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PAPERS

TECHNICAL NOTES

REGULAR ITEMS

Cover design: Helena Brusic

Message from the President

Welcome all to this midyear edition of Acoustics Australia.

The federal councillors who represent your respective divisions recently held their 100th Council teleconference meeting and I would like to take this opportunity to update you of several items that were discussed, noted and agreed on.

Regarding the AAS Website, in the December 2011 issue, I noted that a subcommittee had been formed to investigate its upgrade and asked for input from our members. I am pleased to report that funds have now been budgeted for this upgrade. Responses from divisional committees to the request from the subcommittee for input to the main requirements and recommendations have now been received. The subcommittee is progressing with summarising these inputs and preparing a briefing document of the required changes to assist with selection and engagement of a company to facilitate the changes.

The federal councillors are pleased to note that since November 2011, we have had an increase of 21 new members with a further 18 membership applications currently pending divisional approval. In this past financial year we are seeing a substantial increase in membership.

The WA Division councillors reported on the upcoming Australian Acoustical Society's national conference: Acoustics 2012 Fremantle, to be held at The Esplanade Hotel, Fremantle, Western Australia, from November 21 to 23. Around 160 abstracts have been received to date with approximately 92 papers received, and only two sponsorship booths remain, all indicating a very successful and well attended conference.

As mentioned previously, our website has seen a substantial increase in traffic partly due to the Acoustics Australia journal back issue articles being requested, amongst other items, and this has continued throughout the first six months of this year.

The Federal Councillors have recently agreed that Elsevier is permitted to index and extract data from our conference proceedings through our proceedings sales agent Curran and Associates Inc. Elsevier would only be capturing the citation (Title and Authors), the Abstract or Summary paragraph, and References. Benefits include global visibility, sales, and increased exposure and profile for authors, editors and our publishing organisation.

The AAS is pleased to announce that the Queensland Division has established annual student travel bursaries in honour of Colin Speakman FAAS which will be known as the Colin G Speakman Travel Bursaries. More details for these can be found on the AAS website and on page 150 of this issue.

Regarding international acoustic conferences, please note that the AAS will be the host nation for the InterNoise 2014 Congress to be held in Melbourne at the Melbourne Convention and Exhibition Centre. Norm Broner and Charles Don will be presenting a progress report on our preparation, including budgets, to the I-INCE board at InterNoise 2012 in New York and we wish them well. For more information please refer to the InterNoise 2014 website at http://www.internoise2014.org/

My last comment regarding the upcoming Acoustics 2012 Fremantle conference relates to our AGM. It is a necessary legal requirement and is usually dispensed with in a short time frame given that all documents are issued prior to the meeting and located on our website. If attending the annual conference, please assist your federal representatives and divisional councillors by attending this AGM.

Peter Heinze

Message from the Editor

One of the perks of an academic position is the opportunity to attend conferences. I haven't been able to attend conferences overseas in the last few years due to having a young family. So it was with much delight (and anxiety) when my husband gave me permission to attend the 19th International Congress on Sound and Vibration (ICSV19) in July in Vilnius, Lithuania. It was my first week of freedom in years! (At this point I must acknowledge my wonderful and very capable husband who managed to look after 3 kids aged 3 and under). It was great to catch up with colleagues, both local and international, including a good mate from our student days at UWA. Apart from a presentation (which was thankfully over as the first paper in the first session on the first day), and the opportunity to learn about other's research activities, I hope that the biggest benefit of attending ICSV19 will be the flow of international submissions to Acoustics Australia. Next time I think I'll just wear a shirt stating 'Ambassador for the Acoustics Australia Journal'. I hope you enjoy the issue and see you at Acoustics 2012 Fremantle.

