. Defrance and P. Jean, "Optimisation with genetic algorithm of the acoustic performance of T-shaped noise barriers with a reactive top surface", *Applied Acoustics* **69**, 332–342 (2008)

[10] S. Gasparoni, M. Haider, M. Conter and R. Wehr, "BEM simulations of non-standard noise barriers with a focus on tunability", *AIP Conference Proceedings*, **1389**, pp. 460-464, Halkidiki, Greece, 19-25 September 2011

[11] V.C. Henríquez and P.M. Juhl, "OpenBEM – An open source Boundary Element Method software in Acoustics", *Proceedings of the 39th International Congress on Noise Control Engineering, Inter-Noise 2010*, Lisbon, Portugal, 13-16 June 2010

[12] M.E. Delany and E.N. Bazley, "Acoustical properties of fibrous absorbent materials", *Applied Acoustics* **3**, 105-116 (1970)

[13] J.-F. Allard and Y. Champoux, "New empirical equations for sound propagation in rigid frame fibrous materials", *Journal of the Acoustical Society of America* **91**, 3346-3353 (1992)

REPRODUCIBILITY AND APPLICABILITY OF ENSEMBLE AVERAGED SURFACE NORMAL IMPEDANCE OF MATERIALS USING AN *IN-SITU* TECHNIQUE

Nazli Bin Che Din[1*], Toru Otsuru[2], Reiji Tomiku[2], Noriko Okamoto[3] and Kusno Asniawaty[4]

[1]Faculty of Built Environment, University of Malaya, 50603 Kuala Lumpur, Malaysia

[2]Department of Architecture and Mechatronics, Oita University, 700 Dannoharu, Oita 870-1192, Japan

[3]Department of Architecture, Ariake National College of Technology, 150 Higashihagio-Machi, Omuta Fukuoka 836-8585, Japan

[4]Department of Architecture, Hasanuddin University, Makassar, Jl. Perintis Kemerdekaan Km.10, 90245 South Sulawesi, Indonesia

[*]nazlichedin@um.edu.my

This paper investigates by experiment the absorption characteristics of several materials associated with the proposed acoustics impedance method using the combination of sound pressure and particle velocity sensors in various sound fields. This method is based on the concept of "ensemble averaged" surface normal impedance that extends the usage of obtained values to various applications such as architectural acoustics and computational simulations. The measurement technique itself is an improvement of the method using two-microphone technique and diffused ambient noise. A series of measurement in different sound fields was conducted to expand the relevant applicability of *in-situ* measurement using pu-sensor. The first part of the experiment aimed to confirm the reproducibility of the measured values of the method. Here, comparative round robin measurements in four reverberation rooms were conducted. The general tendencies and discrepancies of ten materials in the various reverberation rooms are discussed. In the second stage, the method was applied with four types of selected materials to examine material's absorption characteristics at different sound fields such as in architectural spaces. This paper revealed the reliability, applicability and robustness of the method despite the room's geometrical differences throughout the *in-situ* measurement.

INTRODUCTION

There are two well-known methods of laboratory measurement of absorption which have been described as international standards [1]−[3] in providing important information about the test material (i.e. reverberation room and tube method). A number of studies [4]−[9] have been conducted in order to check the effectiveness of the standards. In Europe, a set of round robin test was carried out in the past decade to investigate the accuracy of the measurement of the reverberant sound absorption coefficient [4]. Nevertheless, there still remain unresolved issues e.g. diffusivity in the reverberation room, edge effect of specimen, etc. Another series of round robin tests were carried out in Japan [5]−[6] to look into some of the aforementioned problems. Differences of measurement values due to the room volume, measurement instruments, etc. were kept central to the investigation to maintain a satisfactory level of accuracy.

Meanwhile, the accuracy of the performance of the tube method has also been reported [7]−[9]. Horoshenkov et al. [9] presented the dispersion of measured normal incident results of inter laboratory reproducibility experiments of the acoustical properties in Europe and North America. They highlighted the importance of the boundary conditions, homogeneity of the porous material structure and stability of the adopted signal processing method. However, similar mounting conditions are difficult to reproduce and this may affect the measured results.

In our previous paper [10], the theoretical development and concept of ensemble averaged surface normal impedance at random incidences were given. Several boundary element method (BEM) simulations of glass wool both at normal and at random incidences showed that ensemble averaging decreases the interference effect caused mainly by the specimen's edges. The BEM simulation with anisotropy consideration [11]−[13] is compared with the measurement result to give an appropriate expected value of the surface normal impedance of the glass wool. Also, a series of measurements by proposed method using pu-sensor (Microflown, [14], [15]) is presented to investigate the considerable geometrical configurations e.g. the sensor height, and the sample size, in measuring the acoustics behavior of absorptive material [16].

Method reliability is one of the factors that needs to be taken into consideration while aiming toward an efficient *in-situ*

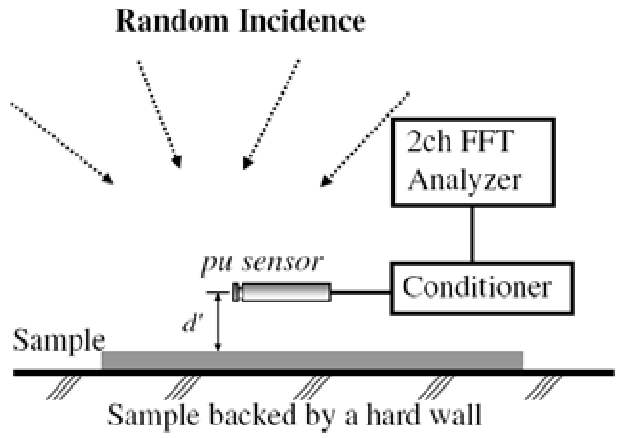


Figure 1. Schematic diagram of the measurement setup with a pu sensor

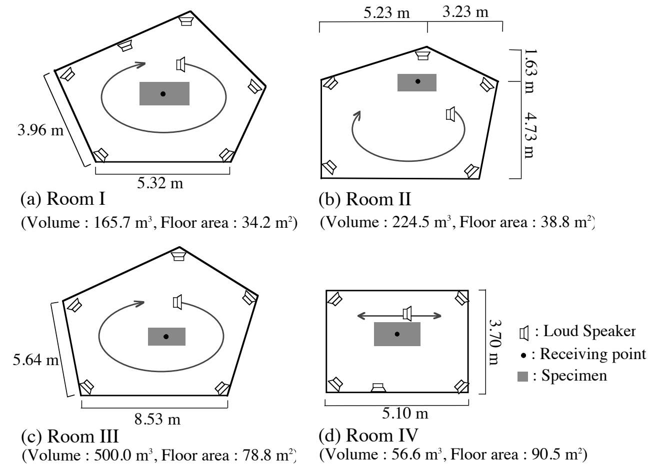


Figure 2. Location of sound sources, receiving points and specimens: (a) Room I; (b) Room II; (c) Room III; (d) Room IV

measurement technique. There is a lack of data on the method reliability of the method that uses pu-sensor. The objectives of this paper are: (i) to investigate whether the proposed method can offer plausible agreements of reproducibility for selected materials between different reverberation rooms; and (ii) to expand the relevant applicability of *in-situ* measurement using pu-sensor outside laboratory rooms.

SHORT DESCRIPTION OF THE METHOD

Ensemble Averaged Surface Normal Impedance

In this section, we summarize the explanation given in our previous papers [10], [16]. The authors proposed an ensemble averaged impedance, $\langle Z_n \rangle$, as

$$\langle Z_n \rangle = \frac{\langle p_{\mathrm{surf}} \rangle}{\langle u_{n,\mathrm{surf}} \rangle} \tag{1}$$

where $\langle \cdot \rangle$ denotes the ensemble average.

In a practical measurement using digital techniques with a fast Fourier transform (FFT), the process of ensemble averaging is assumed that $p(t,\theta_i)$ and $u_n(t,\theta_i)$ denote, respectively, the sound pressure and particle velocity normal to the surface of sound incidence angle, θ_i close to the surface at time t.

The ensemble averaging of the sound pressures and particle velocities $\langle p(t) \rangle$ and $\langle u_n(t) \rangle$, respectively, in incident event number, M, can be written as

$$\langle p(t) \rangle = \frac{1}{M}\sum_{i=1}^{M} p(t,\theta_i) \times W(t), \tag{2a}$$

$$\langle u_n(t) \rangle = \frac{1}{M}\sum_{i=1}^{M} u_n(t,\theta_i) \times W(t), \tag{2b}$$

where $W(t)$ is a window function with length, T_w. Applying the Fourier transform, we have

$$\langle p(\omega) \rangle = \int_{-T_w/2}^{T_w/2} \langle p(t) \rangle \, e^{-j\omega t} dt, \tag{3a}$$

$$\langle u_n(\omega) \rangle = \int_{-T_w/2}^{T_w/2} \langle u_n(t) \rangle \, e^{-j\omega t} dt, \tag{3b}$$

where ω is the angular frequency of sound and j denotes the imaginary number $\sqrt{-1}$.

Next, the transfer function $H_{up}(\omega)$ between the velocity output to the pressure response for averaged sound pressures and averaged particle velocities in the frequency domain becomes $H_{up}(\omega) = \langle p(\omega) \rangle / \langle u_n(\omega) \rangle$.

Here, with the linear averaging number, N, on FFT, we express the ensemble averaged impedance as

$$\langle Z_n \rangle = \frac{1}{N}\sum^{N} H_{up}(\omega). \tag{4}$$

The corresponding absorption coefficient, $\langle \alpha \rangle$, is given by:

$$\langle \alpha \rangle = 1 - \left| \frac{\langle Z_n \rangle - \rho c}{\langle Z_n \rangle + \rho c} \right|^2. \tag{5}$$

Here, ρ and c are the density of air and the speed of sound, respectively.

Measurement outline

Figure 1 shows schematics of the apparatus used in the measurement. The pu-sensor was located at the middle of the specimen with the height of 10 mm above from specimen surface ($d' = 10$ mm) to measure p and u_n. The pu-sensor was calibrated using an acoustic tube with 10 cm diameter for the usage within the frequency from 100 Hz to 1500 Hz. The resolution of the two-channel FFT (RION SA-78) unit was set to 1.25 Hz and a Hanning window, $W(t)$, of duration 0.8 s with the averaging number, N, of 150 was employed to measure the transfer function.

In the original method [17], the sound source was intended for use only with diffuse ambient noise that exists around the specimen to be measured. However, in the case where the noise is insufficient, a supplemental noise source(s) can be added to improve the result. Generally, the loudspeakers were employed

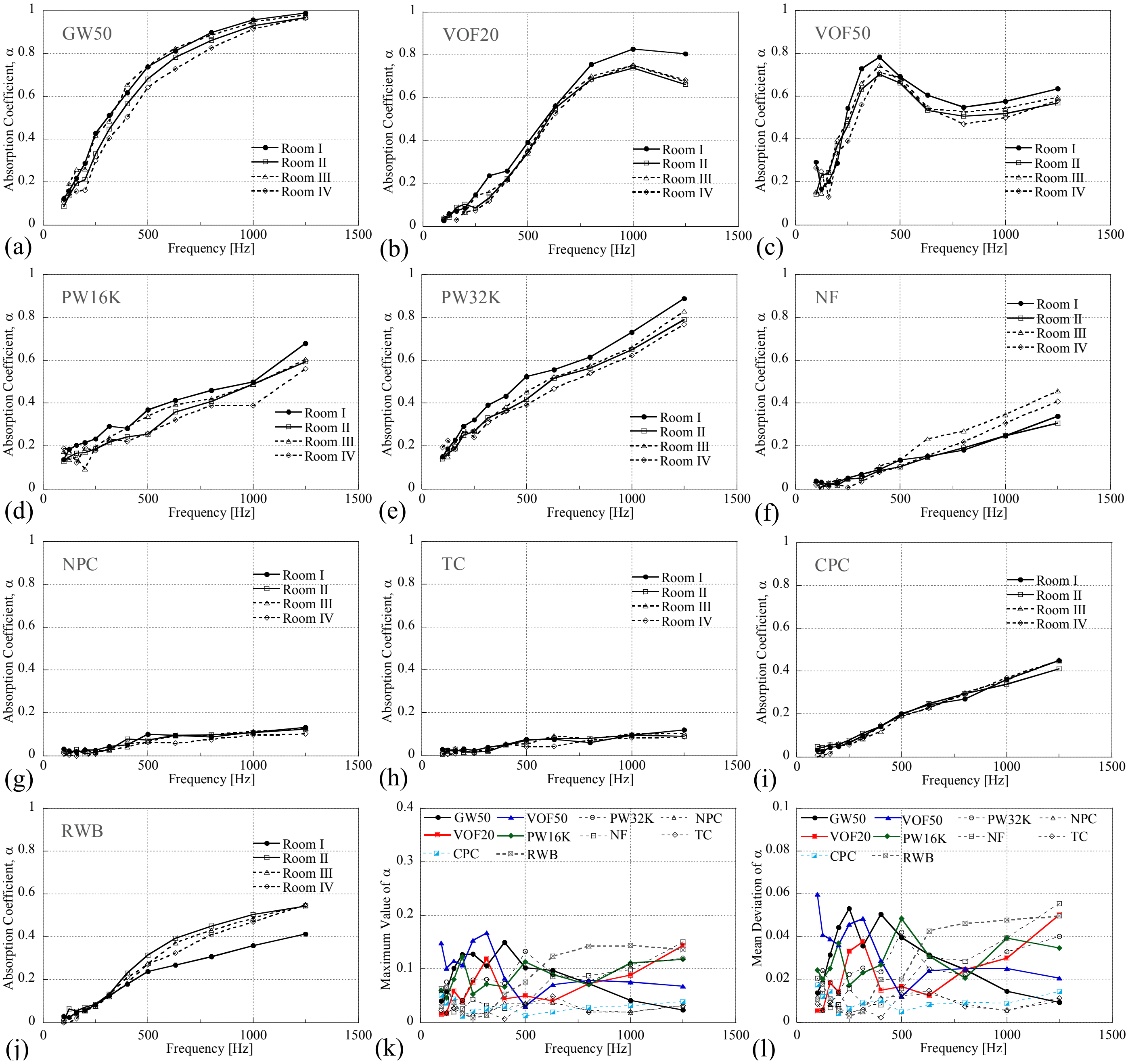


Figure 3. Comparisons of (a) - (j) measured absorption coefficients of ten types of specimens obtained by proposed method in four reverberation rooms; (k) maximum differences value of absorption coefficients; (l) mean deviation of absorption coefficients

to radiate incoherent pink noises and focused to examine the 100 Hz to 1500 Hz in range.

So as to provide a compact presentation and ensure convenience for the reader, all the results are averaged in 1/3 octave band and presented as absorption coefficients base.

METHOD REPRODUCIBILITY

The main purpose of the measurements in this section is to investigate whether the proposed method can offer reproducibility of measured absorption characteristics on various materials in different reverberation rooms. A series of measurement is conducted in four reverberation rooms with kind permission from the participating institutes in Japan as depicted in Fig. 2. Suspended diffuser panels are installed in Room II and the reverberation time in Room IV is compensated as suggested in ISO 354 and JIS A 1409. Figure 2 also indicates the location of sound sources, receiving point and specimen under test. Table 1 shows the details of dimension and volume of each type of reverberation rooms.

Five fixed loudspeakers are employed to radiate incoherent pink noises except in Room I where six fixed loudspeakers are employed. Also, an additional movable loudspeaker are

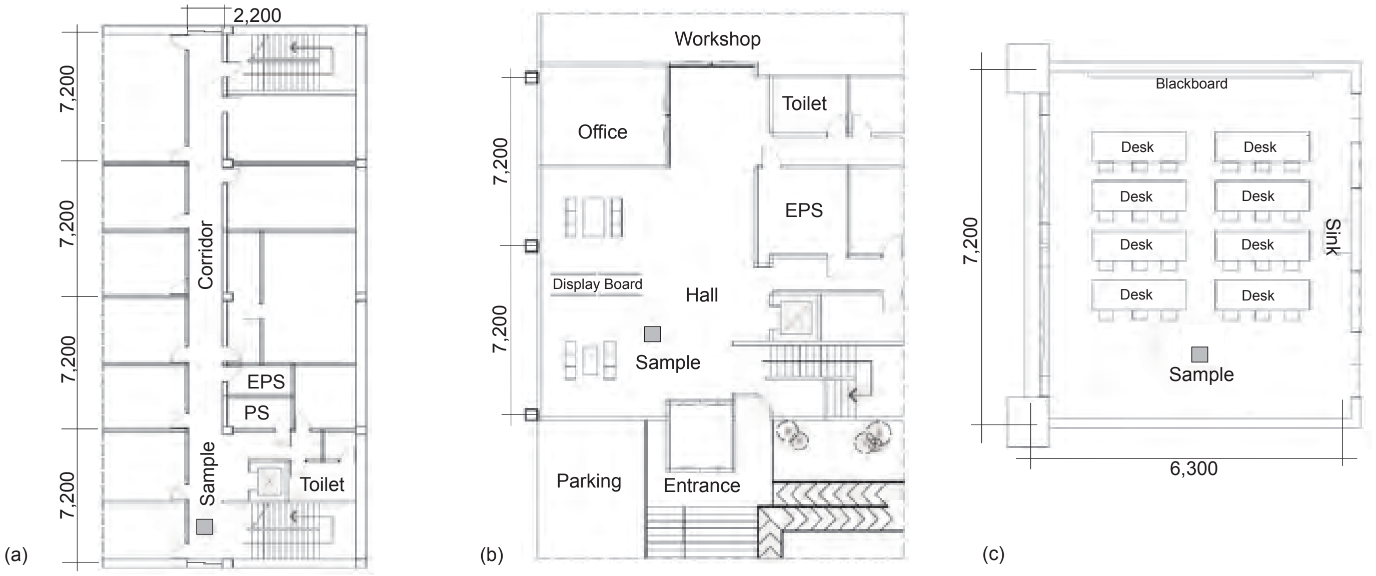


Figure 4. Plan views of furniture layouts and material locations: (a) a corridor; (b) an entrance hall; (c) a seminar room

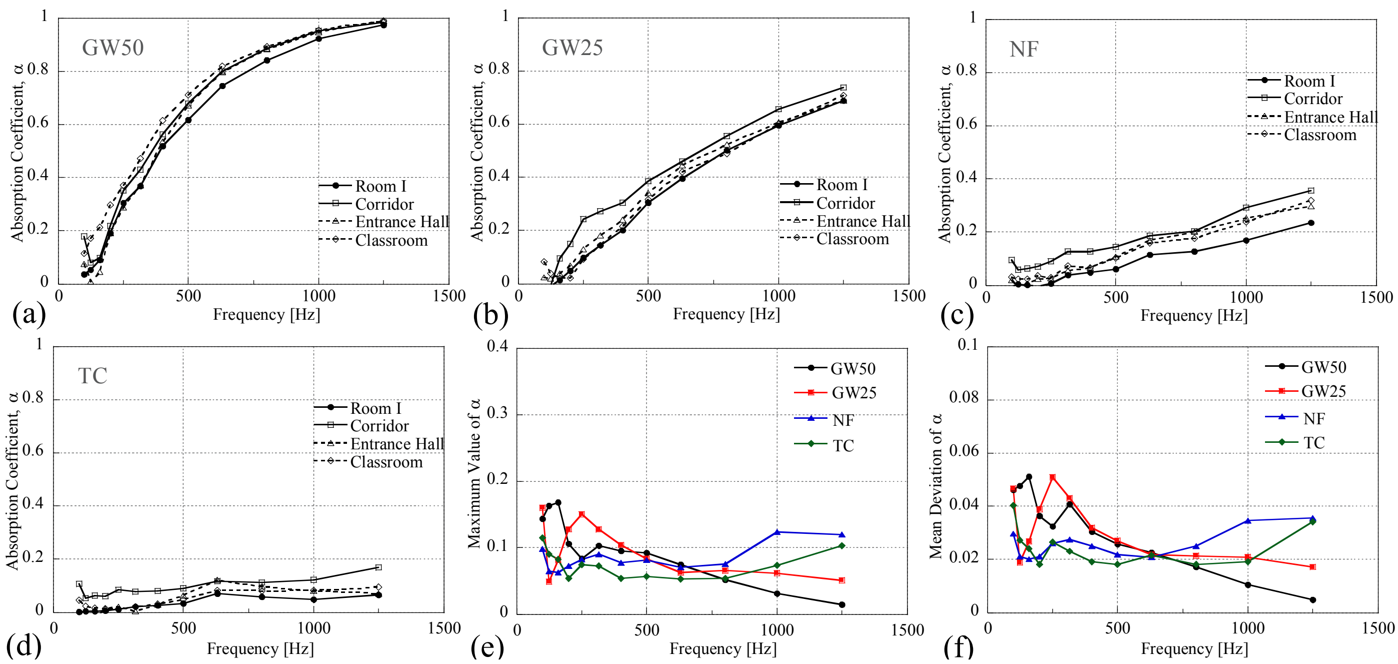


Figure 5. Comparisons of; (a) - (d) measured absorption coefficients of four types of specimens obtained by proposed method in the corridor, the entrance hall and the seminar room; (e) maximum differences value of absorption coefficients; (f) mean deviation of absorption coefficients

used in all reverberation rooms. Ten types of materials with specific dimensions are investigated as listed in Table 2. All of specimens are laid on a 0.02 m acrylic plate. The resolution of FFT settings is set to be 2.5 Hz in all reverberation rooms except in Room I where the resolution is set to be 1.25 Hz.

Figures 3(a) - (j) shows the comparisons of measured absorption coefficients of each type of specimens in four types of reverberation rooms. The maximum differences values and the mean deviation of absorption coefficients for each specimens also provided by Figs. 3(k) and 3(l), respectively. In general, the measured absorption coefficients show the same basic tendency for their respective specimens with some differences in value relatively independent on the frequency. From these results, the good agreements for the measured absorption coefficients obtained in the four reverberation rooms are observed in Figs. 3(g) - (i), whereby the maximum dispersion in the measured absorption coefficients is 0.05 for CPC.

Furthermore, the other specimens can be considered having fair agreements based on the maximum dispersion being

Table 1. Dimensions of the reverberation rooms

Room	Geometry	Volume [m^3]	Floor Area [m^2]
I	irregular	165.7	34.2
II	irregular	224.5	38.8
III	irregular	500.0	78.8
IV	regular	56.6	90.5

Table 2. Materials to be measured

Material	Abbrev.	Size [mm^3]
Glass wool (32kg/m^3)	GW50	1820x910x50
Flexible urethane foam	VOF20	1820x910x20
Flexible urethane foam	VOF50	1820x910x50
Polyester nonwoven (16kg/m^3)	PW16K	1820x910x50
Polyester nonwoven (32kg/m^3)	PW32K	1800x900x50
Needlefelt	NF	1800x900x10
Needle punched carpet	NPC	1800x900x3
Tile Carpet	TC	(500x500x6)x6.5
Cut pile carpet	CPC	1820x910x15
Rock wool board	RWB	(600x300x12)x9

below 0.17. Even though the high dispersion values are observed in the measured absorption coefficients, they can be considered as acceptable discrepancies based on comparison with other results related to acoustics impedances round robin tests [5],[6],[9]. In Fig. 3(l), on the whole, the maximum mean deviation of absorption coefficients is lower than 0.06. At this stage, it can be concluded that the reproducibility of the proposed method is satisfactory, and that the method gives appropriate absorption coefficients despite the geometrical differences of the reverberation rooms.