Nicole Kessissoglou

At the ICSV19 banquet: (l-r) Colin Hansen (Adelaide University), Hugh Hunt (Cambridge University), Nicole Kessissoglou (UNSW), Bob Randall (UNSW)

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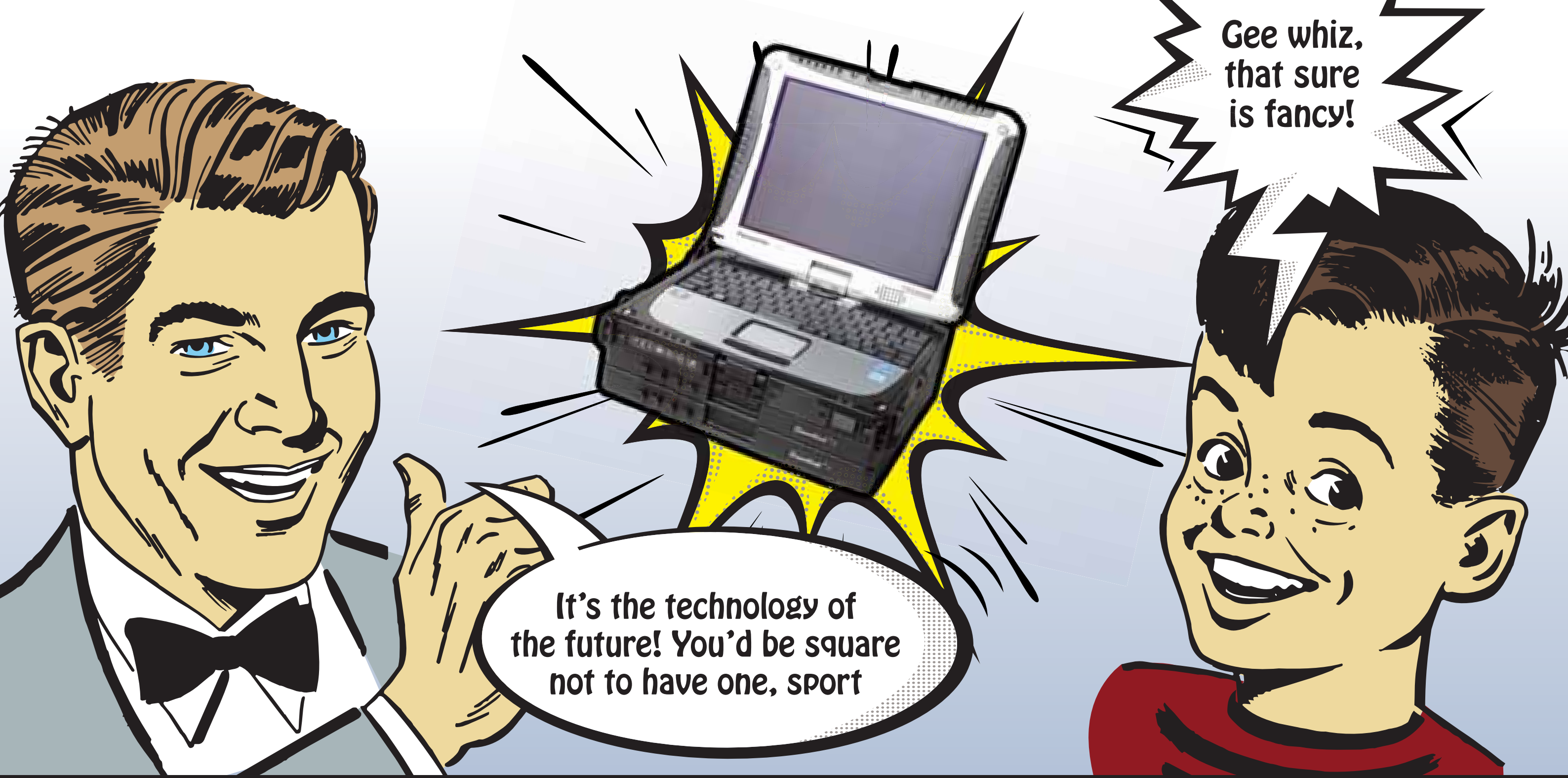

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ACCOUNTING FOR LISTENING LEVEL IN THE PREDICTION OF REVERBERANCE USING EARLY DECAY TIME

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Reverberance, which is an auditory attribute describing the extent to which a room or system is reverberant, is conventionally estimated using early decay time (similar to reverberation time). In a series of recent studies, the authors have shown that reverberance is better estimated using loudness decay parameters, i.e., parameters derived from the decay function of a room impulse response analysed using an objective time-varying loudness model. This approach is based on the notion that the experience of sound decaying in a room is an experience of loudness decay. One reason for the success of this approach is that the loudness decay rate depends on listening level, and this dependency corresponds to subjective experimental data on reverberance. However, loudness-based analysis is neither simple nor computationally efficient, and so this paper proposes a simplified approach to reverberance estimation, using listening level to modify early decay time or reverberation time values.