METHOD APPLICABILITY

To investigate the general applicability of the proposed method, a series of measurements of the four materials has been carried out in three other environments of architectural spaces [a corridor, an entrance hall and a seminar room]. Plan views of furniture layouts and material locations in the field measurements are shown in Fig. 4. Specimens to be investigated are GW50, NF, TC and additional of glass wool 25 mm thick (GW25). All the specimens are laid on a 0.02 m acrylic plate and have the same square areas with 0.6 x 0.6 m^2 except for TC where the area is 0.5 x 0.5 m^2. The specimen's sizes are not exactly identical to that of the investigation in previous section, but sufficient validity can be expected for the discussion as described in Ref. 11. Six portable sound speakers with incoherent pink noises are employed and manual-moved randomly by three people to realize the random noises condition because of insufficient noises in all environments conditions. For a comprehensive comparison, the measurements of similar specimens are conducted in Room I using six fixed loudspeakers to radiate incoherent pink noises.

Figures 5(a) - (d) present the combined results measured in three other environments for all the specimens GW50, GW25, NF and TC, respectively. All the measured absorption coefficients in three other environments are compared with the measured absorption coefficients obtained in Room I. Same as previous section, the maximum differences and mean deviation of measured absorption coefficients are provided in Figs. 5(e) - (f), respectively.

The same basic tendencies can be observed for all specimens in Figs. 5(a) - (d) but there are noticeable differences in the dispersion. The result of measured absorption coefficients of Room I is lower than the results measured in three other environments. There can be complementary aspects that can explain this phenomenon: (i) the result of sound reflections coming from the specimen's surrounding; (ii) the dissimilarity of measurement setting of sound sources where the fixed loudspeakers are employed in Room I. Moreover, all specimens can be considered as having fair agreements based on the maximum dispersion being below 0.17 and maximum mean deviation being lower than 0.06, similar as found in the previous section. The dispersion of measured absorption coefficients can be considered plausible agreements to support the applicability of the proposed method in various sound fields.

CONCLUSIONS

In this study, an extensive measurement of "ensemble averaged" surface normal impedance at random incidences in different sound fields using the pu-sensor has been performed onto various selected materials. A series of measurement in different types of reverberation rooms revealed that the material's absorption coefficients yield relatively small measured maximum mean deviation to confirm the reproducibility of the method. The *in-situ* measurements using pu-sensor offers good applicability of the method to apply onto various practical measurements. Further numerical and experimental investigations are now being pursued intensively.

ACKNOWLEDGEMENTS

The authors would like to thank to Mr. F. Kutsukake, Mr. M. Murakami and Ms. R. Matsumoto for their continuous contribution to this research. The authors are grateful to Prof. H. Shiokawa and Dr. K. Hoshi (Nihon University) and Mr. A. Hiramitsu (Building Research Institute) for their kind permission to support this work. This research is partially supported by a Japan Society for the Promotion of Science (JSPS), Grant-in-Aid for Exploratory Research, 19656143, 2007–9, and was conducted as one Research Project (A) of the Venture Business Laboratory in Oita University.

REFERENCES

[1] International Organization for Standardization ISO 354:2003, *Acoustics - Measurement of sound absorption in a reverberation room*

[2] International Organization for Standardization ISO 10534-1:1996, *Acoustics - Determination of sound absorption coefficient and impedance in impedance tubes - Part 1: Method using standing wave ratio*

[3] International Organization for Standardization ISO 10534-2:1998, *Acoustics - Determination of sound absorption coefficient and impedance in impedance tubes - Part 2: Transfer-function method*

[4] C.W. Kosten, "International comparison measurements in the reverberation room", *Acustica* **10**, 400-411 (1960)

[5] Y. Makita, M. Koyasu, M. Nagata and S. Kimura, "Investigations into the precision of measurement of sound absorption coefficients in a reverberation room (I)", *Journal of the Acoustical Society of Japan* **24**, 381-392 (1968)

[6] Y. Makita, M. Koyasu, M. Nagata and S. Kimura, "Investigations into the precision of measurement of sound absorption coefficients in a reverberation room (II)" *Journal of the Acoustical Society of Japan* **24**, 393-402 (1968)

[7] A. Cummings, "Impedance tube measurements on porous media: The effects of air-gaps around the sample", *Journal of Sound and Vibration* **151**, 63-75 (1991)

[8] T. Iwase and Y. Izumi, "A new sound tube measuring method for propagation constant in porous material - method without any air space at the back of test material", *Journal of the Acoustical Society of Japan* **52**, 411-419 (1996)

[9] K.V. Horoshenkov, A. Khan, F.X. Bécot, L. Jaouen, F. Sgard, A. Renault, N. Amirouche, F. Pompoli, N. Prodi, P. Bonfiglio, G. Pispola, F. Asdrubali, J. Hübelt, N. Atalla, C.K. Amédin, W. Lauriks and L. Boeckx, "Reproducibility experiments on measuring acoustical properties of rigid-frame porous media (round-robin tests)", *Journal of the Acoustical Society of America* **122**, 345-353 (2007)

[10] T. Otsuru, R. Tomiku, N.B.C. Din, N. Okamoto and M. Murakami, "Ensemble averaged surface normal impedance of material using an *in-situ* technique: Preliminary study using boundary element method", *Journal of the Acoustical Society of America* **125**, 3784-3791 (2009)

[11] J.F. Allard, R. Bourdier and A. L'Espérance, "Anisotropy effect in glass wool on normal impedance in oblique incidence", *Journal of Sound and Vibration* **114**, 233-238 (1987)

[12] J.S. Pyett, "The acoustic impedance of a porous layer at oblique incidence", *Acustica* **3**, 375-382 (1953)

[13] M.E. Delany and E.N. Bazley, "Acoustical properties of fibrous absorbent materials", *Applied Acoustics* **3**, 105-116 (1970)

[14] H.-E. de Bree, P. Leussink, T. Korthorst, H. Jansen, T.S.J. Lammerink and M. Elwenspoek, "The μ-flown: A novel device measuring acoustic flows", *Sensors and Actuators A* **54**, 552-557 (1996)

[15] H.-E. de Bree, "The Microflown: An acoustic particle velocity sensor", *Acoustics Australia* **31**, 91-94 (2003)

[16] N.B.C. Din, T. Otsuru, R. Tomiku, N. Okamoto and K. Asniawaty, "Measurement method with a pressure-velocity sensor for measuring surface normal impedance of materials using ensemble averaging: Comparison with other methods and its geometrical configuration", *Acoustical Science and Technology* **33**, 86-95 (2012)

[17] Y. Takahashi, T. Otsuru and R. Tomiku, "*In situ* measurements of surface impedance and absorption coefficients of porous materials using two microphones and ambient noise", *Applied Acoustics* **66**, 845-865 (2005)

ACOUSTIC CORRECTION USING GREEN MATERIAL IN CLASSROOMS LOCATED IN HISTORICAL BUILDINGS

Gino Iannace, Amelia Trematerra, Patrizia Trematerra
Department of Architecture and Industrial Design, Second University of Naples, Borgo San Lorenzo Aversa, Italy
gino.iannace@unina2.it

The acoustic correction inside classrooms located in historical buildings using absorbent panels is difficult for aesthetic reasons. Furthermore, architectural restrictions are often imposed to preserve the historical heritage. The acoustic measurements inside the classrooms show high reverberation time values, which imply an adverse environment for speech reception. In this paper the reverberation time in classrooms located in historical buildings was reduced by installing removable sound absorbent panels. The panels were made with "green material". The absorbent material was obtained by crushing giant reeds of sweet water, a plant which grows quickly in wetlands. The crushed material was then put in jute sachets, installed in the wooden frames and covered with different colours jute cloth for aesthetics. Acoustic measurements were made in the classrooms with smooth plaster walls, without students. A virtual model of the classroom was drawn with 3D CAD. The surface area covered with green material absorbent panels was evaluated by the software Odeon. After the installation of the absorbent panels, comparisons between the virtual classroom acoustic properties and the real classroom acoustic properties were made to validate the effect of the green absorption panels.

INTRODUCTION

Due to aesthetic and historic reasons, it is difficult to install sound absorbent panels for acoustic correction in classrooms located in historical buildings. Usually the dimensions of such classrooms (volume and area) are large so the reverberation time is high (over 2 seconds) [1]. Furthermore, classrooms located in historical buildings are not regular in shape and the ceilings are not plane. To improve the acoustic characteristics of classrooms, sound absorbing materials are usually suspended on the walls; however in historical buildings it is not possible to use fixed structures [2].

Absorbent panels are generally made with traditional sound absorbent materials such as glass wool, rock wool, polyester or polyurethane foam. In this paper a sustainable "green material" was used as the sound absorbent material. This has the advantage that at the end of its useful life, it can be disposed of without difficulty and without damage to the environment. In addition, green materials store carbon dioxide during their growth.

In order to evaluate the classroom surface area to be covered with the green material absorbent panels, this study used the architectural acoustic Odeon software in conjunction with a classroom virtual model. Classrooms of the Faculty of Architecture of the Second University of Naples (SUN) were selected for the study. The Faculty of Architecture is located in an ancient building in the city of Aversa near Caserta (Italy). The building was built in the 10th century as a Benedictine monastery. It was then expanded in the 15th century and later converted into a school and finally into a University in 1990. Figure 1 shows the cloister on two levels with arches and columns. The classrooms were irregularly shaped, with vaulted ceilings and smooth plaster walls. The acoustic parameters measured in each classroom were: Reverberation Time (T_{30}), Early Decay Time (EDT), Definition (D_{50}) and the Speech Transmission Index (STI) [3].

Figure 1. The cloister on two levels with arches and columns

ACOUSTIC MEASUREMENTS

The acoustic measurements were carried out in seven classrooms, using an omnidirectional spherical source fed with a Maximum Length Sequence (MLS) signal. The impulse responses were detected with a measurement microphone GRAS 40 AR endowed with the preamplifier 01 dB PRE 12 H connected with a laptop PC through the interface 01 dB Symphonie. The sound source was placed in each classroom at the teacher's position (height 1.6 m). The microphones measurements were set in different points at a typical ear height of 1.2 m, to obtain average values of the classroom acoustic parameters. The acoustic parameters were measured according to ISO 3382 [4]. Figure 2 shows the omnidirectional sound source in the classroom at the teacher's position, and the measurement microphone between the tables. During the acoustic measurements the background noise was lower than 50 dBA, the classrooms were empty without students and the furniture consisted of hard chairs and rows of hard tables. The classroom had smooth walls, wooden doors and glass windows [5,6]. Table 1 shows the average geometrical dimensions for the seven classrooms. For these same classrooms, Table 2 shows the STI average values measured, Table 3 shows T30 average values measured and Table 4 shows D50 average values measured. All the acoustic parameters considered show that in the classrooms the quality of speech reception is poor.

CASE STUDY

Classroom T4, a room with smooth plaster walls, was chosen as the case study. This plan is 9.0 m long and 5.0 m large. The average height is about 5.0 m and the volume is 240 m^3 (Figure 3). In the classroom there are thirty hard chairs and six rows of hard tables; the students' seating area is 5.30 m × 2.50 m. When the classroom is empty (without students), the measured values of STI, T_{30} and D_{50} are respectively reported in Tables 2, 3 and 4. The acoustic parameters measured indicate that in this classroom, the speech reception was not good. Furthermore tests administered to students have confirmed that the speech intelligibility was poor.

Figure 2. Acoustic measurements in the classroom

Table 1. Classrooms average dimensions

Classroom	T4	T5	P3	P4	S2	S3	T1
Volume, m^3	240	2517	416	1200	626	1850	202
Average height, m	5.0	12.1	5.4	5.5	4.6	7.2	4.5
Base area, m^2	50	208	77	217	136	257	45

Table 2. STI average values measured

Classroom	T4	T5	P3	P4	S2	S3	T1
STI, measured	0.34	0.38	0.48	0.47	0.47	0.46	0.47

Table 3. T_{30} (s) average values measured

Frequency, Hz	125	250	500	1k	2k	4k
T4	3.43	2.44	2.04	1.76	1.70	1.41
T5	4.67	5.41	4.23	4.0	3.15	2.14
P3	2.83	2.67	2.18	2.24	1.81	1.56
P4	3.22	2.88	2.77	2.44	2.22	1.81
S2	2.79	2.78	2.69	2.63	2.25	1.74
S3	3.04	2.73	2.72	2.66	2.24	2.0
T1	3.49	3.43	2.77	2.37	2.21	2.01

Table 4. D_{50} average values measured

Frequency, Hz	125	250	500	1k	2k	4k
T4	0.23	0.16	0.22	0.30	0.27	0.33
T5	0.22	0.16	0.15	0.23	0.29 5	0.50
P3	0.23	0.30	0.30	0.31	0.34	0.40
P4	0.23	0.27	0.27	0.31	0.38	0.44
S2	0.28	0.30	0.23	0.26	0.33	0.41
S3	0.30	0.34	0.31	0.27	0.25	0.34
T1	0.22	0.22	0.28	0.32	0.30	0.35

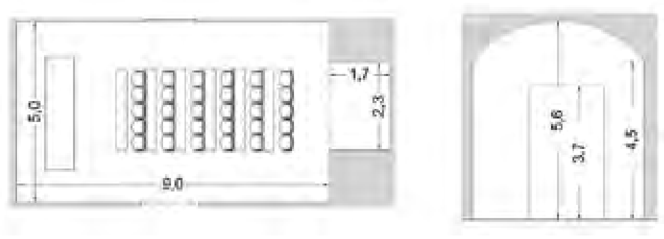


Figure 3. Classroom plan and section dimensions (in metres)

ACOUSTIC PROPERTIES OF THE MATERIAL

For the classroom acoustic correction, sustainable materials termed green materials were chosen [7-11]. These materials are generally employed for energy saving purposes (heat insulating materials). Recently they have also been applied in architectural acoustics to replace the traditional sound absorbent materials (glass wool, rock wool, polyester, polyurethane foam, etc). Sustainable materials have the advantage that they can be disposed of without difficulty and without damage the environment at the end of their useful life. The sustainable material used in this study is a giant reed of sweet water (arundo donax). It is a material commonly available in country sides near rivers, lakes and wetlands. This plant grows very quickly, usually reaching 6 m in height and a diameter of 2-3 cm. The giant reeds were cut, dried and then crushed. They were then shredded into flakes of small size, with average dimensions of 40 mm length, 10 mm width and 3.0 mm thickness (Figure 4). The loose grains obtained were placed in sample sacks made of jute which were tested by the measurement system.

To assess the material acoustic properties, the absorption coefficient at normal incidence was measured with a Kundt's tube, in accordance with ISO 10534-2: 1998 [12]. The Kundt's tube has an inner diameter of 100 mm, length of 560 mm. With distance between the two measuring microphones of 50 mm the absorption coefficient measurement is accurate in the range frequency of 200 Hz – 2 kHz. Using the distance between the two measuring microphones of 100 mm, the absorption coefficient measurement is accurate in the frequency range of 100 Hz – 1 kHz.

Figure 5 presents the average value of the absorption coefficient measured at normal incidence in the frequency range 125 Hz – 2 kHz with Kundt's tube [13]. This average value is obtained from measurements with four different specimens (thickness 40 mm). The material has a good value of the absorption coefficient at the medium frequencies. The loose materials were inserted in the tube measurement and stopped with a net metal, so the Kundt's tube was in a horizontal position. The measured absorption coefficient values are similar to those of limestone chips with the same thickness [14]. The loose material was then inserted in jute sacks in order to obtain a layer of sound absorbent porous material and mounted in wooden frames covered with a jute burlap (Figure 6). Since the jute burlap has a large mesh and low air resistance, it can be considered as an acoustically transparent material.

Figure 4. Loose grains average dimensions

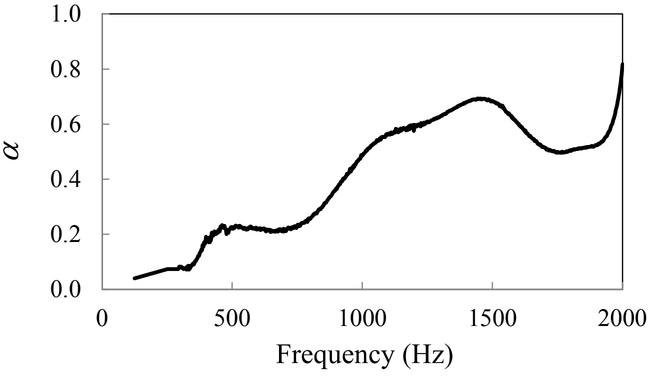


Figure 5. Average values for the absorption coefficient α measured at normal incidence in the frequency range of 125 Hz – 2 kHz

ODEON VIRTUAL MODEL

A software used for architectural acoustics, Odeon, was used to evaluate the classroom surface area to be covered with absorbent material in order to obtain an acoustic correction. The Odeon software imports a virtual model drawn by 3D CAD [15]. Figure 7 shows the 3D virtual classroom model with the virtual omnidirectional sound source in the teacher position, and the virtual receivers in the student positions. Figure 8 shows the classroom render with the absorbent panels insertion. The Odeon virtual model had 88 corners, 35 surfaces in the room and a total surface area of 254 m^2.

The first operation is the acoustic model calibration which consists of setting the absorbent coefficient values for all virtual model surfaces and setting the scattering coefficients. The scattering coefficient s does not depend on frequency, but on the surface geometrical properties; so the desks and chairs were simulated as flat planes, with a scattering coefficient s=0.5 for the unoccupied condition. The calibration operation is stopped when at each octave band frequencies (125 Hz – 4 kHz) the value of reverberation time (T_{30}) calculated is equal to reverberation time (T_{30}) measured.

Figures 9, 10 and 11 respectively show the comparison between the acoustic parameters values T_{30}, EDT and D_{50} both measured and calculated using the Odeon software, when the classroom with smooth walls is empty (without students). After the calibration operation, the acoustic absorbent panels with surface area 11 m^2 with the values of absorbent coefficient given in Figure 5 were inserted in the virtual model. The values of the absorbent coefficient at 4.0 kHz were obtained by extrapolating the measured values. In this configuration, the calculated value for speech transmission index (STI) is 0.45. For the EDT, the difference between measured and calculated values is negligible, while for the STI and the D_{50}, the calculated values are higher than the measured values.

Figure 6. Sound absorbent panels

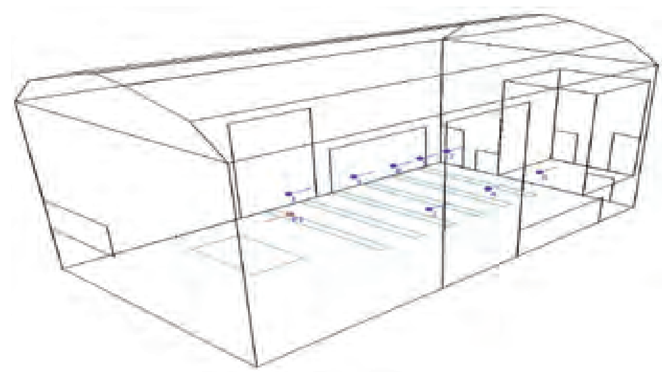


Figure 7. 3D virtual model by Odeon, with the sound source and receiver position

Figure 8. Render of 3D virtual model by Odeon with the absorbent panels insertion on the walls

FULL SCALE ACOUSTIC MEASUREMENTS

Figure 12 shows the green material absorbent panels located in the classroom covering an area of 11 m^2. The absorbent panels are installed in a temporary manner due to aesthetical and historical reasons, and in accordance to Superintendence of historical heritage architectural restriction. During the acoustic measurements, the omnidirectional sound source and the receivers were put in the same initial positions (teacher and students positions). The comparisons between measured and calculated values are presented in Figures 13-15 for reverberation time (T_{30}), early decay time (EDT) and definition (D_{50}), respectively. In this configuration, the calculated value STI = 0.56, while the measured STI is 0.55. The difference between measured and calculated values for the T_{30} and EDT parameters is negligible, while for the STI and D_{50} the calculated values are higher than the measured values.

CONCLUSIONS

This paper demonstrates the possibility to obtain a good acoustic correction inside classrooms using a green material corresponding to the giant reeds, properly shredded and re-assembled into panels. The green material is inserted in jute bags where the jute has a large mesh and a low air resistance and as such is an acoustically transparent material. Using sound absorbing panels of different colours makes this material aesthetically acceptable. The wide availability reduces the cost of production of the panels, and more importantly, the material used is completely recyclable. It was also found that the Odeon software provides a good prediction of the room acoustic correction. The model calibration with the software Odeon was shown to produce acceptable results for the reverberation time T_{30} and the EDT.

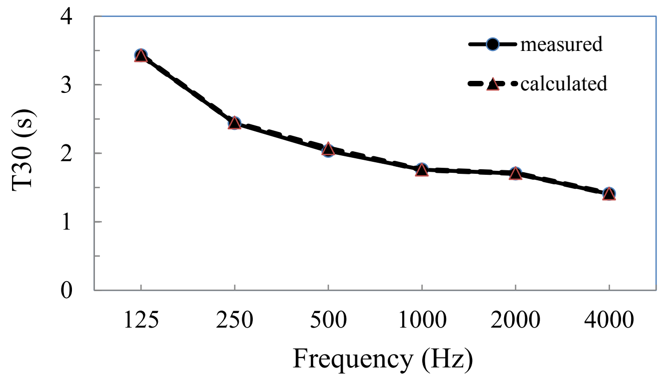


Figure 9. T_{30} measured and calculated values

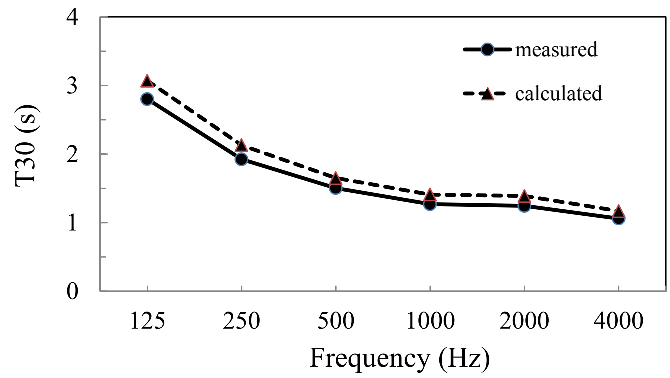


Figure 13. T_{30} measured and calculated values

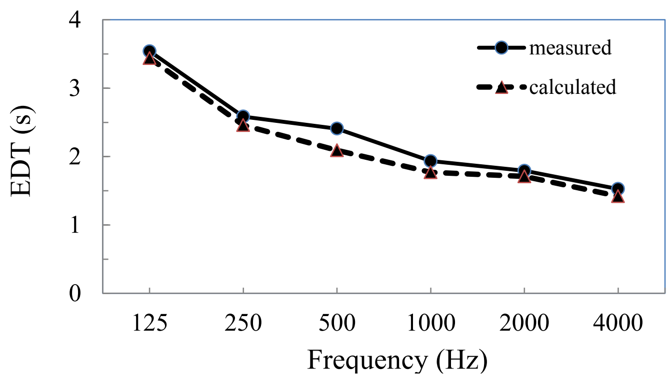


Figure 10. EDT measured and calculated values

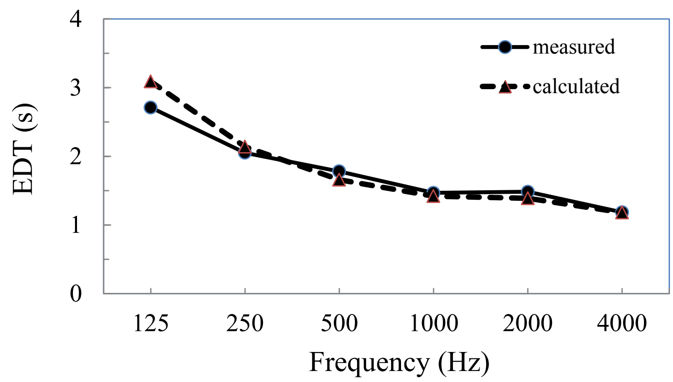


Figure 14. EDT measured and calculated values

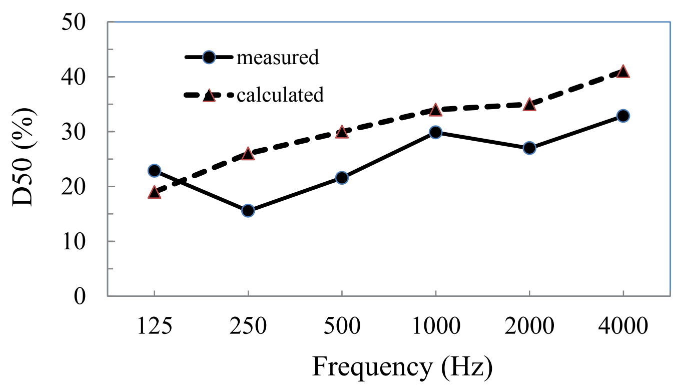


Figure 11. D_{50} measured and calculated values

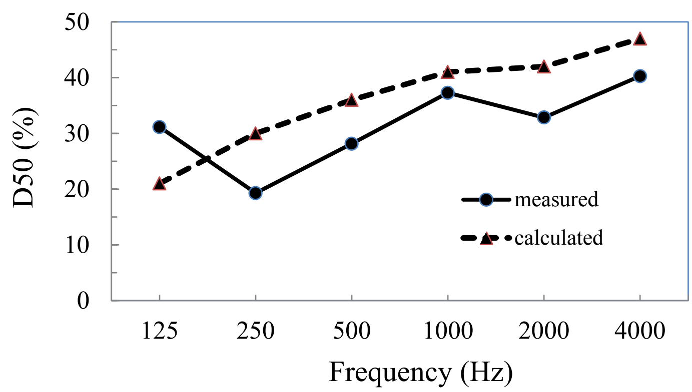


Figure 15. D_{50} measured and calculated values

Figure 12. Sound absorbent panels inserted in a classroom

REFERENCES

[1] G. Iannace, L. Maffei and G. Ciaburro, "Effects of shared noise control activities in two primary schools", *Proceedings of the 39th International Congress on Noise Control Engineering: Inter-Noise 2010*, Lisbon, Portual, 13-16 June 2010

[2] G. Iannace, L. Maffei and M. Masullo, "Are classrooms in historical buildings compatible with good acoustics standards?", *Proceedings of Acoustics'08 Paris*, Paris, France, 30 June - 4 July 2008

[3] G. Iannace, L. Maffei and P. Trematerra, "On the use of "green materials" for the acoustic correction of classrooms", *Proceedings of Euronoise 2012*, Prague, Czech Republic, 10-13 June 2012

[4] International Organization for Standardization ISO 3382-2:2008: *Acoustics – Measurement of room acoustic parameters – Part 2: Reverberation time in ordinary rooms*

[5] M. Hodgson, "Experimental investigation of the acoustical characteristics of university classrooms", *Journal of the Acoustical Society of America* **106**, 1810-1819 (1999)

[6] M. Hodgson and E.M. Nosal, "Effect of noise and occupancy on optimal reverberation times for speech intelligibility in classrooms", *Journal of the Acoustical Society of America* **111**, 931-939 (2002)

[7] R. del Rey, J. Alba, J.P. Arenas and V.J. Sanchis, "An empirical modelling of porous sound absorbing materials made of recycled foam", *Applied Acoustics* **73**, 604-609 (2012)

[8] M. Hosseini Fouladi, Md. Ayub and M. Jailani Mohd Nor, "Analysis of coir fiber acoustical characteristics", *Applied Acoustics* **72**, 35–42 (2011)

[9] P. Glé, E. Gourdon and L. Arnaud, "Acoustical properties of materials made of vegetable particles with several scales of porosity", *Applied Acoustics* **72**, 249-259 (2011)

[10] R. Maderuelo-Sanz, A.V. Nadal-Gisbert, J.E. Crespo-Amorós and F. Parres-García, "A novel sound absorber with recycled fibers coming from end of life tires (ELTs)", *Applied Acoustics* **73**, 402-408 (2012)

[11] S. Fatima and A.R. Mohanty, "Acoustical and fire-retardant properties of jute composite materials", *Applied Acoustics* **72**, 108-114 (2011)

[12] International Organization for Standardization ISO 10534-2: 1998: *Acoustics – Determination of sound absorption coefficient and impedance in impedances tubes – Part 2: Transfer-function method*

[13] G. Iannace and P. Trematerra, "Le caratteristiche acustiche dell'Arundo Donax", *Rivista Italiana di Acustica (Italian Journal of Acoustics)* **37**, January-March 2013 (in Italian)

[14] G. Iannace, C. Ianniello, L. Maffei and R. Romano, "Characteristic impedance and complex wave-number of limestone chips", *Proceedings of Euronoise 2001*, Patras, Greece, 14-17 January 2001

[15] X. Zeng, C.L. Christensen and J.H. Rindel, "Practical methods to define scattering coefficients in a room acoustics computer model", *Applied Acoustics* **67**, 771-786 (2006)

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. Already, over 40 exhibition booths are booked by overseas and local exhibitors. We are also pleased that Ortech Industries are once more Gold sponsors and that CSR Bradford and Pyrotek are Bronze Sponsors. A few more sponsors are in the pipeline, so if you wish to participate as a Sponsor (some Gold, Silver and Bronze opportunities are still available) or in the exhibition, it is suggested that you contact Dr Norm Broner at NBroner@globalskm.com very promptly or for more details, refer to the Congress website www.internoise2014.org

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
Several interesting discussion sessions are being developed for the Congress, including the question "Should wind turbine sounds be regulated similarly to other sources of community noise?" Also under consideration is the multi-disciplinary application of lightweight constructions to buildings and passenger vehicles. Other special sessions will involve active noise control, sound propagation in outdoor and urban situations, maritime noise and people's reaction to noise and vibration. Workplace health and safety issues, vibro-acoustics, buy quiet and modern acoustic materials are a few more of the 100 potential sessions now listed on the Congress website.

The website also gives details of the six exciting distinguished lectures which forms part of the technical program. These include talks on sound visualization and manipulation, aircraft noise, soundscape and the impact of building acoustics on speech comprehension and student achievement.

Abstract submission commences in January using the Congress website while the paper template will also be available on this site. The closing date for abstracts is 10 May, 2014, and this date is firm and will NOT BE EXTENDED. So please ensure your abstracts are submitted early. Registration is also through the Congress website. Don't forget that the early bird registration closes on 25 July 2014 so book early.

More details at www.internoise2014.org

VIBRATION INDUCED DUE TO ACOUSTIC EXCITATION IN DIFFUSE FIELD CONDITIONS

Naveen Garg[1,2] and Sagar Maji[2]
[1] Apex Level Standards & Industrial Metrology Division, CSIR - National Physical Laboratory, New Delhi, India
[2] Department of Mechanical, Production & Industrial Engineering, Delhi Technological University, Delhi, India
ngarg@nplindia.org

The paper presents an experimental approach to quantify the vibrations induced due to acoustic excitation in diffuse field conditions. An empirical formulation correlating the varying sound field and vibration level generated in floors and walls in diffuse field conditions has been developed. A lower limiting frequency of 125 Hz for good diffusion is observed due to random wide band acoustic excitation in diffuse field conditions, below which lower vibration levels are registered due to discrete room modes.

INTRODUCTION

Structural vibrations results from both air-borne and ground-borne excitation. The induced vibration in buildings results from various external sources, for example, traffic, blasts, construction activities, sonic boom, low frequency noise from aircrafts flyover, impulsive impacts or human activities. The acoustic waves exert fluctuating forces over the building elements, causing them to vibrate which may be enhanced by resonances in case the frequency of sound waves interacting lie within the domains of natural frequency of structure. The acoustic-elastic coupling may be pronounced at lower frequencies, that is, at natural frequencies of a building, room, or wall vibration. Hubbard [1] correlated the measured accelerations for a number of different types of noise inputs on the basis of peak noise level and found that measured acceleration levels increase linearly as the input level increases. Walls, floors, ceilings and large windows respond mainly in the "oil canning" modes at frequencies below 100 Hz and their motions are controlled largely by the beam elements. The response of windows was observed to be 0.015 g/Pa, while the wall acceleration levels were observed to be 10 dB lower than the window levels. Hodgdon et al. [2] demonstrated a threshold of rattle to be 97 dB. The investigations revealed that the *A*-weighted sound levels correlate poorly with acceleration levels, while the unweighted Sound Exposure level L_E and the maximum sound pressure level correlate well. Santos Lopes et al. [3] worked on the determination of a noise level limit to be imposed on any music sound equipment operating inside the sensitive building in order to avoid damaging vibrations in the building facades. A sound level limit of 105 dB(A) was proposed corresponding to the maxima velocity value of 0.7 mm/s for the root mean square velocity measured in a direction normal to the wall, and 3.5 mm/s for peak normal velocity. A frequency limitation of 63 Hz corresponding to sound pressure of 80 dB(A) was also proposed as limit to be imposed by an electronic device, connected to a microphone.

The difference between a mechanical excitation and acoustic excitation of a given structure is actual coupling between the structural modes and applied excitation. The coupling efficiency depends upon how well the sound waves interact with structural modes in case of acoustic excitation [4]. The problem of acoustic fatigue is also very critical for design of aircraft structures subjected to high acoustic loads due to which light weight structures are tested in reverberation chambers to simulate the launch conditions. Statistical Energy Analysis (SEA) has been widely employed by many researchers to predict vibroacoustic problems for interconnected mechanical systems [5,6]. Chang and Nicholas [7] used Green's functions to study the frequency response of structural–acoustic systems. The sensitivity of the structure to diffuse acoustic field has been modelled explicitly by Cremer and Heckl [8] using the principle of acoustic reciprocity, wherein the sound power radiated by the structure is analyzed numerically when a mechanical force *F* acts upon the structure.

$$\frac{|v'|^2}{|p'|^2} = \frac{8\pi}{k^2 \rho^2 c^2}\frac{P_{rad}}{|F|^2} \qquad (1)$$

where k is the wave number, ρ is the density of the medium and c is the speed of sound. $|v'|^2$ is the structural velocity response squared at a certain point A of the structure due to a diffuse acoustic sound field with a (spatially averaged) sound pressure level $|p'|^2$ (the sound incidence case), whereas P_{rad} is the acoustic sound power which is radiated by a force F acting on the same point A.

Rozen et al. [9] discussed a numerical procedure to predict the disturbances due to acoustic excitation of machinery. The sensitivity of a simple structure consisting of a cantilever beam and a base plate to diffuse acoustic field excitation typical for the sound fields in clean rooms was predicted and measured. It was observed that simulations agree reasonably well within the measurements in a reverberant room. A recent study by Løvholt et al. [10] reveals that low frequency sound interaction with the fundamental frequencies of the building components combined with air leaks in the building envelope are the main factors that govern transmission of sound into the building. There are very few such studies that discuss the low-frequency coupling of the acoustic pressure field to the building dynamics using a 2D finite element model.

The present work aims in determining the amplitude of vibration levels induced due to sound fields in diffuse field to investigate the probability of damage in the buildings due to intense sound fields. An empirical formulation correlating the noise levels and vibration of walls and floors is developed. The magnitude of vibration levels generated is analyzed in frequency domain to understand the behaviour especially at low frequencies. However, the coupling efficiency gets accentuated in mid frequency region especially at coincidence zone. A source within the room will excite multiple resonances and thus the sound field is composed of the addition of the many standing waves that the room supports, whilst at the high frequencies, the wavelength is small compared to the room dimensions and also the acoustic energy levels are attenuated. Thus, in the high frequency region, sound waves are unable to excite the bending modes in the structure.

LABORATORY INVESTIGATIONS

An investigation was carried out in a diffuse field conditions in laboratory to measure the vibrations levels generated due to diffuse sound field excitation. The measurements were conducted in Reverberation chambers at the National Physical Laboratory, New Delhi. The dimensions of reverberation chamber are 6 m × 6.5 m × 7 m. The reverberation chamber is a room within another room, both rooms being reinforced concrete [11]. The outer room has a floor slab 300 mm thick supported on folded RCC plates and wall and ceilings are 125 mm thick. The inner room is floated on a 150 mm thick bed of coarse dry sand washed free of mud and silt. The sand bed is initially covered with 50 mm thick fiberglass and 25 mm thick particle board. The walls of inner chamber are 125 mm thick RCC resting on the floated floor made of highly polished terrazzo concrete. Imparting high polish to the surfaces, the viscous drag and thereby energy loss is minimized. The measured value of reverberation time for empty room with random uncertainty less than ± 0.1 s is shown in Figure 1.

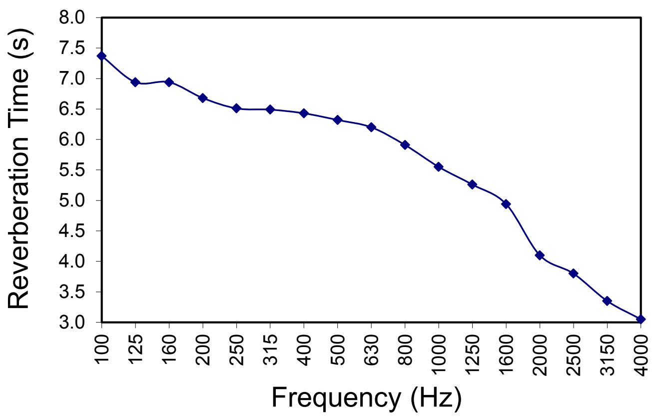


Figure 1. Reverberation time measurements for empty room

The walls, floor and ceiling are non parallel, the average inclination between the walls being 6° and between floor and ceiling 2° to 3°. To prevent the resonance modes, the 125 mm cavity between the inner and outer walls is partially filled with mineral wool blanket to cover 30% of the area. The ceiling of inner chamber is made of polished stone slabs 50 mm thick resting on steel girders and the plenum between the inner ceiling and outer room roof slab is partially filled with mineral wool to damp out resonance modes in this space. The double entry doors as shown in Figure 2 made of sandwich construction consisting of two sheets of 16 gauge mild steel on the outside and one sheet of 1.6 mm thick lead in the middle, with 25 mm air gap on the either side of lead sheet filled with fibre glass. The door panels fit into a rebated 14 gauge sheet steel frame filled with concrete after fixing in position, with rebates lined with soft rubber so as to give a good seal when door is tightly closed with wedged latch [11].

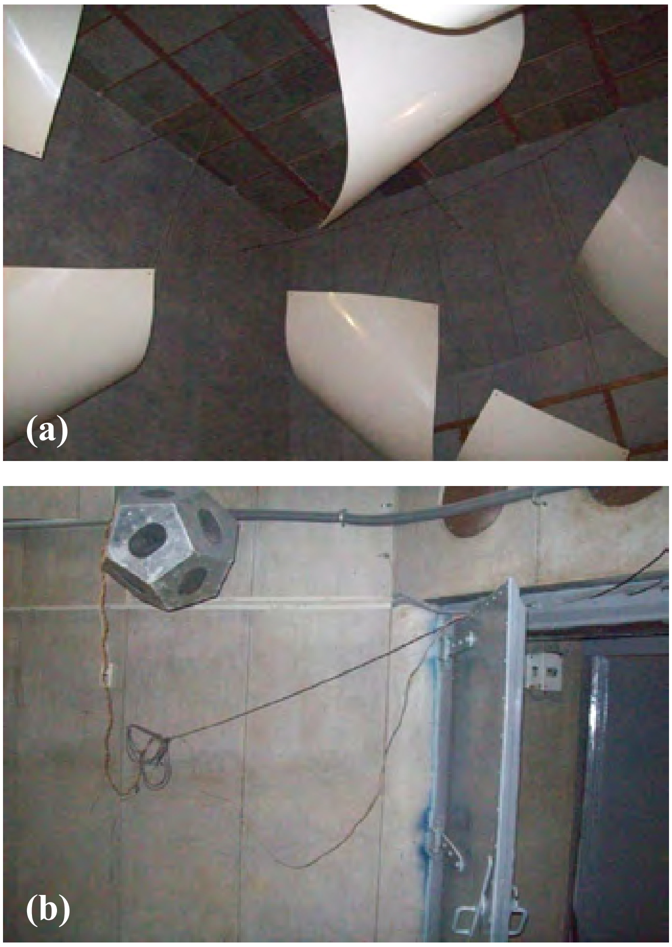


Figure 2. (a) Pictorial view of diffusers installed in reverberation chambers at NPL, (b) View of dodecahedral loudspeaker in reverberation chamber with double entry doors

The sound source installed in the room consists of twelve 100 mm × 150 mm elliptical speaker units mounted in a dodecahedral enclosure fed through a power amplifier delivering up to 20 W (rms) output as shown in Figure 2. The omni-directional microphones measuring the sound field are suspended from the ceiling at different heights and different locations so as to cover spatial zones in the chamber. A pink noise generated the acoustic excitation through a dodecahedral loudspeaker system coupled with an amplifier. Sound pressure levels inside the chamber were measured by a Norwegian Electronics 830 dual channel real time analyzer (RTA 830) in linear weighting. The vibration measurements were conducted using a seismic accelerometer B&K 8318 calibrated on primary vibration calibration system by a laser interferometer in frequency range 0.1 Hz to 1 kHz connected to a B&K measuring amplifier Type 2525 and the frequency spectrum of the induced vibration was obtained using an Agilent Audio Analyzer, Model U8903A. The sound field was generated in a varying range from 50 dB to 120 dB and the vibration levels (1 Hz to 1 kHz) on floor and walls of the chambers were measured as shown in the Figure 3 [12]. The magnitude of vibration level was measured at seven different points on the floor and walls of reverberation chamber. The standard deviation of magnitude was observed to be ± 2.88 mm/s^2 for walls and ± 3.67 mm/s^2 for floor vibration. The linear relationship of vibration levels induced due to acoustic excitation plotted in Figure 3 is consistent with Hubbard investigations [1] whereby the acceleration response increase generally as the sound pressure levels increase and follows a straight line relationship based on the assumption of linear behaviour of the structure.

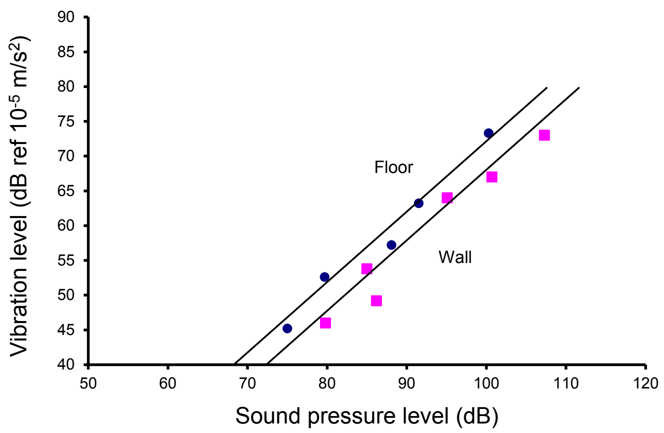


Figure 3. Induced floor and wall vibration due to wideband random acoustic excitation

The induced vibration of the floor ($L_{a(floor)\text{rms}}$) in the frequency domain shown in Figure 4 generated due to the random acoustic excitation is also correlated by a simple regression fit as

$$L_{a(floor)} = L_p - 10\log(f) - 10 \text{ (dB)} \qquad f \geq LLF,\ r^2 = 0.74 \qquad (2)$$

where $L_a = 20\log(a/a_{ref})$ and $a_{ref} = 10^{-5}$ m/s^2. Thus the above empirical relation can be used to predict the floor vibration induced to acoustic excitation in a diffuse field at different frequencies. The reverberation chamber has a lower limiting frequency (*LLF*) for good diffusion of 125 Hz. The relation between random and the lower limiting frequency is observed to be

$$L_{a(floor)}(random) = L_{a(floor)}(LLF) + 3 \text{ (dB)} \qquad (3)$$

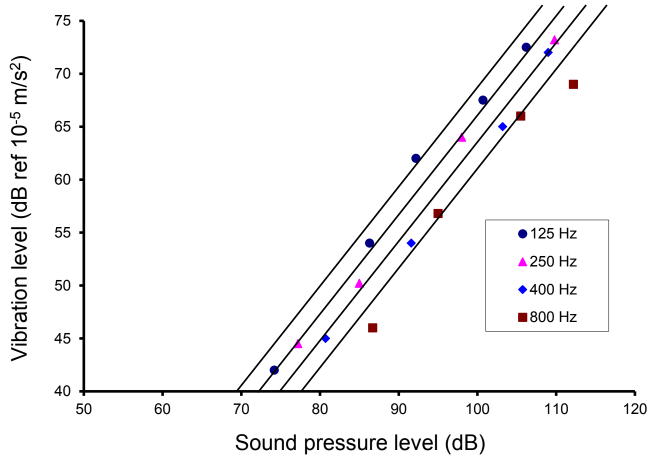


Figure 4. Induced floor vibration due to acoustic excitation of a filtered band at different centre frequencies

DISCUSSION

The enclosed space in the reverberation room can be considered as a complex resonator possessing many normal modes of vibrations excited by introducing a sound source into the room. The acoustic energy supplied by the source is considered as residing in the standing waves established in the enclosed space. The characteristic frequencies of vibration of the standing waves depend upon the room size and shape whereas the damping of these waves depends mainly on the boundary conditions. The extent of diffusion can be judged by uniformity of reverberation time within the volume of room, linearity of sound decay at different points in the room and uniformity of sound intensity distribution within the room. The experimental investigations carried out for measurement of reverberation time show that the decay curves at all positions in the room and at all frequencies have a smooth, linear drop of at least 40-45 dB from the initial sound cut-off. The distribution within the room of sound level of filtered band of white noise is within ± 0.5 dB at high frequencies and within ± 1 dB at low frequencies. Diffusing plates have also been additionally suspended from ceiling and oriented at random to enhance the state of diffusion in the room as shown in Figure 2(a). The standard deviation of correlation coefficient (sin *kr* / *kr*) was measured to be within ± 0.06 [11,13] in the frequency range 100 Hz to 125 Hz.