INTRODUCTION

Reverberation is one of the most important features of room acoustics, so many studies have examined ways of predicting the auditory attribute describing the perceived amount of reverberation, which is referred to as reverberance. Well-established objective parameters are reverberation time (*T*) proposed by Sabine [1] and early decay time (*EDT*) proposed by Jordan [2]. These two parameters are similar in concept, which is to estimate reverberance by determining the time taken for the reverse-integrated sound pressure envelope following an impulse to decay over a certain decibel range (known as the evaluation range) [3]. These two parameters differ by their evaluation range: the evaluation range for *EDT* is from the envelope peak to -10 dB; and that for T_{20} is from -5 dB to -25 dB. The *EDT* evaluation range was inspired by Haas' work [4], which showed the special importance of early reflections in auditory perception. Because *EDT* emphasises the early decay, it is well-suited to account for the reverberance of running signals such as music, in which the vast majority of sound decays are partially masked by subsequent sound events. The efficacy of *EDT* over *T* for estimating reverberance has been demonstrated in subjective listening experiments by Soulodre and Bradley [5] and Barron [6].

A limitation of *EDT* and *T* is that these parameters are derived from sound pressure envelope of room impulse responses (RIRs), whereas the perception of sound decay may be more closely related to the loudness envelope of a signal (such as music) in a reverberant environment. As outlined by Zwicker and Fastl [7], many factors (including, but not limited to, sound pressure) affect loudness, and the calculation of loudness takes into account processes such as temporal integration, spectral masking, auditory filter banks, functions relating auditory excitation to specific loudness and so forth.

Previous studies have shown that reverberance is not only affected by the period of sound decay over the evaluation range, but it is also affected by listening level [8-10]. Simply increasing the listening level of a reverberant stimulus yields increased reverberance. This effect occurs for impulsive stimuli and also for music and speech stimuli. In the case of impulsive stimuli, Lee and Cabrera [9] showed that reverberance is related to both the slope of the loudness decay function (when expressed in logarithmic units) and the duration of the audible decay. For music stimuli, the slope of the loudness decay function dominates, because audiences are unlikely to detect long sound decays [10]. Unlike the logarithmic pressure envelope's slope, the loudness decay function's slope varies with listening level (decaying more rapidly when the listening level is reduced), and changing the slope is a plausible way of manipulating the reverberance of stimuli.

Various recent studies have used an auditory model to estimate aspects of reverberance. The approach of van Schuitman and de Vries [11] was to extract the reverberant sound field from an input signal using an auditory model with a peak detection algorithm, and then to average the reverberant sound field from 250 Hz to 4 kHz over the whole duration of the input signal to predict reverberance. For situations where a dry signal and its reverberant counterpart are both available, Uhle et al. [12] proposed a number of reverberance predictors using a loudness model, from which the unmasked part of the reverberant signal could be predicted. Similarly, Zarouchas and Mourjopoulos [13] estimate the perceived sound alteration due to reverberation using a computational auditory masking model. Matsumoto et al. [14] compared the sound pressure decay envelope of RIRs filtered by simplified auditory filters (dynamic compressive Gammachirp filter) and by the conventional band-pass filters, demonstrating that the auditory filters account better for reverberance. These various approaches show that an auditory model can provide more accurate representations of reverberance than the conventional approach.

Lee and Cabrera [9] proposed loudness-based reverberance predictors, T_N and EDT_N (the subscript '*N*' stands for loudness), using computational objective loudness models such as Glasberg and Moore's Time-varying Loudness Model [15] and Chalupper and Fastl's Dynamic Loudness Model [16] (for this purpose of deriving reverberance predictors, these two models perform equally well). After calculating the loudness decay function of a RIR at the relevant listening level, T_N and EDT_N may be calculated in close analogy with their respective counterparts (*T* and *EDT*). The loudness decay function of an RIR is approximately exponential, and so a linear regression can be conducted after taking the logarithm of the function. According to Stevens [17], loudness approximates sound pressure raised to a power of 0.6 for tones of moderate frequency and listening level, which is consistent with the well-known rule-of-thumb that doubling or halving loudness corresponds to ±10 dB. Hence, an evaluation range from peak to half of the peak loudness is used for EDT_N in analogy to *EDT*, and an evaluation range from 0.708 to 0.178 of the peak loudness is analogous to the evaluation range of T_{20}. Details of T_N and EDT_N calculations are described by Lee et al. [9, 10].