The expression for modal density that applies approximately to rooms of any shape including cylindrical rooms is given by [14]

$$\frac{dN}{df} = \frac{4\pi f^2 V}{c^3} + \frac{\pi f S}{2c^2} + \frac{L}{8c} \qquad (4)$$

where V is the volume, S is the total surface area and L is the sum of length of all edges of the room. At higher frequencies, there is fairly even modal distribution and spacing between the characteristic frequencies is close, while at low frequencies, there are very few modes. So, the average sound energy density is not the same throughout the enclosed space and thus the sound field is not diffuse. A diffuse field can be established in a rectangular room if there is at least 20-30 modes in the measurement bandwidth [15], and there is at least one mode per Hz. In the present case, the number of modes has the value 21 for f = 100 Hz and Δf = 13 Hz (1/6 octave bandwidth). Since the room is not symmetrical, the eigen-tone frequencies cannot be calculated easily. However, if the room is assumed to be rectangular with dimensions corresponding to the average dimension, the eigen-tones between 110 Hz and 125 Hz would have been spaced 1 Hz apart. The lower limiting frequency for good diffusion is observed to be 125 Hz. Where diffuse conditions exist, Figure 3 shows that the acceleration level increases linearly with acoustic excitation (L_{in}).The diffuse field conditions are however difficult to achieve in a normal build up areas and thus there may exist deviations from the results predicted due to empirical formulation. This may be attributed due to the spatial distribution of sound field, inherent damping of the system and excitation of resonances in case the vibration frequency falls within bounds of natural frequency of structure. Hodgson observations [16] reveal that diffuse field theory can be applied in the case of an empty room with quasi-cubic dimensions, specularly reflecting surfaces and uniform surface absorption. However, it has been experimentally demonstrated that even in small rooms, the uniformity of the sound field can be significantly improved with diffusers [17, 18]. Schroeder [19,20] described a cross-over frequency that denotes approximately the boundary between reverberant room behaviour above and discrete room modes below for airborne sound in reverberant enclosures calculated empirically from equating the half-power bandwidth B ($B = 2.2/T_{60}$) of the resonances with three times the average asymptotic spacing Δf ($\Delta f = c^3 / 4\pi V f^2$) between resonance frequencies giving f_s as [20]

$$f_s = 2000\sqrt{\frac{T_{60}}{V}} \tag{5}$$

where T_{60} is the reverberation time of the room in seconds and V is the volume of the room in m^3 and the factor 2000 (which contains the velocity of sound) guarantees that on average, at least three resonances fall within the half-power bandwidth of one resonance at frequencies above f. In the normal rooms, the frequency f of Eq. (2) becomes equivalent to f_s of Eq. (5).

The modified equation suggested by Nélisse et al. [15] is given by

$$f_s \approx \sqrt[3]{\frac{\alpha c^3}{4\pi\eta V}} \tag{6}$$

where α is the model overlap, c is the speed of sound and η is the damping factor. For a damping of $\eta = 5\times10^{-3}$ [15] and model overlap α =3 as proposed by Schroeder, f_s is calculated to be 192 Hz in the present case.

Figure 5 shows the response of floor in g/Pa at different frequency. For sound pressure level of 1 Pa (or 94 dB), Eq. (2) reduces to

$$L_{a(floor)} = 84 - 10\log(f) \quad \text{(dB)} \tag{7}$$

The overall response of the floor is observed as 0.0002 g/Pa (20.4 mm/s^2/Pa) and for the wall as 0.0015 g/Pa (13.2 mm/s^2/Pa). It can be observed that the g/Pa value diminishes at higher frequencies and also strong coupling of sound waves with the structural modes in dominant in the frequency range from lower limiting frequency (*LLF*) up to 500 Hz.

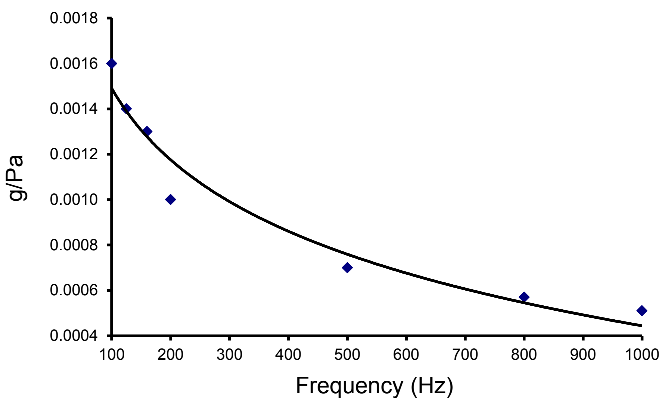


Figure 5. Response of floor to induced acoustic excitation in g/Pa

In most of the practical situations, the acoustic excitation in the free-field conditions and induced vibrations on the facades and low frequency response exhibit a complex behaviour with uncertainties arising from sound structure interaction. Some of the acoustic excitations like sonic boom, cracker bursting and open air detonation of charges can produce high acoustic stimulus of lower frequencies. Since their occurrence is hardly likely and is always confined to outdoors, this affect their coupling to the facade of a structure under free field condition and the induced vibrations never exceed the indoor diffuse criteria as described by Eq. (2). Air borne excitation is mainly low frequency sound waves interacting with building elements like windows, doors etc. causing them to vibrate, while ground borne vibration propagates through building foundation and floor supporting walls. The interaction of sound waves with structure in free-field conditions is quite cumbersome to model as various uncertainties are involved in acoustic-elastic coupling. Thus, a large database for vibration induced due to various noise sources like transportation, aircrafts flyover, blasts etc is required for analyzing the severity and perceived response by the community. For instance, a study conducted to ascertain the magnitude of maximum floor vibration level generated in a historical structure during ceremonial gun firing reveals a vibration level of 6×10^{-3} g (rms) for noise level of 125 dB(A). The floor modes of 25 Hz, 42 Hz, 69 Hz and 100 Hz get amplified during the excitation [21]. Another study conducted for monitoring the transient acceleration induced due to overflying aircrafts landing and take-off over ancient monument reveals a maximum acceleration observed as 3×10^{-3} g (rms) and major resonant modes excited in structure lying in frequency range 10 Hz to 100 Hz [22]. Experience for blasting, explosions and for sonic booms suggest that damage to houses may occur at peak acceleration values between about 0.3 and 3.0 g in the frequency range of 10 to 100 Hz respectively [23]. The widely used German standard, DIN 4150 [24] provides limit values for different types of structures and for different sources of vibration in conjunction with assessment of building damage caused by short-term and long-term vibrations. The generally accepted code of degree of damage to structures is correlated with the ground motion peak velocity and frequency as the strain imposed on the building at foundation level is proportional to peak particle velocity. In present context, for the extreme condition of the random acoustic excitation in diffuse field, Eq. (2) can be considered for predicting the maximum acoustic excitation for structural integrity as prescribed by DIN 4150 and BS 7385 standards [25]. The present study also emphasizes the need for correlating the induced vibration levels with *C*-weighted sound pressure level or sound exposure levels as *A*-weighting devaluates the low frequency noise.

CONCLUSIONS

An empirical formulation correlating the vibration induced due to acoustic excitation in diffuse field conditions is developed. In practical situations, as the diffuse field conditions don't prevail, the interaction of sound waves with structural modes gets diminished resulting in weak coupling and thus lower vibration levels are registered. The transition frequency also called the Schroeder frequency thus governs the interaction of sound waves with structural modes in practical situations. It may be noted that the vibration of any structure is dependent upon the material properties and boundary conditions in addition to the external forcing function. The paper considers the forcing function (diffuse field conditions) only and can't be generally applied to other structures with a definite level of confidence. However, it quantifies the maximum vibration levels induced in a diffuse field set up for adjudging the severity of vibration levels induced due to high acoustic loads. Further investigations in this regard on vibration induced due to acoustic excitation in free field conditions and correlation of the induced vibration with weighted acoustic excitation (*A*-weighting, *C*-weighting, L_{max} and Sound Exposure Level) in free field conditions shall be helpful in better understanding of sound waves interaction with structural elements in practical situations. A comparison with diffuse field conditions is also to be investigated for characterizing the vibroacoustic behaviour of structures explicitly in different situations.

ACKNOWLEDGEMENTS

Authors would like to thank Archaeological Survey of India (*ASI*) for funding the project on investigations on induced vibrations due to acoustic excitation from Sound show at historic monuments, NPLI/CNP 090532 and CNP 090732. The author would also like to express his sincere gratitude to the Director of NPL for allowing the work to be published. Authors also thank retired colleague Mr Omkar Sharma for his support and help in experimental observations and analysis in this study.

REFERENCES

[1] H.H. Hubbard, "Noise induced house vibrations and human perception", *Noise Control Engineering Journal* **19**, 49-55 (1982)

[2] K.K. Hodgdon, A.A. Atchley and R.J. Bernhard, *Low frequency noise study*, Technical Report, The Partnership for Air Transportation Noise and Emissions Reduction, PARTNER-COE-2007-001, 2007 http://web.mit.edu/aeroastro/partner/reports/proj1/lfnreport-2007-001.pdf (last accessed 5 January 2014)

[3] L. Santos Lopes, J. Patrício, Schiappa de Azevedo and F. Aristides Chaves, "Vibrations due to sound fields in a sensitive building. Damage criterion for the noise level", *Proceedings of Acústica 2008*, 20-22 October 2008, Coimbra, Portugal

[4] D.C.G. Eaton, *An overview of structural acoustics and related high-frequency-vibration activities*, European Space Agency Bulletin No. 92, 1997 http://www.esa.int/esapub/bulletin/bullet92/b92eaton.htm (last accessed 5 January 2014)

[5] K. Renji and M. Mahalakshmi, "High frequency vibration energy transfer in a system of three plates connected at discrete points using statistical energy analysis", *Journal of Sound and Vibration* **296**, 539-553 (2006)

[6] A.J. Keane and W.G. Price, "Statistical energy analysis of strongly coupled systems", *Journal of Sound and Vibration* **117**, 363-386 (1987)

[7] L. Chang and J. Nicholas, "Radiation of sound into a cylindrical enclosure from a point-driven end plate with general boundary conditions", *Journal of the Acoustical Society of America* **91**, 1504-1513 (1992)

[8] L. Cremer, M. Heckl and B.A.T. Petersson, *Structure-borne sound*, 3rd edition, Springer, Berlin, 2005

[9] N.B. Roozen and M.J. Vervoordeldonk, "Prediction and control of acoustically induced vibrations of high-precision equipment", *Proceedings of the 22nd American Society for Precision Engineering Annual Meeting*, Texas, USA, 14-19 October 2007

[10] F. Løvholt, C. Madshus and K. Norén-Cosgriff, "Analysis of low frequency sound and sound induced vibration in a Norwegian wooden building", *Noise Control Engineering Journal* **59**, 383-396 (2011)

[11] M. Pancholy, A.F. Chhapgar and O. Sharma, "Construction of a reverberation chamber at National Physical Laboratory of India", *Journal of the Acoustical Society of India* **V**, 27-33 (1977)

[12] O. Sharma, M. Singh and N. Garg, *Investigations on induced vibrations due to acoustic excitation from sound and light show at Sri Brihadeeswarar temple*, Thanjavur, Tamil Nadu, NPL Technical Report No. AC.C.10-1.F, 2010

[13] C.G. Balachandran, "Random sound field in reverberation chambers", *Journal of the Acoustical Society of America* **31**, 1319-1321 (1959)

[14] D.A. Bies and C.H. Hansen, *Engineering noise control, Theory and practice*, 4th edition, Spon Press, 2009

[15] H. Nélisse and J. Nicolas, "Characterization of a diffuse field in a reverberant room", *Journal of the Acoustical Society of America* **101**, 3517-3524 (1997)

[16] M. Hodgson, "When is diffuse-field theory applicable", *Applied Acoustics*, **49**, 197-207 (1996).

[17] P. D'Antonio and J. Konnert, "The reflection phase grating diffusor: Design theory and application", *Journal of the Audio Engineering Society*, **32**, 228-238 (1984)

[18] S.J. Loutridis, "Quantifying sound-field diffuseness in small rooms using multifractals", *Journal of the Acoustical Society of America* **125**, 1498-1505 (2009)

[19] M.R. Schroeder, "Frequency-correlation functions of frequency responses in rooms", *Journal of the Acoustical Society of America* **34**, 1819-1823 (1962)

[20] M.R. Schroeder, "The "Schroeder frequency" revisited", *Journal of the Acoustical Society of America* **99**, 3240-3241 (1996)

[21] O. Sharma, V. Mohanan and M. Singh, "Damage criteria for induced vibration in buildings", *Journal of the Acoustical Society of India* **25**, III-3.1- 3.5 (1997)

[22] V. Mohanan and O. Sharma, "Induced structural vibration on Khajuraho temples due to overflying aircrafts", *Journal of the Acoustical Society of India* **27**, 123-127 (1999)

[23] H.R. Nicholls, C.F. Johnson and W.I. Duvall, *Blasting vibrations and their effects in structures*, US Bureau of Mines, Bulletin 656, Washington DC, 1971

[24] DIN 4150-3:1999, *Structural vibration, Part 3: Effects of vibration on structures*

[25] British Standard BS 7385-1:1990, *Evaluation and measurement for vibration in buildings*

ACOUSTICS AUSTRALIA REVIEW SURVEY 2013

In July 2013, the Council of the Australian Acoustical Society established a panel to review the current production of Acoustics Australia, the value of the journal to the membership of the Society and propose changes as necessary.

The editorial board, with reference to the review panel, developed a survey to assess the opinions of the membership regarding the journal itself and on plans for alternative distribution means. The questions in the survey were aimed to provide a balance between ranking type responses and free text comments.

Soon after the distribution of the August 2013 issue of the Acoustics Australia journal, the members of the Australian Acoustical Society (AAS) were asked by email to give their views on Acoustics Australia via an online survey. Of the approximately 500 members, 156 responded to the survey. A summary of the finding is given here and the full report an analysis is available on the webpage at

http://www.acoustics.asn.au/members/forms/Acoustics_Australia_Review_Survey-2013.pdf

This is in the members only area so you will need to log in to access.

While it is clear from the results is that there is a great diversity in the opinion of the journal, the bulk of those that responded do read and find the journal useful at least some of the time. The comments suggesting Acoustics Australia be more like the Acoustics Bulletin, (a non peer review journal) were offset by those saying Acoustics Australia should strive to achieve a higher impact factor. There were a number of comments suggesting preference for more Australian related articles and fewer theoretical papers from overseas. There also needs to be more clarity to the readership of the editorial process and the plans for future issues.

There was a mixed but strong feeling about editorial control for the letters to the editor with particular recommendations to clearly identify any background or vested interest of the letter writer and, where appropriate, to allow the right of reply in the same issue.

In regard to the format for distribution, the results indicate that it is time to go to a full electronic (pdf) distribution for the journal as the bulk of the respondents accepted a move to an electronic version. The distribution of responses to the question about preference for hard copy versus pdf if there was no change in membership fee is shown below. A move away from hard copy distribution will be a major cost saving for the journal production. For those few who are prepared to pay for a hard copy at a cost recovery price, a limited print run using a more cost effective process, could be made available.

Some commented that more articles may be submitted if the website is made clearer, the future planning of special issues is publicised and if members are invited to submit articles. This could be achieved with ad hoc emails from the General Secretary and specific prompts to individual members plus more information in the journal itself about the process and seeking submissions.

Marion Burgess

INVERSE GABOR TRANSFORM FOR SPEECH ENHANCEMENT

Mohammed A. Al-Manie[1] and William J. Wang[2]
[1]Computer Research Institute, King Abdulaziz City for Science and Technology, Riyadh 11442, Saudi Arabia
[2]School of Engineering and Information Technology, University of Sussex, Brighton, BN1 9QT, England
malmanie@gmail.com

In this paper, the Inverse Discrete Gabor Transform (IDGT) is proposed for signal recovery buried in board-band non-stationary noise. Time-frequency masking filtering technique is implemented to reject the noise from corrupted speech while at the same time maintaining the desired waveform. A synthetic multicomponent non-stationary test signal made up of two chirps was first used to simulate noise; the signals were then separated using this technique. Four English speech signals recorded in different environments such as airport, restaurant, and train buried in wide-band noise were reconstructed. The extracted signals were then compared with the original ones in terms of cleanness and noise removal. The implemented procedure is suitable for this type of wide-band non-stationary interference, which cannot be canceled in the Fourier (frequency) or time domain.

INTRODUCTION

In the year of 1946 a physicist by the name Dennis Gabor proposed the decomposition of a continues-time signal into a set of shifted and modulated elementary discrete signals [1]. The spectrum in this case is obtained by multiplying the analysis signal by a Gaussian window of a chosen length, the Fourier transform is then, calculated for this particular function. The Gabor transform is a time-frequency distribution with a number of useful applications such as speech, seismic, and image-processing signals that have characteristics of a time-varying spectrum not appropriate to analyze in the Fourier transform method [2-4]. The main advantage of this technique is in the ability to detect frequency contents changes with respect to time, which may be very important in cases, such as medical imaging where diagnoses of abnormalities are related to time. Another example is detection of localized faults in mechanical and electrical systems that also vary with time [5,6]. This is in contrast to the classical Fourier analysis, which gives information about the overall spectral content of a particular signal without providing information about how these frequencies evolve in time.

In general, the discrete Gabor transform (DGT) can be thought of as a windowed Fourier transform, which provides a representation of a signal in time and frequency simultaneously. The Gabor coefficients are calculated as the inner product of the test signal and a single analysis window used to calculate the transform [7,8]. A major advantage that DGT offers is the ability to reconstruct the original signal. This is accomplished by summing the translated and modulated parts of the test signal obtained by the fixed synthesis window after being weighted by the Gabor coefficients. The Gabor transforms also offers the ability to change the time-frequency distribution (TFD) of a signal by adjusting the magnitude of the Gabor coefficients and recovering the original signal. This procedure is used as a time-varying filter for non-stationary signals that cannot be recovered in the time or frequency domain. The result of reconstruction can also be compared with the original value for an accuracy test of the Gabor transform representation.

In addition, the inverse discrete Gabor transform (IDGT) has been implemented by a number of researchers for various applications such as seismic de-convolution proposed by Margrave [9]. In this approach, a non-stationary filter was accomplished by modifying the Gabor time-frequency decomposition in order to recover the desired seismic signal. In another procedure of time-frequency synthesis, Xiang et al. [10] proposed an iterative time-varying filtering algorithm in the discrete Gabor domain. A non-stationary chirp test signal was extracted from a wide-band noise by applying an iterative algorithm, then comparing the result with the noisy waveform. In order to decrease computational complexity, Tao et al. [11] suggested a block time-recursive algorithm to find the discrete Gabor coefficients then, extract the original signal for both the critical sampling and the over-sampling cases.

This paper is organized as follows. In Section 2 a brief overview of the discrete Gabor transform (DGT) is provided. The inverse discrete Gabor transform (IDGT) and its applications to signal reconstruction is presented in Section 3. In the next Section 4, experimental results and findings including implementation procedure for this experiment are outlined. In the final part of the paper, Section 5, analysis of results and conclusions are presented.

THE DISCRETE GABOR TRANSFORM (DGT)

In this section, a review of the discrete Gabor expansion is presented, before implementation procedure is conducted in the later part of the experiment.

The discrete Gabor expansion

For a discrete-time finite, real and periodic function $x(n)$ with a period L, the Gabor expansion can be written in the following form [7,12]

$$x(n) = \sum_{m=0}^{M-1} \sum_{k=0}^{N-1} a(m,k) h(n, m\overline{N}) e^{\frac{j2\pi kn}{N}} \tag{1}$$

where M is the total number of time sampling points and N is the number of points sampled in frequency. The values of $\overline{N}$ represents the time sampling interval while $\overline{M}$ is the frequency-sampling interval.

On the other hand, the coefficients $a(m,k)$ are calculated according to the following formula

$$a(m,k) = \sum_{n=0}^{L-1} x(n) \gamma(n, m\overline{N}) e^{\frac{j2\pi kn}{N}} \tag{2}$$

Note that in the above equation the equality $L = \overline{N}M = N\overline{M}$ holds. The critical sampling point occurs when $\overline{N}\,\overline{M} = NM = L$ where the number of samples from the original signal is equal to the number of Gabor coefficients. Under sampling occurs when $NM > L$ which can leads to information loss. Therefore, for perfect reconstruction of the original signal, the following condition must be satisfied $NM \geq L$.

Moreover, both the analysis window $h(n)$ and the synthesis window $\gamma(n)$ must be real, periodic, satisfying the biorthogonality condition given by:

$$h(n) = \sum_{n=0}^{L-1} h(n + mN) e^{\frac{j2\pi kn}{\overline{N}}} \gamma(n) = (L / MN) \delta_m \delta_k \tag{3}$$

where $0 \leq m \geq \overline{M}$ -1, $0 \leq k \geq \overline{N}$ -1 and $\delta(n)$ denotes the Dirac delta function. The above expression in Eq. (1) is defined as the inverse discrete Gabor transform; while Eq. (2) is known as the DGT.

THE INVERSE DISCRETE GABOR TRANSFORM (IDGT)

The de-convolution of the discrete Gabor transform refers to recovering the original signal from the time-frequency distribution. In classical Fourier analysis if the signal is narrow-band and stationary, then linear filtering is implemented to recover the original signal. In this case, a simple procedure made up of a band-pass filter covering the band of the signal is used to recover the desired output from the wide-band-noise. In the frequency domain, the band-pass filter transfer function is multiplied by the Fourier transform of the noise as a form of a mask for the spectrum. Then, the inverse Fourier transform of the result is calculated to recover the noise-free signal.

However, when dealing with non-stationary signals such as speech, biomedical or seismic data buried in noise, traditional linear filtering cannot be used. In this case, a different approach known as time-frequency filtering or masking is utilized to extract the original signal. In the discrete Gabor method, reconstruction of the signal $x(n)$ can be obtained from Eq. (1) after satisfying the conditions mentioned in the above section. That is, the inverse FFT of the coefficients $a(m,k)$ is computed for each index k, then taking the point-by-point product with the synthesis window $h(n)$ to obtain a specified portion of the output. To recover the whole signal, the windowed slices are summed over the index m.

Consequently, the above procedure can be used for time-frequency filtering by modifying the Gabor coefficients used to calculate the time-frequency distribution prior to signal reconstruction. This is known as time-varying filtering where two or more non-stationary signals can be separated and the desired one is synthesized. A number of applications take advantage of this particular technique for the objective of signal recovery. Some of these examples include a non-stationary seismic data de-convolution designed to remove unwanted earth attenuation effects and source signature [10]. Another example is the implementation of a recursive Gabor filter for image processing with the least possible number of operations [13]. An important application is speech enhancement, where this approach is exploited in the time-frequency representation to remove inherent noise and recreate the uncorrupted time wave.

EXPERIMENTAL RESULTS AND DISCUSSION

In this section, the IDGT is implemented as a time-varying filter for noise removal and enhancement of speech. First a synthetic non-stationary signal made up of tow chirps that cannot be separated using traditional linear filtering in the Fourier (frequency) or time domain is used to simulate a noise-corrupted waveform. One chirp is treated as noise while the second simulates the desired time-series wave. This is illustrated in Figure 1 showing the time-frequency representation (TFR) of the test chirp. On the other hand, Figure 2 shows the two chirps' distribution with one of the signals depicting noise presence. The original signal is then recovered or separated from the time-frequency distribution by utilizing a masking procedure to remove the present noise. The result is given in Figure 3 with the noise-free signal recovered using the proposed masking technique. The time series plot of error between the original and recovered signals is also shown in Figure 4 providing a fairly accurate result.

In order to create a suitable mask, the time frequency distribution (TFD) of the distorting noise is set to zero while at the same time the desired part is multiplied by unity. This will keep the desired components inside the mask and reject other parts of the signal. The calculation of an appropriate time-frequency mask can be done according to the following procedure:

$$M(n, m) = \begin{cases} 1, & (n, m) \in C \\ 0, & otherwise \end{cases} \tag{4}$$

The value of C represents the pass region of the time-frequency mask. Hence, the final result of the masked time-frequency distribution is in the form:

$$\overline{S}(n,m) = \widehat{S}(n,m)\, M(n,m) \tag{5}$$

where $\overline{S}(n,m)$ stands for the result of masking, $\widehat{S}(n,m)$ is the corrupted value, and $M(n,m)$ is the mask function. The signal can then be reconstructed using the inverse GDT from the masked output. In order to obtain an appropriate mask, a threshold can be calculated as the average value of the peaks in the noise- corrupted value. It is important to note that time frequency distributions are not one to one (redundant) or onto transformations. As a result, not every signal in the joint time-frequency representation corresponds to a signal in the time domain.

Furthermore, as an application to the above time-frequency masking procedure, speech utterances corrupted with different kinds of noise are synthesized for the objective of interference cancellation. This type of noise is very similar to the clean signal, which makes the task of removal very difficult or impossible using traditional Fourier techniques. The first example is the word "long" spoken by a female and recorded inside a talking crowed, which of course will affect the fidelity of speech. The signal length is 27 ms, with a sampling rate of 11 kHz. The original noise-free time-frequency distribution is illustrated in Figure 5. On the other hand, Figure 6 depicts the noisy speech, while Figure 7 is the masked result with most of the present noise removed. The error between the extracted waveform and the original is shown in Figure 8. As can be seen from this output, the recovered signal is a good approximation of the original speech with a very small difference.

As a second example, the word "down" spoken by a male of length 355 ms has a sampling rate of 8 kHz with added noise from an airport environment. The time-frequency distribution is depicted in Figure 9 for the clear signal; while Figure 10 clearly shows noise presence in the time-frequency plane. The following Figure 11 depicts the masking result of the time-frequency distribution after interference cancelation. The extracted waveform represents a very good approximation of the original as Figure 12 illustrates.

The above technique is also implemented for two other English speech examples. The next one is the word "great" spoken by a female and corrupted by a moving train sound. This has a length of 215 ms and a sampling rate of 11 kHz. The outputs are displayed in Figures 13, 14, 15, and 16 showing the error between the recovered speech and the original signal. The fourth and final example is the word "pleasant" spoken by a male inside a noisy restaurant atmosphere. In this case, the signal has a length of 420 ms with a sampling rate of 8 kHz. The answers are displayed in Figures 17, 18, 19, and 20. The extracted wave in this example also provides a promising outcome when compared with the clean signal. The overall results are also displayed in Table 1 representing signal to noise ratio (SNR) of individual speech signals before and after the proposed TFR filtering procedure. A significant improvement in the speech quality and noise reduction is achieved.

Table 1. Signal to noise ratio (SNR) in dB of the four speech utterances in different noisy environments

Test Word	**SNR Before Filtering**	**SNR After Filtering**
Long (crowed noise)	1.4804 dB	6.3620 dB
Down (airport noise)	-1.6488 dB	6.2684 dB
Great (train noise)	5.5486 dB	9.2970 dB
Pleasant (restaurant noise)	1.1858 dB	8.7993 dB

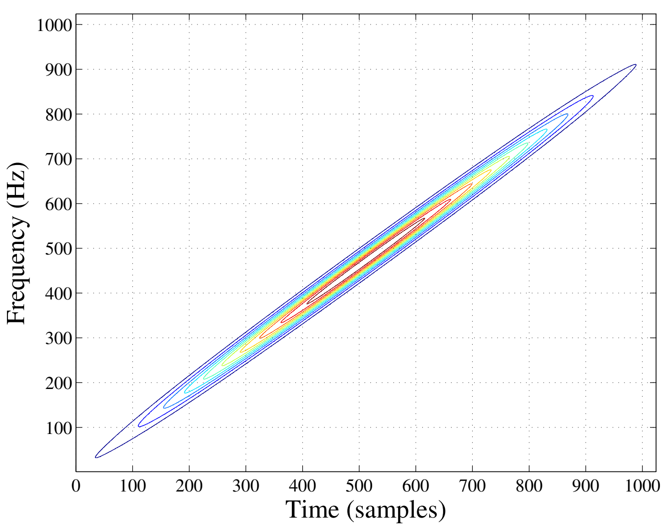


Figure 1. Time-frequency representation of the noise-free test chirp signal

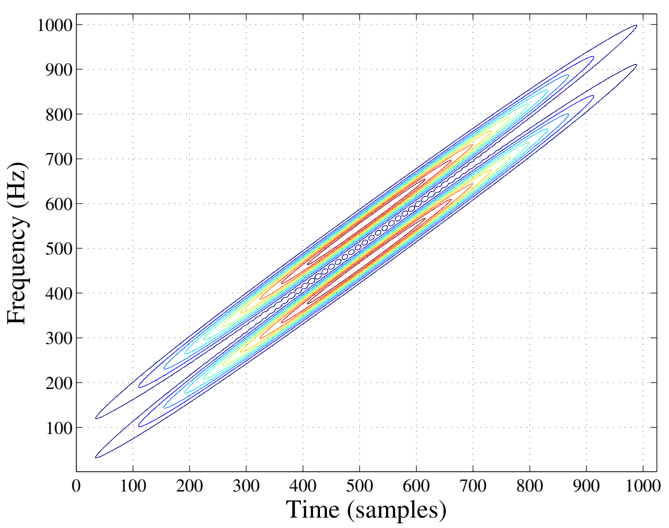


Figure 2. Time-frequency representation of the test signal with noise present

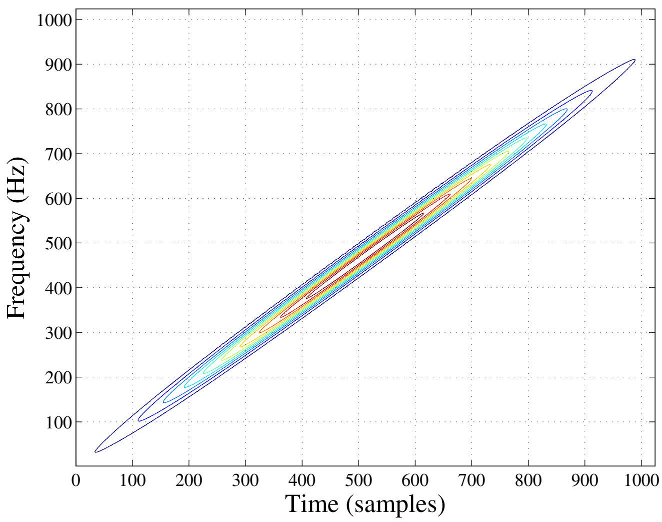


Figure 3. Time-frequency representation of the recovered test signal with noise removed

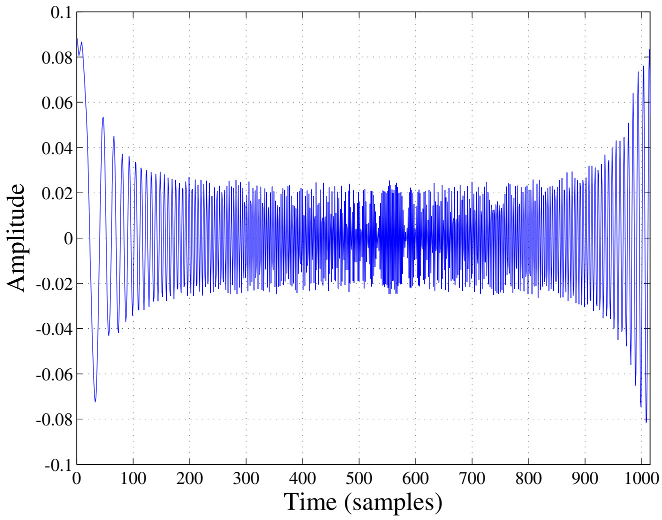


Figure 4. Plot of error between the original and the recovered signal

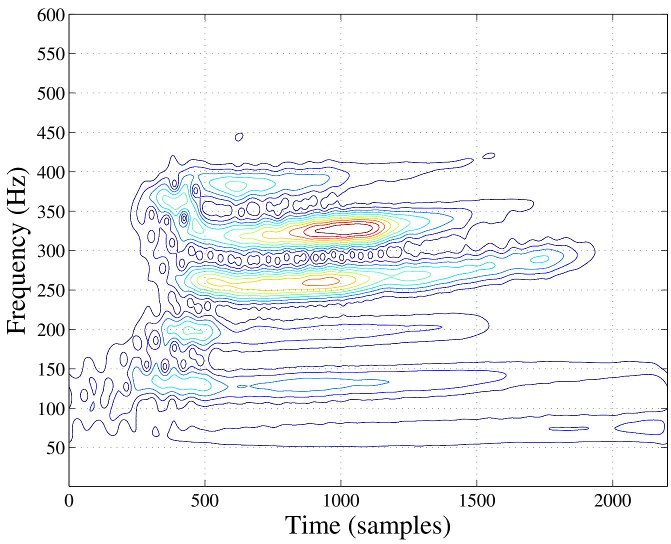


Figure 5. Time-frequency representation of the noise-free speech signal "long"

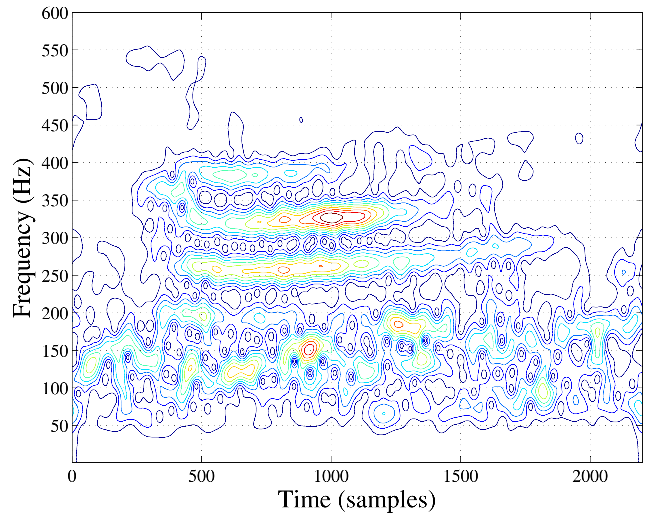


Figure 6. Time-frequency representation of the noisy speech signal "long"

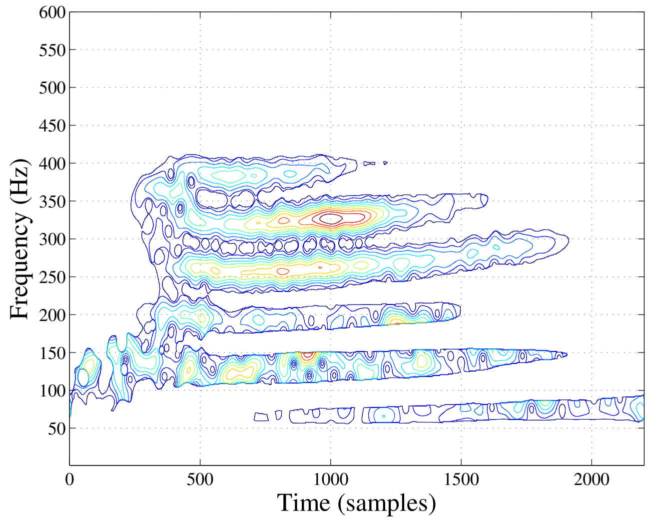


Figure 7. Time-frequency representation of the recovered speech signal "long" with most of the noise removed

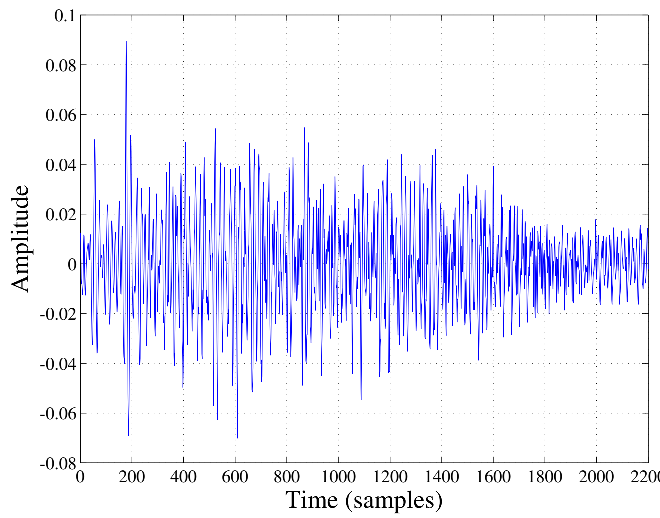


Figure 8. Plot of error between the original speech signal "long" and the recovered signal

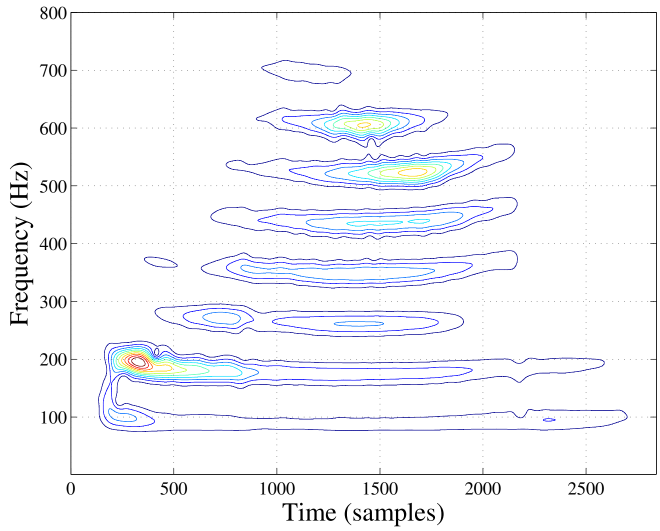


Figure 9. Time-frequency representation of the noise-free speech signal “down”

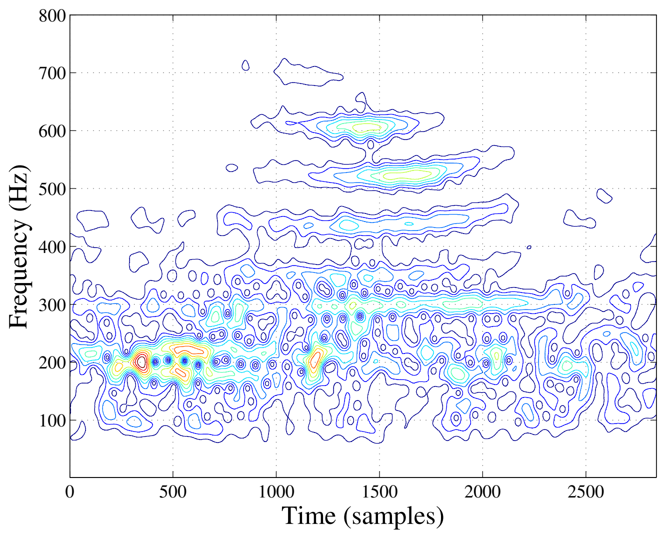


Figure 10. Time-frequency representation of the noisy speech signal “down”

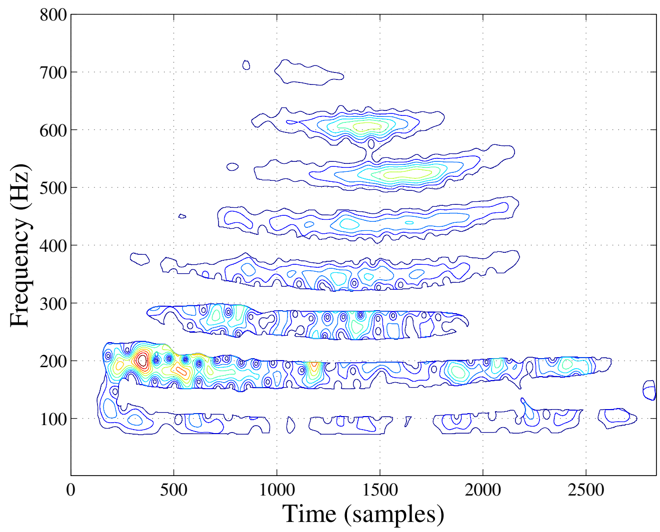


Figure 11. Time-frequency representation of the recovered speech signal “down” with most of the noise removed

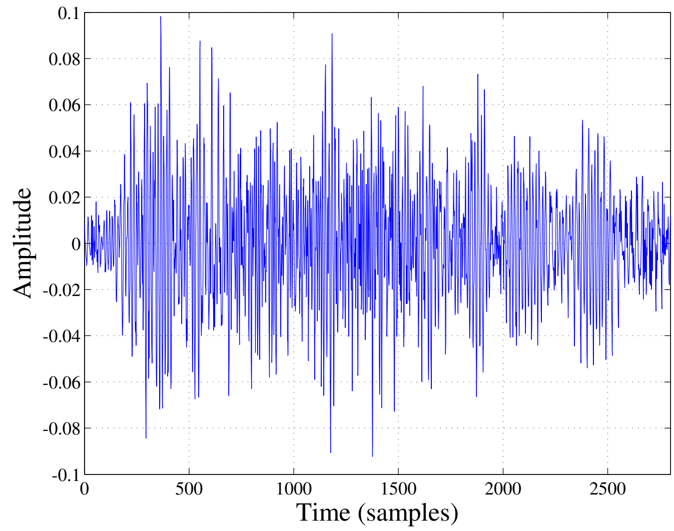


Figure 12. Plot of error between the original speech signal “down” and the recovered signal

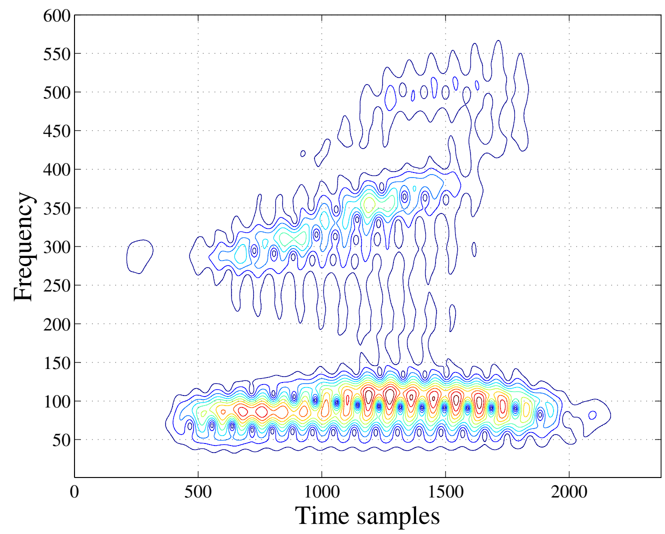


Figure 13. Time-frequency representation of the noise-free speech signal “great”

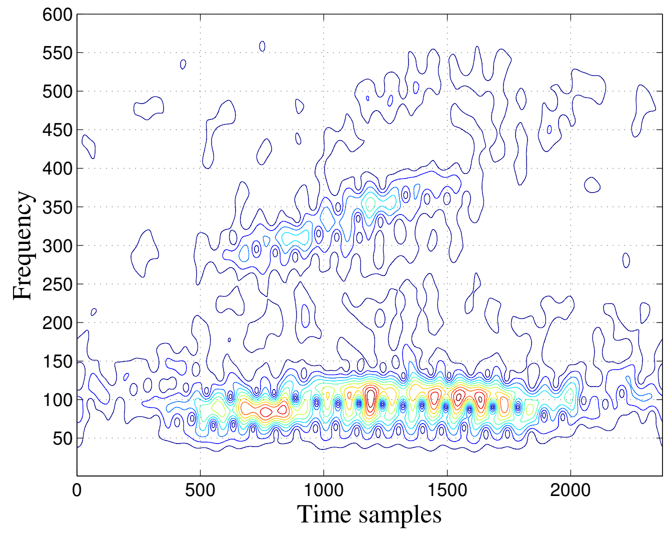


Figure 14. Time-frequency representation of the noisy speech signal “great”

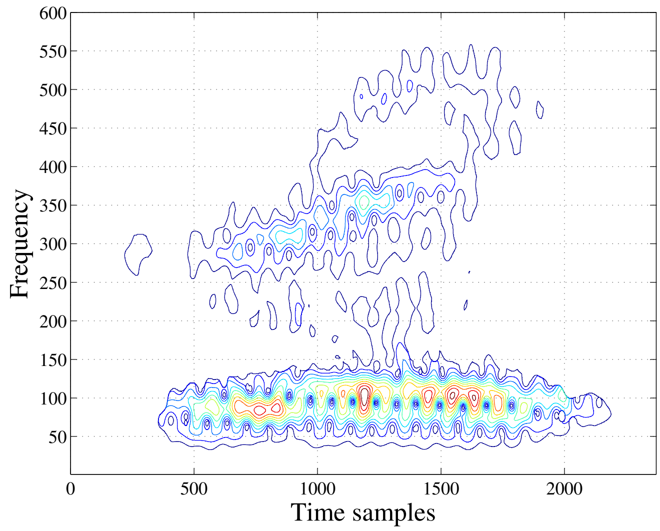


Figure 15. Time-frequency representation of the recovered speech signal “great” with most of the noise removed

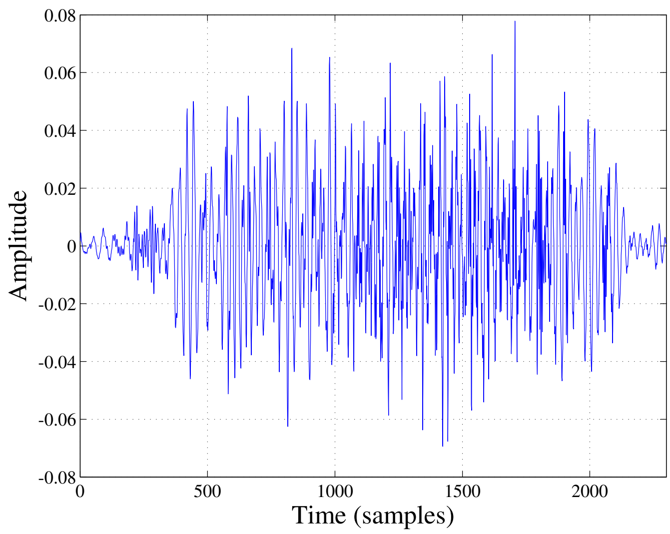


Figure 16. Plot of error between the original speech signal “great” and the recovered signal

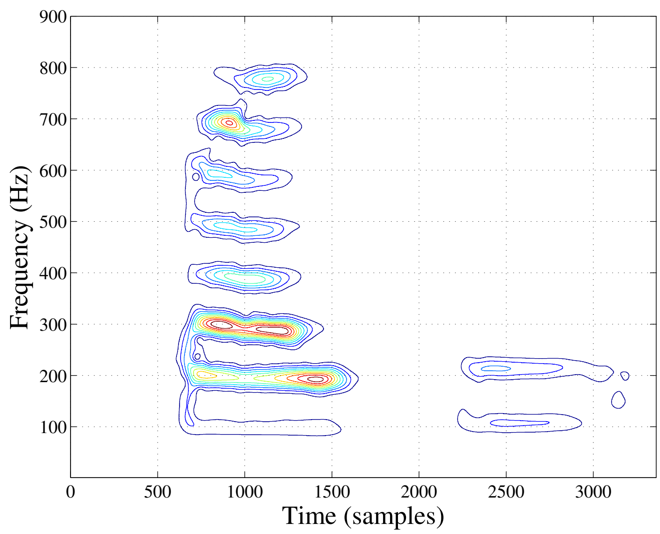


Figure 17. Time-frequency representation of the noise-free speech signal “pleasant”