While these loudness-based predictors of reverberance have been shown to be substantially more effective reverberance predictors than *EDT*, they are neither straightforward to apply nor easily interpreted. The present paper examines whether a simpler and more accessible approach to estimating reverberance could be made, by using a combination of familiar parameters. Results from the experiments previously conducted by the authors are re-analysed, and a simple model is proposed.

DESCRIPTION OF EXPERIMENTS

Five listening experiments were conducted. These experiments, which have been described in detail previously, followed a similar methodology, and their results have been analysed previously in terms of loudness decay parameters [9, 10, 18, 19]. The participants' task in all experiments was to adjust the reverberance of each stimulus to match that of a reference stimulus. This adjustment was achieved within the computer-based experiment software by altering the decay rate of the room impulse response (RIR) associated with the stimulus, by multiplying it by an exponential function. This was implemented in the experiment software as per Equation 1, where *d* is used to increment or decrement the reverberation time. In equation (1), $p(t)$ is the sound pressure of the RIR, *t* is time in seconds, *d* is the decay rate adjustment value and $p'(t)$ is the sound pressure of decay-rate adjusted RIR. Further details relating to such manipulation of RIRs are given by Cabrera et al. [20]. Hence, the participant would press the 'More' or 'Less' button on the graphical user interface (GUI) to incrementally increase or reduce the reverberation time of each stimulus, so as to perceptually match it to a reference stimulus. Figure 1 shows a screenshot of the Matlab-based GUI used in all the experiments, except for Experiment 1, which was realized using different software (Max/MSP), but with a similar GUI. Note that the maximum stimulus number in Figure 1 was changed for different experiments. The initial value of *d* for each stimulus was randomised by the software. Note that the just-noticeable difference (JND) of reverberance is conventionally given as a 5% change of EDT_{mid} [3], so a unit change of *d* yields a change of 4%. (The subscript "mid" indicates an average of the 500 Hz and 1 kHz octave band values.) A side effect of the decay rate adjustments is a small change in the listening level of comparison stimuli, which was compensated for in the computer-based experiment software before presenting to the subjects. Stimuli were presented via headphones (Sennheiser HD600) and the experiments were conducted in quiet environments. Table 1 provides information about each experiment, including the stimulus signal type, the type of reverberance examined, presentation conditions, and the number of participants (following the removal of a small number of atypical and/or unreliable participants, as described in [9,10]).

$$p'(t) = p(t)\exp\left(\frac{\left(-3 + (3 \times 1.04^{d})\right) t}{1.04^{d}}\right) \qquad (1)$$

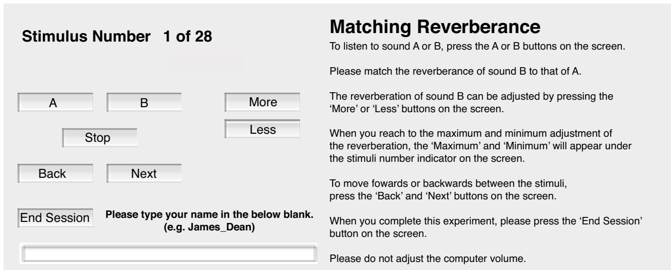

Figure 1. Matlab-based graphical user interface (GUI) used in Experiments II, III, IV and V. The reference stimuli were loaded on the 'A' button and the comparison stimuli were loaded on the 'B' button. The subjects adjusted the reverberance of comparison stimuli by pressing the 'More' or 'Less' buttons on the GUI.

Experiment I (previously reported by Lee and Cabrera [9]) tested impulsive reverberance, by presenting RIRs directly (i.e., without any convolution with a dry source such as music). The experiment used eight RIRs measured by Farina and Ayalon [21] in three auditoria within the *Parco della Musica* in Rome. The small auditorium has 700 seats, medium one has 1200 seats and the large one has 2800 seats (in Table 2 these are labeled 'S', 'M' and 'L' respectively, and the numbers after the letters indicate different receiver positions with a fixed on-stage source). The eight RIRs were recorded with identical equipment and gain, so that the RIRs retain relative levels. In Table 2, the L_{AFmax} is the maximum A-weighted sound pressure level. As the RIRs were single channel, the stimulus presentation over headphones was diotic. In order to investigate the effect of listening level on impulsive reverberance, additional gains of -5, 0 and 5 dB were applied to the RIRs. Therefore, twenty-four comparison stimuli (eight RIRs multiplied by the three additional gain settings) were generated and paired with a single reference stimulus of RIR M1. RIR M1 was chosen as the reference stimulus because it came the mid-sized auditorium, and chosen over the other two stimuli in that auditorium because RIR M2 has the lowest values of conventional parameters and RIR M3 was measured at a source-receiver distance not available in the small auditorium. The listening levels shown in Table 1 take the additional gains of ±5 dB into account.

Experiments II and III (reported by Lee et al. [10]) used the same set of RIRs as Experiment I (including the ±5 dB additional gain), but the RIRs were convolved with anechoic music recordings. For Experiment II, the music was orchestral (bars 1-18 of the *Overture* to *The Marriage of Figaro* by Mozart – which is the same excerpt as that used by Soulodre and Bradley [5]). Because this anechoic recording is stereophonic [22], the presented stimuli (after convolution with a single-channel RIR) are best described as stereophonic, unlike the diotic stimuli used in other experiments. Table 2 shows the L_{Aeq} (the power-average of A-weighted sound pressure level) of the experiment stimuli (at 0 dB gain). For Experiment III, the music was a recording of an opera singer singing the final sixteen bars (11.5 s) of *Torna a Surriento* by Ernesto Di Curtis (which is an Italian song in *bel canto* style). Apart from Experiment III, all of the experiments were conducted in the anechoic room at the University of Sydney; Experiment III was conducted in an audiometric booth in the Advanced Acoustic Information Systems Laboratory at the Research Institute of Electrical Communication at Tohoku University in Japan.

Table 1. Summary of the five experiments

Exp.No.	Stimulus Signal	Type of Reverberance	Headphone Presentation	Listening Level (dBA)	Reverberation Time	No. of Participants
I	Real RIRs	Impulsive	Diotic	58.7 to 80.4	2.01 s to 2.66 s	18
II	Orchestral Music	Overall	Stereophonic	60.1 to 81.0	2.01 s to 2.66 s	16
III	Tenor Singing	Overall	Diotic	60.2 to 82.5	2.01 s to 2.66 s	11
IV	Synthetic RIRs	Impulsive	Diotic	50.0 to 80.0	1.00 s to 3.00 s	10
V	Orchestral Music	Running	Diotic	60.0 to 80.0	1.00 s to 3.00 s	10

Table 2. Source-receiver distance, mid-frequency early decay time (EDT_{mid}), mid-frequency reverberation time (T_{mid}) and maximum sound pressure level (L_{AFmax}) of the RIRs (Experiment I); and equivalent sound pressure level (L_{Aeq}) of dry signals convolved with corresponding RIRs (Experiments II-III)

	S1	**S2**	**M1**	**M2**	**M3**	**L1**	**L2**	**L3**
Distance	12	24	10	19	31	20.5	30	48
EDT_{mid} (s)	1.89	1.98	1.83	1.77	2.00	2.44	2.25	2.38
T_{mid} (s)	2.06	2.07	2.01	2.03	2.17	2.66	2.60	2.53
Exp. I L_{AFmax} (dB)	75.4	74.9	75.0	72.7	70.9	69.9	69.5	63.7
Exp. II L_{Aeq} (dB)	76.0	75.6	75.5	73.7	72.4	71.3	70.7	65.1
Exp. II L_{Aeq} (dB)	77.5	76.1	76.8	74.6	73.5	72.1	71.1	65.2