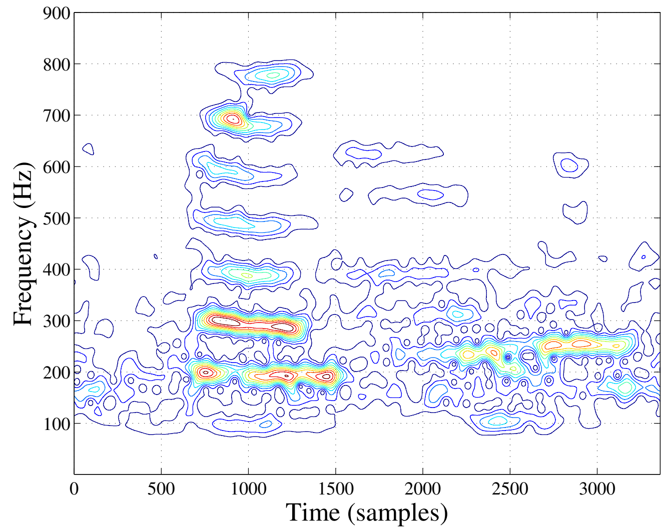


Figure 18. Time-frequency representation of the noisy speech signal “pleasant”

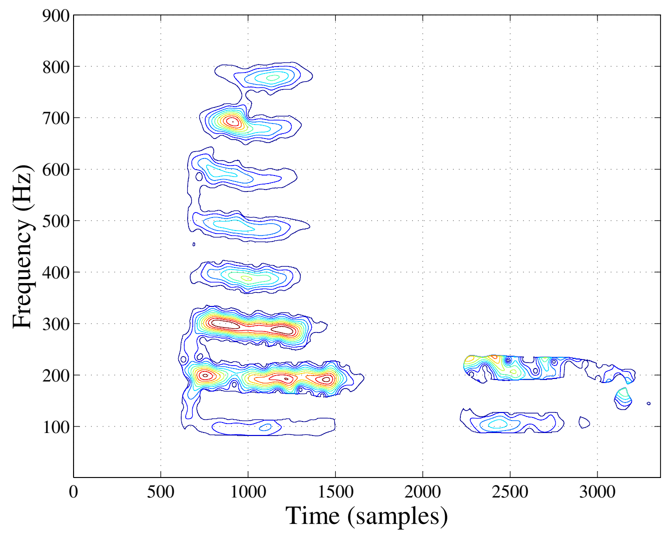


Figure 19. Time-frequency representation of the recovered speech signal “pleasant” with most of the noise removed

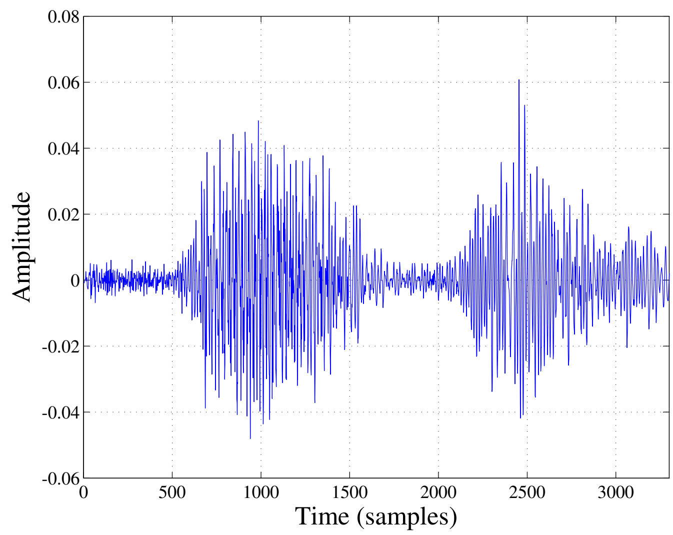


Figure 20. Plot of error between the original speech signal “pleasant” and the recovered signal

CONCLUSION

The Inverse discrete Gabor Transform was implemented to recover corrupted speech signals using a time-frequency masking approach. The main contribution of this work is that the speech signal is buried in a wide-band non-stationary noise very similar in characteristics to the original wave form, which makes it impossible to recover using classical Fourier transform techniques. First, a synthetic non-stationary multicomponent chirp was tested then; four different examples of English words recorded in noisy environments were used to evaluate the effectiveness of the proposed procedure. The original signals were then reconstructed from the time-frequency distribution of the noise-infected speech after processing. The recovered waveforms were compared to the originals in terms of fidelity or noise presence and were found to be a very good approximation of the recorded clean ones. Since this type of noise cannot be separated from the desired signal in time or frequency; this technique provides a practical alternative to the traditional methods that are not capable of resolving this issue.

REFERENCES

[1] D. Gabor, "Theory of communications", *Journal of the Institution of Electrical Engineers - Part III: Radio and Communication* **93**, 429-459 (1946)

[2] M.P. Lamoureux, P.C. Gibson, J.P. Grossman, and G.F. Margrave, "A fast, discrete Gabor transform via a partition of unity", *CREWES Technical Report Volume 15*, University of Calgary – Consortium for Research in Elastic Wave Exploration Seismology, 2003

[3] T.T. Chinen and T.R. Reed, "A performance analysis of fast Gabor transform methods", *Graphical Models and Image Processing* **59**, 117-127 (1997)

[4] A. Akan and L.F. Chaparro, "Multi-window Gabor expansion for evolutionary spectral analysis", *Signals Processing* **63**, 249-262 (1997)

[5] M.A. Al-Manie and W.J. Wang, "Time-frequency analysis by evolutionary periodogram with application in gear fault diagnosis", *International Journal of Wavelets, Multiresolution and Information Processing* **8**, 679 -693 (2010)

[6] W.J. Wang and P.D. McFadden, "Application of wavelets to gearbox vibration signals for fault detection", *Journal of Sound and Vibration* **192**, 927-939 (1996)

[7] S. Qian and D. Chen, "Discrete Gabor transforms", *IEEE Transactions on Signal Processing* **41**, 2429-2438 (1993)

[8] J. Yao, P. Krolak and C. Steele, "The generalized Gabor transform", *IEEE Transactions on Image Processing* **4**, 978-988 (1995)

[9] G.F. Margrave, P.C. Gibson, J.P. Grossman, D.C. Henley and M.P. Lamoureux, "Gabor deconvolution: Theory and practice", *Proceedings of the EAGE 66th Conference and Exhibition*, Paris, France, 7-10 June 2004

[10] X.-G. Xia and S. Qian, "An iterative algorithm for time-variant filtering in the discrete Gabor transform domain", *Proceedings of the 1997 IEEE Conference on Acoustics, Speech, and Signal Processing (ICASSP-97)* **3**, pp. 2121-2124, 21-24 April 1997

[11] L. Tao and H.K. Kwan, "Parallel lattice structures of block time-recursive discrete Gabor transform and its inverse transform", *Signal Processing* **88**, 407-414 (2008)

[12] M.J. Bastiaans and C.W. Geiln, "On the discrete Gabor transform and the discrete Zak transform", *Signals Processing* **49**, 151-166 (1996)

[13] I.T. Young, L.J. van Vliet and M. van Ginkel, "Recursive Gabor filtering." *IEEE Transactions on Signal Processing* **50**, 2798-2805 (2002)

Technical Note

Note: Technical notes are aimed at promoting discussion. The views expressed are not necessarily those of the editors or the Australian Acoustical Society. Contributions are not formally peer-reviewed.

REDUCED CONDITIONS ON AMBIENT NOISE LEVELS FOR IN-SITU AUDIOMETRIC TESTING

M. Fisher and W. Williams
National Acoustic Laboratories, Sydney, NSW

The combined Australian/New Zealand Standard AS/NZS 1269.4:2005 *Occupational noise management, Part 4: Auditory assessment* is currently undergoing revision [1]. The main change to the Standard will be with the requirements for audiometric testing, viz: *Appendix C Maximum Permissible Ambient Noise Levels for Workplace Audiometry Programs (Normative)*. In effect the current version of the Standard has a device specific requirement. This is, that testing must be performed using one of several specified audiometric headsets. The revised version will be performance based, presenting the acoustic conditions, maximum permissible ambient noise levels (MPANL), required to reliably perform audiometric testing to within acceptable limits.

This article, based on work presented previously by one of the authors [2], was prompted by enquiries received within the public comment phase during preparation of the revised Standard, requesting information on the attenuation performance when insert-earphones are used in combination with an earmuff. Such a combination with its increased attenuation possibilities, allows audiometric testing in higher ambient noise than is currently possible with either earphones or insert-earphones alone. The ability to use the insert-earphone/earmuff combination with increased attenuation means that audiometric testing can be carried out at a variety of locations where previously a portable test booth may have been required.

This is particularly significant when travel to distant or difficult-to-access sites may be involved such as remote communities and isolated mining sites. Many permanent regional or urban based locations also experience difficulties, particularly in the lower frequencies, from ambient noise produced by large, high volume, air-conditioning systems.

A particular combination of insert-earphones and earmuff as shown in Figure 1 has been tested by National Acoustic Laboratories (NAL) in accordance with the requirements of AS/NZS 1270:2002 *Acoustics – Hearing protectors* [3]. The test results are presented in Table 1.

Figure 1. MSA 766243 High Profile Earmuff and ER-3A insert earphones

These attenuation values may be used to calculate the maximum permissible ambient noise levels in octave-bands required to test in accordance with the revised Standard. The procedure follows that outlined in ISO 8253-1:2010 [4] and Williams [5]. These attenuation values will easily be reached when the insert-earphones

Table 1. Mean octave-band and overall attenuation values provided by the combination of ER-3A insert earphones in combination with an MSA 766243 High Profile Earmuff [1]

	Attenuation (dB) at octave-band centre frequency (Hz)						
Frequency	125	250	500	1k	2k	4k	8k
Mean	35.3	40.7	49.5	43.9	37.2	49.0	45.0
SD	8.3	6.4	7.2	6.1	4.4	4.7	5.6

and earmuffs are well fitted by the tester. With judicial fitting attenuation above the mean can be expected. The use of the MSA 766243 High Profile Earmuff enclosure allows for sufficient room to cover the ER–3A inserts without impeding the ER-3A performance while the large volume provides better attenuation at low frequencies as compared to enclosures with less volume. (Note: It must be ensured that the tubing connected to the ear-insert is in no way bent or crushed during use.)

This information and attenuation data will be of interest to those who supply audiometric test services and associated equipment.

REFERENCES

[1] Australian/New Zealand Standard AS/NZS 1269.4:2005 *Occupational noise management, Part 4: Auditory assessment*, Standards Australia, Sydney, 2005

[2] M. Fisher, B. Rudzyn, G. Jarvis and H. Dillon, "Improved hearing assessment accuracy in noisy environments", *Proceedings of the Audiology Australia XX National Conference 2012*, Adelaide, Australia, 1-4 July 2012

[3] Australian/New Zealand Standard AS/NZS 1270:2002 *Acoustics – Hearing protectors*, Fifth edition, Standards Australia, Sydney, 2002

[4] ISO 8253-1:2010 *Acoustics - Audiometric test methods - Part 1: Pure-tone air and bone conduction audiometry*, International Organization for Standardization, Geneva, Switzerland, 2010

[5] W. Williams, *The calculation of maximum permissible ambient noise levels for audiometric testing to a given threshold level with a specified uncertainty*, NAL Report No. 133, National Acoustic Laboratories, Sydney, 2010 http://www.nal.gov.au/pdf/NAL%20Report%20133%20-%20The%20calculation%20of%20B-G%20noise%20for%20audiometric%20testing.pdf

Technical Note

Note: Technical notes are aimed at promoting discussion. The views expressed are not necessarily those of the editors or the Australian Acoustical Society. Contributions are not formally peer-reviewed.

LOW COST REMOTE DATA ACQUISITION SYSTEM

Kristoffer K. McKee, Gareth L. Forbes, Ilyas Mazhar, Rodney Entwistle and Ian Howard
Department of Mechanical Engineering, Curtin University, Perth, Australia, and CRC for Infrastructure and Engineering Asset Management, k.mckee@curtin.edu.au

Remote data acquisition (RDAQ) is required in a number of vibration and noise measurement settings. Specialised systems exist for RDAQ, however they require significant investment to set up. This paper describes a low cost method of RDAQ which utilises a laptop computer, data acquisition system and a USB internet dongle. The system described in this paper would allow a professional vibration/noise measurement specialist to modify their existing data acquisition system for remote use with minimal cost. The system was tested both in an industrial and an academic setting for verification.

INTRODUCTION

It is common for researchers and maintenance professionals who perform diagnostics on machines to travel considerable distances to the location of a machine, acquire the needed data from the machine via a data acquisition system (DAQ), and then return to the office for data processing. As a result, a considerable amount of time, money and energy is lost in the process. Problems sometimes arise when unexpected complications and failures in the machine arise in between visits and are not captured by the DAQ, resulting in a loss of valuable information.

The ability to remotely acquire the data from onsite sensors would solve these problems and possibly create the ability to perform condition based monitoring of a machine or any other kind of continuous monitoring required. Sensors, in this case, would be left on site to continually or periodically monitor the machine. Data gathered on site could then be processed and key indicators extracted to be uploaded via internet to a server that is accessible to the user.

A GENERIC LOW COST RDAQ SYSTEM

There are two main parts to this RDAQ system: hardware and software. The hardware components can be replaced with similar parts than those presented as long as they are able to meet the task. The software components are sometimes based on the hardware components, such as those used for the data acquisition, while others are freeware and are only dependent on the type of environment used, such as command line interfaces versus graphical user interfaces.

Hardware

The remote data acquisition system (RDAQ) usually consists of 5 pieces of standard hardware, as given in Table 1. Figure 1 also shows an example of the system used for this study.

Figure 1. Set up for Remote Data Acquisition System

Software

It is assumed that the RDAQ system will be unmanned for the majority of the time. Hence, the software utilised must automate the major processes needed. Three main steps are required in the RDAQ: data acquisition, data processing and uploading the data. These three steps were programmed to occur at different times so that the maximum amount of computing resources available could be dedicated to the task.

Data acquisition

The purpose of the work presented in this paper is to outline a process to convert an existing DAQ system into a remote(R) DAQ system. In the example system used in the RDAQ measurements presented later in this paper, two DAQ systems were employed, being:

1. SignalCalc ACE Quattro by Data Physics (controlled by propriety software)
2. National Instruments DAQ (controlled by Matlab)

Data Acquisition is usually performed with a program that is associated with the DAQ hardware. In this particular case, the National Instruments' cDAQ 9178 system was controlled using a script written in MATLAB®, which possibly has become available to users in recent years. MATLAB® was chosen over National Instruments' Labview environment due to the accessibility of the program and expertise in writing

Table 1. RDAQ hardware requirements

	General hardware component	Specifications of hardware used in this study. c.f. Figure 1
1	Portable DAQ	The DAQ used was the National Instruments' cDAQ 9178 system, along with 4 B&K accelerometers to gather the data. The data was sampled at 25.6 kHz. Another DAQ also utilized was called Quattro by Data Physics. Quattro has a four channel sensor input, a trigger input, and two channels for output. Newer systems, such as the B&K Pulse System, contain a static IP address, and thus allow easier access over the internet.
2	Computer	Dell Latitude with an Intel® Core™ i5-2520M CPU, running at 2.50 GHz with 4 GB of RAM and 300 GB of hard drive space. It was running with Windows 7 Enterprise Edition, 64-bit operating system using Service Pack 1. A suitable computer would be able to analyse the collected data in real time, if necessary, or in the amount of time provided between collecting the data. Slower or less powerful processors, such as the Intel® Core i3, Celerons, Atoms, or Pentium series, may not provide the processing power needed to perform real time calculations. Experimentation with this hardware prior to deployment in the field is necessary.
3	External hard drive	500 GB external hard drive, which was deemed as being sufficient for the task.
4	USB dongle from an internet provider	Telstra Elite on the Next G network. Telstra was chosen due to its availability as an internet service provider Australia wide. A suitable internet provider should be selected based on coverage, strength of signal, and available data plans.
5	Powered USB hub (optional depending on the distance between the machine and the DAQ, and between the DAQ and the computer)	This was used to boost the signal from the DAQ to the computer since the location of the sensors was found to be too far from the computer. In some instances, the distance between the DAQ system and the computer can be at least 50 metres; hence the signal must be amplified in order to successfully transmit it. This component, which is not shown in Figure 1, may not be necessary in those instances in which the location of the DAQ and the computer are in close proximity.

scripts in the program.

For environments in which scripts are not able to be written to control the DAQ process, and only a GUI is provided, additional software is needed in order to carry out this process. This was the case when using the Quattro DAQ. In such cases, two software components were utilised, AutoHotKeys and Devcon, both of which are freeware. AutoHotKeys is a Windows scripting language that allows the user to automate the clicking and inputting of needed information. Using a text editor, scripts can be created that recognise the appearance of graphic user interfaces (GUI), locate and input strings of characters or numbers at predetermined locations on the GUI, and click on buttons found on the GUI to proceed with the tasks. Devcon is a system tool provided by Microsoft in their Windows SDK for Windows 7 package. Devcon is utilised through a command line interface, such as the dos command window, and thus can be called by AutoHotKeys via batch files. It provides the ability to turn off and on drivers for the DAQ. This is needed during those unforeseen times in which either the DAQ or the Windows operating system has been rendered inoperable and must be rebooted in order to proceed. Devcon must be run with administrator privileges.

Data processing

The extent of the data processing step is dependent on the amount of information that is desired to be uploaded onto a remote server or system. In most cases, key indicators are extracted from the collected data and are uploaded to represent features of the data. This is preferred over uploading the raw data collected because of the cost per megabyte to upload data. In this particular case, MATLAB® was utilised to process the data due to its ability to handle large amounts of data in matrix form efficiently. Due to the speed of the computer, and the efficiency of MATLAB®, data processing was performed in a matter of seconds and thus deemed to be done in 'real time'.

Uploading the data

Uploading the data may require a secure ftp (SFTP) program such as WinSCP, which is also freeware, or the use of the ftp command at the dos command window. In this particular case, a session of WinSCP was pre-configured with the information required to access a local server. In addition, a text file containing commands to open a session, load the data files, synchronise files, and close the session were created to run the ftp session.

EXAMPLE LOW COST RDAQ SETUP

The following is a description of the procedure on how the RDAQ was implemented, using SignalCalc ACE software with Quattro DAQ within a Windows 7 environment. Prior to creating and implementing the needed scripts, three software packages were needed, as given in Table 2 (all the listed software is freeware).

System Schematic

An example remote data acquisition setup is shown in Figure 1. The general system is shown schematically in Figure 2 consisting of 4 parts: a DAQ system, a computer, a wireless dongle, and a web server. The DAQ system, computer and wireless dongle are directly connected, and thus have information flowing back and forth between them. The web server could be located in a project office or this facility could be obtained from an internet service provider. The task of remotely measuring data and transferring it to the server is

Table 2. Software required in RDAQ system

	Software	Description	Download link
1	WinSCP	WinSCP is used as the SFTP or FTP program to transfer files from the computer on site to the local server. The program is freeware and provides the level of security requested by the server the data is placed on, as well as the ability to control it using a script.	http://winscp.net/eng/download.php or http://download.cnet.com
2	AutoHotKeys	AutoHotKeys is a windows scripting language, and is used as a means of clicking on parts of the screen without needing the user to be at the remote location.	http://www.autohotkey.com (tutorials provided) or http://download.cnet.com
3	Devcon	Devcon is used as a means to stop and reset a DAQ hardware in the windows environment.	http://suppport.microsoft.com/kb/311272
4*	Teamviewer	TeamViewer is a remote desktop software that needs to be placed on the laptop computer on site as well as the local computer in the office. This program can be used to login to the remote computer if it is thought to have frozen, or investigation is needed determine if the procedure is being carried out correctly on the remote computer.	www.teamviewer.com
5*	LogMeIn	Similar to TeamViewer except it can be accessed through a web browser and does not require stand-alone software.	www.logmein.com

*Optional software which was used to help to monitor the progress of the RDAQ system remotely

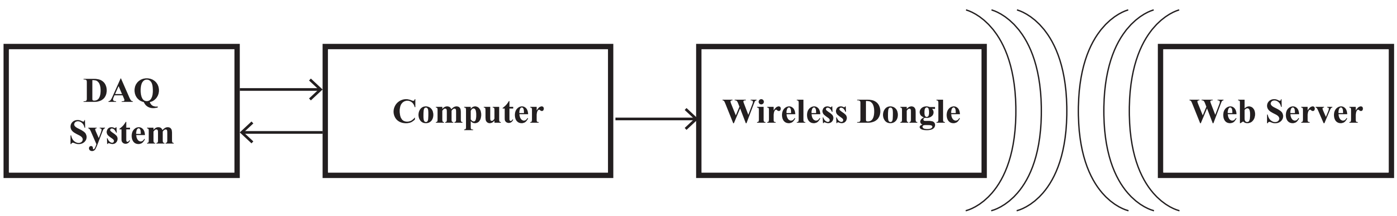


Figure 2. Schematic diagram of RDAQ System

Table 3. Main processes involved in RDAQ system

	Process	Description	Software used
1	Data acquisition	Monitors DAQ software, obtains data, controls file names and saves data to the remote computer	AutoHotKeys, Devcon, DAQ specific software
2	Data processing	Runs any processing of data before it is to be uploaded (this step is not required if the raw captured data is to be uploaded to server)	Any data processing software required e.g. MATLAB
3	Uploading data	a) Connect and monitor internet connection with USB dongle (not required if LAN connection used) b) Synchronisation of data files on remote computer and server	USB dongle software, AutoHotKeys, WinSCP

performed by utilising three main scripts to govern the flow of information; these three main scripts are shown schematically in Figure 3.

The tasks the scripts undertake are outlined in Table 3. The process can also be made to be automatic upon booting up the computer by placing the scripts into the "Startup" folder in Windows. An overall view of this 3 step process is controlled by 4 main scripts as outlined in Figure 3.

EXAMPLE IMPLEMENTATION

The example system was tested in two locations

- onsite at a remote pumping station for Gold Coast Water
- cooling water pumps for Curtin University campus wide AC system

The remote Gold Coast Water site contained two ITT Flygt CT-3231, 2 Blade, 85kW, 1475 rpm pumps which were housed in a concrete silo-like structure. These pumps were located

Data Acquisition

1. Load up DAQ software
2. If DAQ hardware can't be accessed, uses Devcon (using batch files) to reboot hardware
3. Calls "List_files.ahk" to obtain next name for data files.
4. Run DAQ
5. Save data to folder called "DAQ_files"
6. Wait for next cycle, then go to step 3.

Data Processing

1. Obtain files from folder called "DAQ_files".
2. Use own software, such as MATLAB® or scripts, to process data before uploading.
3. Save the file with a name that details the time and date of processing.
4. Place the processed data files for uploading into a directory called "current"

Uploading and Synchronising Data

USB Internet Dongle Software
Connect to the internet

1. Checks for Internet connection
   a. Pings website to see if connection is available
   b. If not, wait another cycle and try again.
2. If connected, upload all files in folder called "current" and move files to folder called "moved"
3. Synchronises folder called "moved" on local computer with server.