Experiment IV [18] (like Experiment I) tested impulsive reverberance by presenting RIRs directly as stimuli. However, the RIRs of Experiment IV were synthesized (rather than measured from real rooms). The synthetic RIRs were generated using octave-bands of white noise (centered on 31.5 Hz – 16 kHz), which were multiplied by exponential decay functions, following a simple impulse representing the direct sound. Details of the procedure of generating the synthetic RIRs are provided by Lee et al. [18, 19]. As seen in Table 1, Experiment IV tested impulsive reverberance over a greater range of listening levels and reverberation times than Experiment I. The two main reasons for performing Experiment IV was to determine if the loudness-based predictors (i.e., T_N and EDT_N) perform well over a wider range of listening levels and reverberation times (when reference stimuli also have various listening levels and reverberation times); and to construct equal-reverberance contours for impulsive signals. Figure 2 shows the structure of Experiment IV. In Part A (hereafter, Experiment IV-A), the effect of listening level on impulsive reverberance was tested using reference stimuli with a fixed T value of 2 s and various listening levels (L_{AFmax}) from 50 dBA to 80 dBA. Part B (hereafter, Experiment IV-B) tested the effect of T on impulsive reverberance with reference stimuli having a constant listening level of 60 dBA and various T values ranging from 1 s to 3 s. Four comparison stimuli were paired with each reference stimulus and the participants adjusted the reverberance of comparison stimuli to match the reverberance of the corresponding reference stimulus. Hence, Experiments IV-A and IV-B tested sixteen pairs each (four comparison stimuli multiplied by four reference stimuli). For presentation, the two parts of the experiment were mixed together in randomized order. Two sets of equal-reverberance contours were derived from this experiment. Note that there are four pairs common to IV-A and IV-B, which include the reference stimulus having a listening level of 60 dBA and T of 2 s. In order to shorten the experiment time, they were tested only once, but the results from common pairs were included in the analyses of both parts of the experiment.

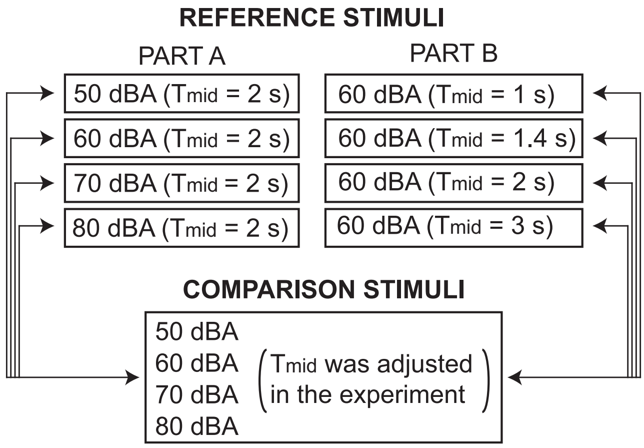

Figure 2. Structure of Experiment IV for Parts A and B

Experiment V [19] used the same synthetic RIRs as Experiment IV, but the RIRs were convolved with an anechoic musical excerpt of *Water Music* by Handel from *Denon Professional Test CDs No.2* [22]. In Experiment V, running reverberance was tested, which is the reverberance experienced while a stimulus is playing [23]. Hence, a very rapid decay was applied to the last note of the convolved musical stimulus in order to eliminate the stopped (or terminal) reverberance following the last note. This experiment was conducted with the same form as Experiment IV, except that the listening level of 50 dBA (L_{Aeq}) was excluded. Hence, Experiment V-A tested nine pairs and Experiment V-B tested twelve pairs. Similarly to Experiment IV, there were three pairs common to V-A and V-B (i.e., when the reference stimulus has a listening level of 60 dBA and T of 2 s) and they were also tested only once to shorten the experiment time.

All experiments yielded significant effects, indicating that listening level and reverberation time both significantly affect reverberance. In Experiments I-III, the RIR was an experimental variable (rather than reverberation time directly), and the effect of RIR was significant. Table 3 shows the analysis of variance (ANOVA) results for Experiments I-III combined. The effect size (which can be expressed as η^2, or the sum of squares for the factor, divided by the total sum of squares) was approximately 1.4 times greater for gain adjustment than for the RIR. While the three experiments yielded significantly different results, the effect of experiment number is substantially smaller than the effects of RIR or gain. There are no significant interactions, as none of the two-factor interaction analyses has a prob>F value less than 0.05 (this indicates that the subjective responses to one independent variable are not affected by another independent variable).

Table 3. Result of the analysis of variance (ANOVA) of Experiments I-III combined, analysed in terms of experiment number (Exp. No.), room impulse response (RIR) and additional gain of ±5 dB (Gain). Values are the sum of squares (Sum Sq.), degrees of freedom (d. f.), mean square (Mean Sq.), the F statistic, significance (prob>F) and effect size (η^2). For a confidence level of 95%, prob>F must be 0.05 or less, and the respective sizes of significant effects are shown as η^2

Source	Sum Sq.	d. f.	Mean Sq.	*F*	prob>F	η^2
Exp. No.	546	2	272.98	19.85	0	0.028
RIR	1610.2	7	230.02	16.73	0	0.083
Gain	2255	2	1127.48	81.98	0	0.117
Exp. No * RIR	291.5	14	20.82	1.51	0.099	
Exp. No * Gain	43.6	4	10.9	0.79	0.5302	
RIR * Gain	130.1	14	9.3	0.68	0.7996	
Error	14248	1036	13.75			
Total	19259.7	1079				

Due to their more complex structure, the statistical analysis of Experiments IV and V is more involved, and details are given in [18, 19]. In Experiment IV-A, the effects of reference stimulus listening level and comparison stimulus listening level were both significant (p<0.0001), and similarly, in Experiment IV-B, the effects of reference stimulus reverberation time and comparison stimulus listening level were also significant (p<0.0001). In Experiment V (which examined running, rather than impulsive reverberance), the effect of reference stimulus listening level was only significant at 90% confidence in V-A (p=0.0904) but the effect of comparison listening level was significant (p=0.0374); and in V-B the effect of reference stimulus reverberation time was significant (p<0.0001) along with the effect of comparison stimulus listening level (p=0.0276).

In all experiments, it was shown that loudness decay analysis provides a better model for reverberance than conventional parameters such as *EDT* [9, 10, 18, 19]. In the following section the experiment results are re-modeled using a simpler alternative approach.

RE-ANALYSIS

To derive acoustical parameters representing the experiment results, the subjective responses (represented by the decay adjustment value of d) were averaged, and adjusted RIRs were generated using the averaged d values. Since the experiment task was to match reverberance, this procedure yielded sets of RIRs with approximately equal reverberance. Then, the acoustical parameters were derived from these adjusted RIRs. This process was performed for each experiment.

A simple function that expresses reverberance in terms of listening level and EDT (or T) was sought – and possible functions were tested and refined using the parameters of the adjusted stimuli. Listening level was defined as L_{AFmax} for impulsive stimuli and L_{Aeq} for running (music) stimuli, which was previously shown to be a useful correspondence for loudness-based reverberance modeling [10]. These listening levels were those presented to the participants in the experiments (measured using a Brüel & Kjær type 4100 *Head and Torso Simulator* wearing the employed headphones). Goodness of fit was assessed by the extent to which a function yielded minimal deviation from equal reverberance for each of the sets of equally reverberant stimuli generated from the results of the five experiments.

Equations (2) and (3) are the most successful succinct functions. In these equations, L represents the listening level (L_{AFmax} for impulsive stimuli, L_{Aeq} for music), and the listening level-modified T and EDT are shown with L as a subscript. The exponent acts to compress (or expand) the relationship between the decay time (T or EDT) and the reverberance predictor (T_L or EDT_L), and the extent of this compression or expansion is determined by L. For T, the best fit comes with a unit exponent (i.e., no compression or expansion) when the listening level is 70 dBA; and for EDT, the best fit has a unit exponent when L is 80 dBA. These listening levels (70 and 80 dBA) are, of course, round numbers, but there was little to be gained from the added complexity of using more precisely determined values, given the limited experimental data. An important concept underlying the development of these functions is that the effect of listening level on reverberance is greatest when the reverberation time is long, and Experiments IV and V yielded scarcely any effect of level when the reverberation time was 1 s. The functions only apply to decay times greater than or equal to 1 s (listening level has no effect on the predictor when the decay time is 1 s).