Figure 3. Overall process found in four main scripts

Table 4. Summary of RDAQ Sites

	Time interval left on-site	Time interval between uploads	Problems
Gold Coast Water	1 to 3 days	15 minutes	System freeze System reboot failed
Curtin University	20 hours	1 hour	Gaps in uploaded times

approximately 50 metres underground, while the computer, hard drive, and USB dongle were located in an aluminium cabinet above ground. The system was left at the site multiple times at different intervals, ranging from overnight to three days. At this location, the SignalCalc ACE system was used for data acquisition. The second location contained KSB E150-40 75kW, 1475rpm which was housed in a concrete building. Readings were taken from this location three times, each for an interval of approximately 20 hours. At this location, the National Instrument DAQ system was utilised. The results of these two locations are found in Table 4.

At both sites, data was transferred to the remote server with success. However, different problems arose within the interval during which the system was left onsite. At the Gold Coast Water site, the Windows Operating System would at times freeze after about a day of usage. To overcome this problem, remote access to the computer using LogMeIn and TeamViewer was used to check if this phenomenon occurred and to reboot the system if needed.

At the Curtin University site, data was acquired for the full 20 hours that it was left using the National Instruments' cDAQ system. However, data was not consistently being placed onto the server. Gaps in uploaded times occurred between 3am and 8am, which is assumed is due to the server shutting down during this time. As a result, after 8am, the files on the computer and the files on the server were synchronised, which caused a large amount of files to be placed onto the server in one cycle. Although not a problem to the system, it may cause problems when the server is inaccessible for long periods of time and the data uploaded is either large in size, or multiple files with a short interval between performing the data acquisition. This could be a problem since this might interfere or impinge on the resources that are allocated to another task while the system is attempting to compensate by synchronising data files.

CONCLUSIONS

There is a large need for a remote data acquisition system in industry as well as in the research community. However, most RDAQs that are available are very expensive, with the need of pre-existing networking structure to handle the data transfer either wirelessly or through a local area network. As a result, these systems are avoided due to cost and available infrastructure. This paper presents a cost efficient method of providing a RDAQ utilising systems a noise/vibration analyst would already have at hand. It is believed that this cost efficient method of providing an RDAQ would be beneficial to both industry and academic settings where the user constantly needs data but cannot constantly be present at the site.

ACKNOWLEDGEMENTS

This research was conducted within the CRC for Infrastructure and Engineering Asset Management, established and supported under the Australian Government's Cooperative Research Centres Programme. The authors would like to thank Gold Coast Water and Curtin University for facilitating the vibration measurements on the industrial water pumps used for the validation measurements.

Technical Note

Note: Technical notes are aimed at promoting discussion. The views expressed are not necessarily those of the editors or the Australian Acoustical Society. Contributions are not formally peer-reviewed.

NEW GRANDSTAND AT RANDWICK RACECOURSE

Steve Drury
The P.A. People Pty Ltd, Rhodes NSW 2138

After a 20 month total build time including a 10 month AV fit-out, the new $152 million Grandstand at the Royal Randwick Racecourse in Sydney boasts the largest integrated network AV system in Australia. Over six levels, two structures and 25,000 m^2 of floor space, a 1000 port Ethernet network carries audio to and from more than 60 separate zones, as well as sixty channels of video and thirty channels of digital signage to over 800 screens.

The Australian Turf Club engaged consulting firm Norman Disney & Young to meet the required acoustic performance criteria for each space and a number of zones, along with an overview of functional requirements and environmental constraints. The number of TV channels required and an indicative number of TV screens were also proposed.

The P.A People were employed to deliver the new networked television distribution and audio systems. The greatest acoustic challenge was maintaining a balanced audio level across adjacent zones with entirely different environments; internal to external, concrete to carpet and with varying ceiling heights. To this end, The P.A. People deployed over 1000 individual JBL loudspeakers across ten different models, including the Control Series, AWC Series, CBT Series and AE Series. This choice enabled a consistent brand 'voicing' across the site while serving each area's particular application and conditions.

The power amplification and processing is spread across eight rack rooms, with 11 racks of equipment in total. Audio signal transport between the rack rooms is achieved with Audinate's Dante run over a dedicated HP Procurve Ethernet network utilising both fibre optic and Cat6 cabling. Twelve BSS Blu series digital signal processors fitted with Dante interfaces handle audio mixing duties and distribution to zones. Signal is connected directly via Cat5 to the 80 Crown amplifiers that power the system via the Blu's built-in BluLink proprietary digital bus and PIP-Blu modules fitted to the amps.

Four control PCs are installed across the site for audio system operation and monitoring. A virtual control desk for the DSPs and amplifiers was created using the BSS London Architect software, giving the operator both overall and individual zone level control. Within the DSP, paging sources including the race caller, the Weighing Room and lines from interstate sites are automatically prioritised over background music according to a three-stage hierarchy.

Among its more than 800 screens, the Grandstand's IPTV network includes the joint largest LED screen in the Southern Hemisphere, measuring 11 metres high by 40 metres long. Two 18 metre long super screens are located on ground level betting auditorium. The system deployed is from Scottish specialist supplier Exterity. The facilities in the new Royal Randwick grandstand represent one of the largest integrated networked AV systems in Australia.

Figure 1. The Royal Randwick racecourse and grandstand

Richard Lancelot Waugh

(Dick Waugh)

9th August 1935 – 6th August 2013

Richard (Dick) Waugh was a leading researcher in the area of hearing conservation in Australia. He was passionate about the prevention of noise induced hearing loss in industrial settings.

Dick completed six months of compulsory national service. He elected to train in the Australian Airforce where he flew tiger moth aircraft at the RAAF Base in Richmond Sydney. He completed an apprenticeship as a technician with the Post Master Generals Department. He is fondly remembered by his fellow trainees as a leader in his group – often mentoring and supporting his fellow trainees. He won the medal for Apprentice of the Year in 1953. From these beginnings as a field worker installing equipment in conditions that were often hazardous, Dick developed an interest in the social and economic conditions that affected workers in heavy industry and the lack of research and policy about workplace safety.

Dick studied psychology and philosophy graduating from the University of Sydney in 1965 with second class honours. His career as a research psychologist and advocate for safer working conditions to address industrial deafness began with his work as Psychologist in Charge of the Hearing Conservation Group in the National Acoustic Laboratories of the then Commonwealth Department of Health. Dick also graduated from the University of Sydney in 1990 with a Masters of Public Health. For his research thesis he developed and conducted a hearing conservation program for small industry workers. Dick was a pioneer in the area of hearing conservation and highlighted the incidence and prevalence of noise induced hearing loss in the Australian workforce. He argued that workers needed to be aware and informed of the burden of noise induced hearing loss and given the tools to protect their hearing at work. While the political debate raged about the difficulties in getting workers to wear cumbersome and often ineffective hearing protection vs the reluctance of employers to reduce workplace noise, Dick pointed out succinctly that "A more constructive approach... might be to study the actual conditions of those who work in noise as a basis for working out realistic policies" (Waugh 1984 Safety in Australia). In this context, the responsibility of employers to reduce workplace noise is a key issue: the use of hearing protection devices by the worker is often not realistic, effective or safe.

Dick moved to the Information and Preventive Services Branch of Worksafe Australia and began work on national policy to address hearing conservation issues. He was active in a number of professional committees to implement standards for hearing protection and to raise awareness of noise induced hearing loss. In a number of international presentations and articles, Dick advocated for those affected by noise induced hearing loss as a consequence of workplace conditions. He was highly critical of the low importance given to the whole area of workplace induced hearing loss – from the poor compensation for workers disabled by hearing impairment, the lack of adequate national standards for hearing protection and workplace noise safety and the reluctance of employer groups to address noise reduction in the workplace as an occupational health and safety issue.

Dick was the brains behind the Worksafe Noise Management at Work Control Guide and related materials. Although developed more than 20 years ago, it still remains a definitive guide to noise control in the workplace and nothing really new has been developed in this space since Dick did this work. Similarly, he did the policy work and internal advocacy work which saw the workplace noise exposure standard change from 90 dB to 85 dB (that equates to a halving and almost a halving again of the legal level of noise exposure; decibels being a log scale). This is not a small achievement and serves as a testament to his professional and tenacity that he was able to contribute to such significant change – that it was worth the effort he put in even if he personally didn't reap the benefits of the changes secured.

Dick retired from active full time work when the Howard Government severely curtailed Worksafe Australia. He continued to advise on standards for hearing protection and to work with other professional colleagues to continue to raise awareness of the importance of hearing conservation amongst workers and employers.

Dick is remembered fondly by his family – Hilary, Guy, Meg and Pamela – as a gentle compassionate father, friend and partner who cared deeply about the people he loved.

Dr Meg Smith

Mr Jack Rose F.A.A.S.

21st July 1919 - 29th October 2013

Jack Rose commenced his career in the field of acoustics when he joined the Commonwealth Acoustic Laboratories (CAL now known as National Acoustic Laboratories NAL) in 1948. Previously he had been a cadet engineer in AWA where he obtained wide ranging experience in the development, design and manufacture of electronic devices mostly in the radar and wireless fields. On leaving AWA he worked on the design and manufacture of transformers at Transmission Products, a small Sydney based company.

When Jack joined CAL it was in its infancy. It was being established as part of the Commonwealth Department of Health to carry out scientific investigations related to hearing aids and their use and problems of noise as it affects people. Jack was part of small team of engineers which designed and built facilities for hearing testing, development and servicing of hearing aids, noise measurement, calibration of equipment and administration in Erskine House, York St. Sydney. Because of his background in development and manufacturing of electronic equipment, he worked with a team designing and making the early prototypes and drawings of the first valve operated hearing aid to be manufactured for the government. His work was extended to include the planning and design of facilities for hearing testing and aid fitting services being developed in capital cities in each State.

Early in the 1950's Jack participated in an extensive survey of hearing impairment of workers and their noise exposures in a number of railway workshops and ship building and repair shops around Sydney. So that hearing tests could be conducted in these inherently noisy sites, Jack designed a small acoustic booth which could be assembled onsite from sound insulating panels. This enabled some 500 workers to be audiometrically tested in appropriate acoustic conditions on site. The results showed an extremely high incidence of hearing impairment which lead to CAL increasing efforts to promote conservation of hearing in industry. Over time Jack and others from CAL investigated a range of industries and developed more practical methods of measuring noise and assessing hearing impairment amongst workers. Jack became conversant with noise control methods, including personal hearing protection and because of his early experience in industry and engineering background was very suited to liaising with industry to encourage protection of workers. He also became involved in investigating problems of noise in the armed services. As his expertise became more widely known, he was frequently called upon for advice on noise control and hearing protection from industry and Departments of Occupational Health and Safety in most States of Australia as well as acoustic consultants. His work and that of others at CAL resulted in the design of a hearing conservation program to combat the effects of noise exposure in industry. This was published in 1962 in a report by CAL entitled "Hearing Conservation in Industrial Noise". Jack was a co-author. He became very dedicated to this work and enthusiastically promoted hearing conservation programs wherever he could. Today efforts to reduce the effects of noise on hearing are common place and follow on from this early work. Jack by his efforts must share some credit for this.

During the latter part of the 1950's Jack became involved with others from CAL in an extensive survey of aircraft noise around Sydney Airport. During the survey he took a special interest in efforts being made here and in other parts of the world to achieve some relief for communities exposed to flyover noise. As a result of this he was able to assist local civil aviation authorities who were starting to receive numerous complaints by helping them to develop strategies to mitigate the problem. By the late 1960's aircraft noise was receiving world-wide attention and a special meeting of the International Civil Aviation Authority (ICAO), stimulated by Australia, was devoted entirely to this. Jack joined Australian Department of Civil Aviation officers in attending the conference which was regarded by them as highly successful as a large number of their recommendations were agreed to. Also from about 1968, Jack was seconded to the Federal Parliamentary Select Committee on Aircraft Noise. The Chairman, Mr Buchanan, when presenting the report of the Committee, commenced his statement to Parliament by saying "Australia has been at the vanguard of world-wide efforts to achieve some relief from exposure to aircraft noise." He went on to say "The Select Committee has been fortunate in having as technical adviser Mr J. A. Rose of the Commonwealth Acoustic Laboratories, undoubtedly the foremost scientist in his field in Australia. He represents this country with singular distinction at the International Standards Organisation and ICAO on matters of aircraft noise and other acoustic problems. At their conference, Mr Rose is highly respected by overseas authorities for his outstanding knowledge of his subject and his representation of this country. The Committee is deeply indebted to him for his unselfish dedication to this inquiry." Jack was also involved in the Major Airport Needs of Sydney study in 1977-79.

In 1964 Jack supported Mr Peter Knowland's suggestion that an association of people working in the various fields related to acoustics should be formed. He took a leading role in promoting the idea which received significant support from many working in the field in Melbourne and Sydney. Regular meetings were arranged not only to allow exchange of information and ideas on technical matters but also to present opportunities for social interaction which many enjoyed. Jack and Peter promoted this as a special feature of these early meetings usually involving a dinner in which member's partners often participated. Because of his previous experience in a bush walking club, Jack played a significant role in guiding the association towards registration and in 1970 the Australian Acoustical Society became incorporated. In 1971 he became President. He realised the importance of gaining international collaboration and recognition for the Society and its members and together with Professor Anita Lawrence succeeded in persuading the International Congress on Acoustics (ICA) to hold their 10th Congress in Sydney. Jack was appointed Chairman of the Executive Committee to plan and organise the event which was a huge undertaking for a fledgling society with limited funds. The Congress which was held in 1980 attracted over 800 acousticians from around the world, was very successful. For his services to the Society Jack was awarded the grade of Fellow.

In the last few years of his career at NAL Jack headed the Noise Investigation Section. In this role he continued to guide staff on a range of investigations including a major study of community reactions to aircraft noise which was completed in 1982. Following a study tour of important acoustic testing facilities in Europe and the U.S.A. he developed the specifications and design of world class standard anechoic and reverberation rooms included in the new laboratories built at Chatswood for NAL. Although he retired before the laboratories were completed in 1986 he was very proud they achieved the standards he aimed for.

It is evident Jack had a very distinguished career in the field of noise. He must be given a lot of credit for saving many people from the insidious effects of noise on their hearing and for giving many others some relief from aircraft noise. The Society extends its condolences to Jack's wife Betty and his daughters Sue, Chris and Kathy and their families.

Ray Piesse

Jack Rose was part of the Society since Day One

You may well ask – When was Day One? For some it could be when I sat opposite Jack Rose at his desk in the Commonwealth Acoustics Office in Customs House, Sydney. I asked Jack "What does he think of the few of us that are interested in acoustics forming some sort of association?" Jack said that we should do it. It was important to have Jack involved as at that time, in the Sydney scene he was "Mr Acoustics", the person who seemed the most knowledgeable, the most competent and the most respected in the field of acoustics.

Jack said that he would help put together a list of names to complement the names I had on my list. The next stage was to get these potential members together. I organised a meeting at my employer's office, Norman & Addicote at 48 Mitchell St, Artarmon for the 5th August 1964 and this was attended by 16 like-minded people. At this meeting many spoke of the need to be in some form of association where we could share knowledge and experience. Of the many speakers, Jack Rose was the voice that helped unite those in that room. Jack had a wonderful smile that encouraged you to come on board.

So at the meeting it was agreed that:-

- We would start the preparation of an association or society for the benefit of those involved in the field of acoustics.
- We would encourage a similar society in Victoria with the aim to form a federal society. Fortunately we had in attendance at the meeting Hugh Vivien Taylor who was the true pioneer of acoustic consulting in Australia. He was operating in both Sydney and Melbourne and therefore was in a good position to plant the seed for a Melbourne division.

Many of you would be thinking that this was really Day One. Yes, we can say that but the very important day in the history of the Society could not have occurred without that list of people that we should approach.

From that first meeting on 5th August 1964 emerged a group that would be the basis of an association for those involved in the field of acoustics. We must also be thankful to Norman & Addicote as they provided the venue so that we could get together and help the potential society to grow.

The second meeting of the NSW group was held on 23rd September 1964 which included the original 16 people and fortunately John Irvine and Anita Lawrence came on board. Together with Jack Rose and others the infant society now had a formidable core that would help to bring this much needed society to fruition. Meanwhile we were gladdened to learn that the progress of a Melbourne division was well underway. The Society in those years had a wonderful spirit, very enjoyable times were spent at Anita Lawrence's house putting the proposed Constitution together and then our efforts would be rewarded later in the day by good food and good wine. It could be said that we were a combined acoustic and wine and food society.

The funeral of Jack Rose was held on Friday 1st November at Macquarie Park, North Ryde. Past and present Society members Peter Alway, Ray Piesse, George Kimpton were there. I asked whether I could represent the Society on this occasion. I can say that I was proud to be there as a past Chairman of the Society. I was saddened by the poor representation by the Members of the Australian Acoustical Society however keeping in mind some members are no longer with us and the notification of the funeral did not give a great deal of time for some to rearrange their schedules.

Jack Rose had put so much effort into helping the Society to be born and to grow in those important first years. I feel that Jack Rose needs recognition for what he has done. The Society owes him so much. I hope the Society will continue to remember this contribution and suggest that Council considers some enduring award named after Jack Rose.

Peter Knowland MAAS

News

AAS Annual Conference

The 2013 AAS annual conference, with the title Science, Technology and Amenity was held at Victor Harbor, South Australia, 17 to 20 November. Congratulations to Peter Heinz and his team from the South Australian Division for the excellent preparation and the successful conference. There were 183 participants and 27 exhibitors. The conference program comprised two streams with 84 contributed papers plus two plenary presentations and three keynote presentations. The plenary speakers were Dr G.P. Frits van den Berg of the Environmental Health Centre in the Netherlands and Tim Duda of the Woods Hole Oceanographic Institution in the USA. The keynote presenters were Ian Bedwell, Con Doolan and Kerstin Persson Waye.

The President's Prize for the best paper presented at the conference was awarded to William Robertson with colleagues Ben Cazzolato, Anthony Zander, from the University of Adelaide for their paper *Planar analysis of a quasi-zero stiffness mechanism using inclined linear springs*.

Excellence in Acoustics Award

The prestigious AAS Excellence in Acoustics Award award supported by CSR Bradford Insulation aims at fostering and rewarding excellence in acoustics. The entries are judged on demonstrated innovation from within any field of acoustics. The prize is a trophy and a gift to the value of $1500 to the winner.

The award for 2013 has been made to the team of Jonathan Cooper and Tom Evans from Resonate Acoustics and Dr Dick Petersen from the University of Adelaide. Their project was titled *Detailed tonality assessment procedure for a wind farm*. The project is described in the following executive summary of their submission:

"Current assessment procedures for tonality from wind turbines only require a limited amount of data points to be assessed in the downwind direction, and are complicated to implement at a residence due to the requirements for narrowband analysis and exclusion of extraneous noise. Therefore, previous studies have typically involved the assessment of individual data points rather than longer periods of data.

In response to a recent complaint from a resident near a wind farm in South Australia that related to the character of the noise, we developed and implemented a procedure that utilised continuous audio recording to enable tonality to be assessed at the residence over extended periods. A tone was identified that occurred in excess of the criteria at night and during upwind conditions. The supplier was able to subsequently make modifications to the turbines, completed recently, and the resident noted an improvement in the character of the noise."

iPad Mini winner at Acoustic 2013

Congratulations to Radek Kochanowski from NSW Transport who won the iPad mini at the SAVTek stand during the Australian Acoustical Society Conference at Victor Harbor (17-20 November 2013).

New Fellows for the AAS

The following Members of the Australian Acoustical Society have recently been awarded the grade of Fellow

- Norbert Gabriels, WA Division
- Gillian Adams, QLD Division
- David Mee, QLD Division
- Peter Heinze, SA Division

The citations for each of these new Fellows of the AAS are as follows.

Norbert Gabriels qualified as an architect in 1971 (Bachelor of Architecture, University of WA) and became Architects Board of WA registered as such in 1974. He has been working full time in the area of acoustics and the building environment for the last forty years since. In 1976, Norbert was the Manager of the Environmental Design Branch in the State Public Works, where for 18 years he was responsible for the environmental and acoustic design of all State Government projects. In this capacity he was involved in the acoustic design of numerous major civic and institutional projects throughout the state. Following this role he established Gabriels Environmental Design in 1994, a member firm of the Association of Australian Acoustical Consultants since 1996 to which he still holds the position of Director.

Norbert is currently chairman of the Australian Standard's committee on architectural acoustics (AV/4), responsible for all Australian Standards relative to the performance, measurement and assessment of acoustics in buildings, a position held since 1988. Norbert is also the Australian representative on the International Standard Organisation committee ISO TC43 on Building Acoustics, involved in the writing and review of International Standards. His extensive professional experience includes a variety of landmark projects in Western Australia. Norbert has been a member of the Australian Acoustical Society since 1976, and was chairman of the WA Chapter of this body, from 1993 to 1995. Over the last two decades, he has been a tireless supporter of Western Australian Division events and has contributed to committee function and activities.

Gillian Adams graduated from The University of Queensland in 1991 with a Bachelor of Engineering degree, majoring in civil engineering. In 1998 she obtained a Master of Science (Environmental) from Griffith University. Gillian has been a practicing consulting acoustic engineer in Queensland since graduating from university. She has worked on a broad range of major acoustic projects across Queensland and has risen to the top of her profession as the Managing Director of ASK Consulting Engineers. Gillian has worked tirelessly to promote the role of women in engineering, giving numerous presentations to high school and university students across Queensland.

Between 2007 and 2013 Gillian was a guest lecturer in acoustics for interior designers and architects at the Queensland University of Technology. She was also a guest lecturer for Central Queensland University's Environmental Master's Program in 1998. In 1995/1996, Gillian was involved with the research and development of a test trailer capable of conducting dynamic measurements of tyre/road noise. As a result of this research Gillian won several awards for innovation that increased the profile of acoustics and women in engineering in Queensland. Between 1994 and 2003 Gillian served on the Queensland Division Committee of the Australian Acoustical Society. She was the Queensland Divisional Chair and the Federal Treasurer of the Acoustical Society between 1999 and 2001. Gillian has been appointed to the Grade of Fellow for her dedication in furthering the development and increasing the profile of acoustics, encouragement of women to consider engineering as a career and in recognition of the contribution she has made to the Australian Acoustical Society.

Professor David Mee graduated with a bachelor degree in engineering from the University of Queensland in 1982. After obtaining his PhD in 1986, he commenced his academic career conducting research on behalf of Rolls-Royce at the University of Oxford. In 1991 he returned to the University of Queensland where he started teaching undergraduate students. In 2001 Professor Mee Joined the Australian Acoustical Society and started teaching the third-year undergraduate course MECH325O Engineering Acoustics at the University of Queensland, That same year Professor Mee was elected to the Queensland Divisional Committee of the Society, on which he has served continuously since that time. As a member of the Society he was on the organizing committee for Acoustics 2004 and Acoustics 2011 and was co-editor for the proceedings of both of these conferences. He has represented the Society at the Science Teachers' Association of Queensland Science Contest to judge the contest entries and to award the contest winner with a bursary sponsored by the Queensland Division.

Professor Mee has promoted acoustics as a career option to his students and as a result many of his past students are now consulting acoustic engineers in Queensland. He has supervised numerous final year undergraduate research theses on acoustics and encouraged his students to apply for the research bursaries that have been offered by the Queensland Division. Professor Mee has been appointed to the Grade of Fellow for the service he has given the Australian Acoustical Society and the efforts he has made in advocating acoustics as a career option for engineering students.

Peter Heinze graduated from the University of South Australia with a Bachelor of Engineering. He started working for various organisations including BHP, Department of Defence, Telecom, Pryce Goodale & Duncan/ Bassett and currently works for Marshall Day Acoustics. In his professional life he has been a consultant across a wide range of projects along with the important role of being a teacher and mentor to many consultants working in the industry today. Peter has had a very long an active involvement with the AAS including Federal roles as Federal Treasurer, General Secretary, Vice-President and President of the Society. Within the SA Division he is also the primary organizer of AAS 2013 in Victor Harbor and has been the SA Division Chairperson and Federal Councillor.

Educators in acoustics group

One of the objectives of the AAS is "To promote education in acoustics for the public, and for technical or professional persons." There are many people in Australia involved in the education of acoustics that play a vital role in the education of the next generation of acousticians. At the time of the Acoustics 2013 conference in Victor Harbor, a group was established called Educators in Acoustics. The group intend to share information, resources, discuss successful (and unsuccessful) teaching methods, organise guest lectures either in person or video conference. If you are involved in acoustics education and would like to join this group, please contact Carl Howard, carl.howard@adelaide.edu.au, tel (08) 8313 5460 at the University of Adelaide.

International keynotes by AAS members

During 2013 the Australian accent has been on the main stage for plenary/distinguished presentations at two major international conferences. These presentations are important parts of the technical program for the conferences and it is an honour and recognition of standing in the field.

At the 20th International Congress on Sound and Vibration conference (ICSV20) in Bangkok, Thailand in July 2013, **Nicole Kessissoglou** from the University of NSW, Sydney presented a keynote paper title *Numerical prediction of the signature of maritime platforms*.

At Inter-Noise 2013 in Innsbruck, Austria in September 2013, **Marion Burgess** from the University of NSW, Canberra presented a plenary paper titled *Community noise management and control: successes and challenges*.

This trend for major presentations by Australians at international conferences is continuing with the 21st International Congress on Sound and Vibration conference (ICSV21) to be held in Beijing, China in July 2014. **Jie Pan** from the University of Western Australia will be presenting a keynote paper on *Acoustics of ancient Chinese chimes*. At Inter-Noise 2014 in Melbourne in November 2014, **Lex Brown** from Griffith University will be presenting a plenary paper titled *Soundscape planning as a complement to environmental noise management*.

New open-access acoustics journal

The first issue of *Acoustics in Practice*, the new open-access journal of the European Acoustic Association (EAA), has been published. The aim is to cover the practical aspects of all areas of acoustics and to provide a Europe-wide platform for authors to disseminate their work. To download the journal go to http://euracoustics.org/AiP/1_1_2013/

Articles on wind turbine noise

A number of articles on wind turbine noise and concerns about infrasound from wind turbines have been recently published in *Acoustics Today*, and can be downloaded at
http://dx.doi.org/10.1121/1.4821143;
http://dx.doi.org/10.1121/1.4827009;
http://dx.doi.org/10.1121/1.4821142

Increase in AAS membership dues

Federal Council has increased the 2014-2015 membership fees for Member and Fellow grade to $150 inc GST. This is a 15.4% increase and is the first increase since 2007. Other grades increase proportionally. Amongst other things, the increase will assist with the AAS recent initiatives including the upgrade of the AAS website, which allow greater ease of use and improved functionality, as well as the AAS Research Grants. The new fee structure for the year 2014/215 is as follows.

Grade	Fee ($)
Associate	115
Graduate 1	115
Graduate 2	127
Graduate 3	138
Graduate 4	150
Maternity	45
Member/Fellow	150
Retired	45
Subscriber	115
Student	0
Honorary	0
Life	0
Retired Fellows	0

Standards Australia

AS/NZS 2107 review to start

In 2013 the Australian Acoustical Society submitted an application to Standards Australia Project Priority Assessment for a project to be established to review AS/NZS 2107 *Recommended design sound levels and reverberation times for building interiors*. This is a well used standard and the application for the review sought to bring the guidance up to date as the current version was published in 2000. In order to submit this application it was essential to obtain support for the initiation of the review from the wide range of stakeholders. Advice has been received that the application has been successful so the work on this review will commence by the committee in 2014. If you have any comments or suggestions in relation to this review please advise Norbert Gabriel who is the chair of the relevant committee and Marion Burgess who prepared the application on behalf of the AAS; norbert@gabriels.net.au; m.burgess@adfa.edu.au

AS 2021 review in progress

In 2013, Committee EV11 of Standards Australia commenced a review of AS 2021 *Acoustics - Aircraft noise intrusion - Building siting and construction*. The current version of this standard we published in 2000 and it provides a critical framework for managing land use and development outcomes in the vicinity of airports. The review of parts of this standard has commenced and it is currently still going through the committee drafting stages. Once this has been completed there will be a public comment period.

AS/NZS 1269.4 review

AS/NZS 1269 is an important standard for occupational noise measurements and assessments. The review of *Part 4, Auditory Assessment*, focuses on updating in line with current best practices and equipment. The work on this review commenced in early 2013, it has been through the public review process and it is close to a final version of Part 4 being produced in early 2014. Warwick Williams in the chair of the committee and has been the leader for this project.

New Products

Stereoscopic 3D View in CadnaA Version 4.4

The last step of an acoustical consultation is the presentation of measures and effects to a third party which is often not an acoustician. Together with a 3D enabled screen (3D

Monitor, 3DTV) and 3D passive glasses, CadnaA projects can now be presented in a realistic 3D format. The new features of the program are

- Stereoscopic 3D (video)
- Apply 3D Objects (trees, cars, trains etc.) to provide a clear overview for third parties

In addition, the following new noise calculation standards are available as an option

- Aircraft noise calculation with INM 7.0 and ECAC Doc.29 3rd Edition (video)
- Industry noise calculation with NMPB08-Industry and Nord 2000 (version 2013)
- Rail noise calculation with NMPB08-Fer and Schall03 (2013)

Visit www.datakustik.com for more detailed information or contact Rodney Phillips at Renzo Tonin & Associates, tel (02) 8218 0500 or email rphillips@renzotonin.com.au

Wireless accelerometers

MEDA RedSens is the only available WiFi-measuring-system with a 100% synchronicity for all attached channels – up to 10 triaxial units can be used at once. Ranges of up to 150 m can be reached and the system, which covers a band width up to 4 kHz, lends itself to every application in engineering, building vibrations and in the automobile industry. With 24 bit resolution and 110 dB dynamics even the tiniest indicators can be captured. The possibility of preselection between two input ranges allows the optimal adaption of MEDA RedSens to a variety of applications. For further information contact Darryl Watkins, SAVTek, (07) 3300 0363; enquiry@savtek.com.au, www.savtek.com.au

New architectural timber modules

Screenwood has developed architectural linear timber modules to include Acoustic Solutions. Commonly used in large commercial fitouts, education and medical facilities, Screenwood is manufactured from solid PEFC certified timbers in a variety of profiles and finishes. These linear timber modules are prefinished, assembled and cut to length in factory allowing for a swift installation upon arrival onsite. In response to the increasing incidence of acoustic requirements, Screenwood now includes a black acoustic scrim (SC15) with all orders. This backing is a factory-fitted no-cost addition to the product. For higher NRC requirements a thick polyester backing (SCP50) is added for a minor additional cost. For more information visit www.screenwood.com.au or tel (02) 9521 7200

Divisional News

NSW Division

On 19 August, Marion Burgess, UNSW Canberra, gave a presentation titled *Community noise management and control: A discussion on some successes and challenges*. Over recent decades there have been some clear achievements in the acknowledgement of the importance of addressing noise the community. The focus has been on the major noise sources associated with transportation and industry that globally affect the larger number of people. The publication of guidelines for noise level limits and for establishing noise control policies and approaches to noise management provides a good basis for further applications. This talk discussed some of the successes and also some of the remaining challenges in developing and adopting the most appropriate noise management and control policies.

The AGM of the NSW Division was held on 28 October. After the AGM a technical talk on the topic *Childcare Centre Noise: Where did we get to?* was given by three speakers: Stephen Gauld (Day Design), Suri Mora (Wollongong City Council) and Michael Gange (Renzo Tonin & Associates). Six years ago the NSW division held a discussion panel on *Are we assessing childcare centre noise fairly?*. Since that time there have been some key additions and changes to the planning and assessment guidelines in relation to childcare centre noise. Many Council DCPs include guidelines for childcare centres. The Australian Association of Acoustical Consultants (AAAC) has published a guideline recommending a method of assessing noise impact in relation to childcare centres. The revised Noise Guideline for Local Government provides references to assist Councils in formulating a position on child care centre noise. A panel of three people was assembled to present a different perspective on assessing noise from childcare centres based on their own experiences in recent years. Stephen Gauld was largely responsible for the preparation of the AAAC guideline. Suri Mora is an Environment Assessment Officer with Wollongong City Council and in this role has been involved in the planning process for many childcare centres. Michael Gange has been involved in the assessment of many childcare centres.

The end of year Christmas breakfast was held on 4 December at AECOM on George Street, Sydney. An interesting talk was given by Ron Carpenter. Ron is a composer, multi-instrumentalist, band leader and touring musician, recording artist with AC DC/Cold Chisel, session musician, synthesist and programmer, producer and engineer, one-man band, vocalist, sound designer, futurist and sound spatializer. Ron has also been a teacher of music for over 30 years. He has worked innovatively in isolation on self-funded projects, manipulating DSP technology to best express/manifest/showcase his multiple skills and 3D sonic abstractions.

VIC Division

The VIC Division celebrated Christmas on 3 December at the Society restaurant in Bourke Street Melbourne. This was a change from our usual haunt and one that was welcomed by many members, with a record number of city based consultants turning up. Good representation was made by consulting firms Arup, Aecom, SLM, Acoustical Design, Audiometric and Acoustic Services, Aurecon and SLR. Industry was represented by Boral and Embelton. John Davy valiantly and singlehandedly stood in for all of Victorian academia and research. It was great to catch up with Graeme Harding and Michael Gregory, who are enjoying retirement after illustrious careers in noise and vibration, and Winston Vaz who is moving into a new phase of his noise and vibration career with his own company VazCorp.

SA Division

The SA Division of the Australian Acoustical Society held its 37th Annual General Meeting on 18 September 2013. The AGM was followed by a technical talk entitled *Recent developments in wind farm noise in Australia*. The talk was presented by Chris Turnbull, the principal acoustic engineer of Sonus, an acoustic consulting firm he formed 11 years ago. Chris has 20 years of experience in acoustics and has developed a particular interest in wind turbine noise. Chris' talk summarised the current state of environmental noise assessments for wind farms in Australia, highlighting: the common concerns raised during applications for proposed wind farms; the reaction of regulators to controversial aspects such as additional requirements for noise assessments, minimum setback distances between turbines and residences and comprehensive post construction monitoring regimes; and the reaction of professional acoustic bodies.

The SA Division were pleased to host the Society's annual conference *Acoustics 2013* in November at Victor Harbor. The committee would like to take this opportunity to thank the conference chair, Peter Heinze, who took on the lion's share of the organisation. His efforts and co-ordination resulted in a conference which ran smoothly, receiving positive feedback from many of the attendees.

The SA Division ended 2013 with its annual Christmas Dinner, which was held at Chianti Classico on Friday 29 November 2013. A highly enjoyable evening was had by all that attended.

WA Division

The Western Australian division held their annual State Seminar on 16 October, at the John Worsfold Function Room at Patersons

Stadium. The seminar included nine interesting presentations by AAS members, covering both underwater and air acoustic topics. The seminar also incorporated the state AGM.

On 14 November the WA division hosted Professor Gary W. Evans at the offices of NDY for an interesting technical presentation titled *The effects of environmental noise on children's health and cognition*. Professor Evans, of Cornell University New York State, gave an informative presentation regarding the non-auditory impacts of noise on children – the effects not caused by hearing damage. The WA division also sponsored a presentation by Professor Evans on 15 November for non AAS members, which was well attended by the Department of Health, Department of Education, and universities.

The WA Division held its annual Christmas Lunch on Friday 13 December at The Brown Fox in West Perth.

FUTURE CONFERENCES

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 are included in the registration and will be provided in the middle of the exhibition area, permitting good interaction of delegates and exhibitors. Already over 40 booths have been booked so intending exhibitors should contact Norm Broner [NBroner@globalskm.com] about securing one of the remaining locations.

After the welcoming ceremony on Sunday 16 November the first plenary lecture is on a newly developing area, *Sound Sketch: its theory and application using loudspeakers* by Professor Jung-Woo Choi from Korea. The four keynote lectures will span a range of topics including Aircraft Noise, Wind Turbines and LFN, Active Noise Control, and the impact of Building Acoustics on Speech Comprehension and Student Achievement. The second plenary talk by Professor Lex Brown from Australia on *Soundscape planning as a complement to environmental noise management* will be just before the closing ceremony. Over 120 specialists from around the world, including Australia and New Zealand, have agreed to chair sessions during the Congress. About 100 potential sessions are listed on the congress website, including transport noise and vibration, gas-turbine and jet-engine noise, computational aero-acoustics, signal processing for active noise control, education and policy, modern acoustic materials, low-noise tyres and underwater noise.

Abstracts should be submitted through the Congress website: www.internoise2014.org, before the closing date of 10May. This is a firm date and will not be extended. Final papers are required by 25 July. Details of the paper format and submission procedures are given on the website.

ICSV21 2014

The 21st International Congress on Sound and Vibration (ICSV21) will be held from 13-17 July 2014, at the China National Convention Center (CNCC), 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 are also welcome. The ICSV21 Scientific Programme will include invited and contributed papers and seven plenary presentations. There is an option to have your full paper refereed. Full paper(s) should be submitted by 31 January 2014. For further details visit www.icsv21.org

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 www.isma-isaac.be

Diary

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/

8-10 September, Fort Lauderdale, Florida
Noise-Con 2014
http://www.inceusa.org/nc14

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

9-12 August, San Francisco, USA
Inter-Noise 2015
http://internoise2015.com

2016

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

10-14 July, Athens, Greece
23rd International Congress on Sound and Vibration (ICSV23)
http://iiav.org/index.php?va=congresses

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

BARRISOL®

Les Acoustics® - Microperforated Stretch Ceilings

Barrisol® Acoustics® Lumière®
Federation Square arch. :
Lab + Bates Smart acoustics :
Marshall Day

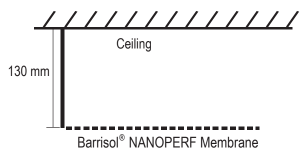
Barrisol ceilings with invisible microperforations provide exceptional acoustic performance without compromising your design.

Each perforation is just 0.1mm diameter, with up to 500,000 microperforations per square meter. Ceiling panels are not restricted to fixed panel sizes or shapes, with single custom ceiling panels up to 50 square meters in size.

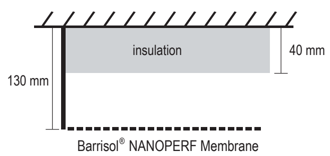
The sound absorption works by converting sound energy into thermal energy through friction with the microperforations. The friction is increased by the resonance of air within the cavity between the microperforated membrane and ceiling.

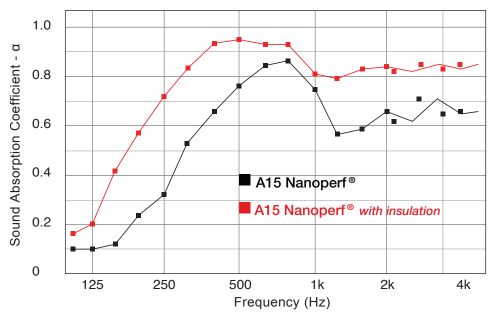
Barrisol microperforated acoustic solutions are available across the entire Barrisol range of 230 colours and 18 finishes, including gloss, satin, matt, translucent and recycled.

A15 NANOPERF® *without insulation*



A15 NANOPERF® *with insulation*





A15 Nanoperf®	A15 Nanoperf® *with insulation*
NRC : 0.62	NRC : 0.83

Barrisol® Acoustics - A20 Acoperf®
Oslo Opera House
4000m², matt white ceiling panels (60m²/panel)
architect : Snohetta Architects
acoustics: Arup
2009 European Award for Contemporary Architecture

Barrisol® Acoustics - A15 Nanoperf®
Marquee Nightclub, The Star, Sydney
High gloss black acoustic curved ceilings
architect : Squillace Nicholas Architects
acoustics : AECOM

Barrisol® Lumière® Acoustics - A30 Microacoustic®
London Aquatics Centre
139 backlit translucent acoustic petal shaped panels
architect : Zaha Hadid
acoustics : Arup

Barrisol® Acoustics - A15 Nanoperf®
Bucharest Sunplaza - Romania
10,000m² Barrisol rectangular ceiling panels
finish: gloss, matt, satin
architect : Chapman Talyor

www.barrisol.com.au

melbourne 03 98790628 **sydney** 02 96606044 **brisbane** 07 38510055 **adelaide** 08 82926600 **perth** 08 92482245 **nz** +64 9 525 6575

SUSTAINING MEMBERS

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

3M AUSTRALIA
www.3m.com

ACOUSTIC RESEARCH LABORATORIES
www.acousticresearch.com.au

ACOUSTICA
www.acoustica.com.au

ACRAN
www.acran.com.au

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

ADAMSSON ENGINEERING
www.adamsson.com.au

AERISON PTY LTD
www.aerison.com

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

BARRISOL AUSTRALIA
www.barrisol.com.au

BORAL PLASTERBOARD
www.boral.com.au/plasterboard

BRUEL & KJAER AUSTRALIA
www.bksv.com.au

CSR BRADFORD INSULATION
www.bradfordinsulation.com.au

EMBELTON
www.vibrationisolation.com.au

HOWDEN AUSTRALIA
www.howden.com.au

IAC ACOUSTICS (AUSTRALIA)
industrialacoustics.com/Australia

INSTYLE CONTRACT TEXTILES
www.instyle.com.au/#

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

PEACE ENGINEERING
www.peaceengineering.com

PYROTEK NOISE CONTROL
www.pyroteknc.com

REGUPOL (AUSTRALIA)
www.regupol-vibration-technology.com.au

SINCLAIR KNIGHT MERZ
www.globalskm.com

SOUND CONTROL
www.soundcontrol.com.au

SOUNDSCIENCE
www.soundscience.com.au

VIPAC ENGINEERS AND SCIENTISTS
www.vipac.com.au

AUSTRALIAN ACOUSTICAL SOCIETY ENQUIRIES

NATIONAL MATTERS

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

Richard Booker - General Secretary
Australian Acoustical Society
PO Box 1843 Toowong DC QLD 4066
Tel: (07) 3122 2605
email: GeneralSecretary@acoustics.asn.au
www.acoustics.asn.au

SOCIETY SUBSCRIPTION RATES

For 2013/14 Financial Year:

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

Including GST

DIVISIONAL MATTERS

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

AAS - NSW Division
Laura Allison
c/- AECOM
Level 21, 420 George Street
Sydney, NSW 2000
Tel: (02) 8934 0035
Fax: (02) 8934 0001
Laura.Allison@aecom.com

AAS - Queensland Division
PO Box 760
Spring Hill Qld 4004
Sec: Richard Devereux
Tel: (07) 3217 0055
Fax: (07) 3217 0066
rdevereux@acran.com.au

AAS - SA Division
AECOM,
Level 28, 91 King William St
ADELAIDE S.A. 5005
Sec: Darren Jurevicius
Tel: (08) 7100 6400
Fax: (08) 7100 6499
darren.jurevicius@aecom.com

AAS - Victoria Division
c/- Simon de Lisle
Arup Acoustics
Level 17, 1 Nicholson Street
Melbourne VIC 3000
Tel: (03) 9668 5580
Fax: (03) 9663 1546
simon.delisle@arup.com.au

AAS–WA Division
Unit 3
2 Hardy Street,
SOUTH PERTH 6151
Sec: Norbert Gabriels
Tel (08) 9474 5966
Fax (08) 9474 5977
gabriels@iinet.net.au

ACOUSTICS AUSTRALIA INFORMATION

GENERAL BUSINESS

Advertising Subscriptions

Mrs Leigh Wallbank
PO Box 70, OYSTER BAY 2225
Tel (02) 9528 4362
Fax (02) 9589 0547
wallbank@zipworld.com.au

ARTICLES & REPORTS NEWS, BOOK REVIEWS NEW PRODUCTS

The Editor, Acoustics Australia
c/o Nicole Kessissoglou
School of Mechanical & Manufacturing Engineering
University of New South Wales
NSW 2052 Australia
Mobile: +61 401 070 843
acousticsaustralia@acoustics.asn.au

PRINTING, ARTWORK

Cliff Lewis Printing
91-93 Parraweena Road
CARINGBAH NSW 222
Tel (02) 9525 6588 Fax (02) 9524 8712
email: matthew@clp.com.au

SUBSCRIPTION RATES

	Aust	Overseas
1 year	A$77.20	A$88.33
2 year	A$133.10	A$157.30
3 year	A$189.00	A$227.27

Australian rates include GST.
Overseas subscriptions go by airmail
Discounted for new subscriptions
20% Discount for extra copies
Agents rates are discounted.

ADVERTISING RATES

B&W	Non-members	Sus Mem
1/1 Page	$784.00	$699.15
1/2 Page	$509.40	$457.40
1/3 Page	$392.00	$352.90
1/4 Page	$326.70	$294.00

Spot colour: available
Prepared insert: $424.70 Conditions apply
Column rate: $26.40 per cm (1/3 p 5.5cm width)
All rates include GST

Discounted rates for 3 consecutive ads in advance

Special rates available for 4-colour printing

All enquiries to: Mrs Leigh Wallbank
Tel (02) 9528 4362 Fax (02) 9589 0547
wallbank@zipworld.com.au

ACOUSTICS AUSTRALIA ADVERTISER INDEX - VOL 41 No 3

SVAN 979 ANALYSER

Sound & Vibration Analyser

Product description

The Most Advanced & Powerful Single Channel Instrument Ever Made!

Features

- Two DSP for superior performance
- 3 parallel independent profiles
- 1/1 & 1/3 octave RT & FFT anallysis
- 1/6 or 1/12 octave real-time analysis
- Advanced logger with audio events
- Reverberation time measurements
- Time-domain signal recording
- Wireless data transfer
- Signal generator
- Micro SD card
- User programmable Band-Pass Filters

More info: www.acu-vib.com.au

NATA Calibrations

Acoustic and vibration instrument calibration laboratory servicing the whole of Australia and Asia Pacific region. FAST turnaround. NATA registered. All brands

Sales & Service

We supply instruments from several manufacturers, giving you a wider selection to choose from. We also service ALL brands providing you a one-stop solution.

Instrument Hire

We have a wide range of acoustic and vibration instruments for hire. Daily, weekly or monthly rental periods available. Call for advice on the best instrument to suit you

ACU-VIB ELECTRONICS- 14/22 Hudson Ave - Castle Hill 2154 - Phone 02 9680-8133 - eMail info@acu-vib.com.au