$$T_L = T^{L/70} \quad (T \geq 1\text{ s}) \tag{2}$$

$$EDT_L = EDT^{L/80} \quad (EDT \geq 1\text{ s}) \tag{3}$$

Figure 3 compares performance of the proposed parameters ($T_{L,\mathrm{oct}}$ and $EDT_{L,\mathrm{oct}}$) with the conventional parameters (T_{oct} and EDT_{oct}) and loudness-based parameters (T_{N} and EDT_{N}). The subscript 'oct' indicates parameter values averaged over 125 Hz to 4 kHz octave bands. Note that octave-band values of the loudness-based parameters are not available, because the loudness model incorporates integration across the auditory filter-bank. The y-axis of the figures shows the coefficient of variation, which is the standard deviation divided by mean. This statistical parameter eliminates a mean-related bias that is likely to exist in the standard deviation (because larger means may be accompanied by larger standard deviations). As the reverberance of all the comparison stimuli was adjusted to that of a reference stimulus, an ideal reverberance predictor should yield a coefficient of variation of zero. As described in the previous section, more than one reference stimulus was used within Experiments IV and V. Hence, the coefficients of variation were calculated over subjective responses for each reference stimulus and theses values were averaged to yield a single-value representation in Figure 3. As seen in the figure, $T_{L,\mathrm{oct}}$ and $EDT_{L,\mathrm{oct}}$ perform similarly well to their respective loudness-based parameter counterparts (T_{N} and EDT_{N}), and in most cases $EDT_{L,\mathrm{oct}}$ performs somewhat better than EDT_{N}. The conventional parameters exhibit the worst performance in every case.

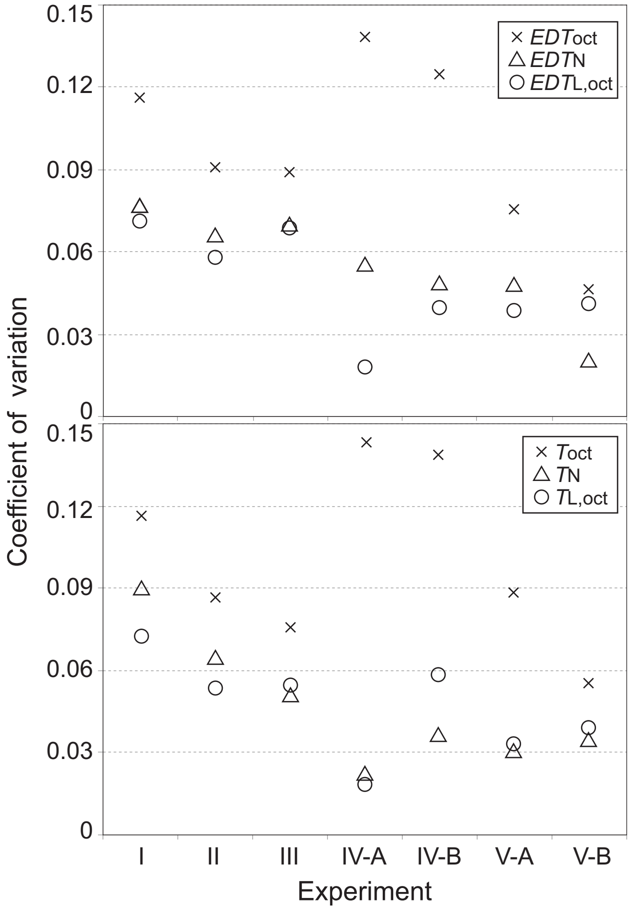

Figure 3. Comparisons of $EDT_{L,\mathrm{oct}}$ with the conventional EDT_{oct} and EDT_{N} (upper figure) and of the modified $T_{L,\mathrm{oct}}$ with the conventional T_{oct} and T_{N} (lower figure). The y-axis is the coefficient of variation, which is the standard deviation divided by mean.

As Experiments IV and V tested reference stimuli with various listening levels and reverberation times, the subjective responses obtained from these experiments enable the derivation of equal-reverberance contours. Figure 4 shows these equal-reverberance contours expressed in terms of the conventional parameters (EDT_{oct} and T_{oct}) and the proposed parameters ($EDT_{L,\mathrm{oct}}$ and $T_{L,\mathrm{oct}}$) for Experiment IV, as a function of the listening level of comparison stimuli. An ideal reverberance predictor should yield flat horizontal contours. As seen in the figures, the contours derived from the proposed parameters are much closer to this ideal than those from the conventional parameters. For the conventional parameters,

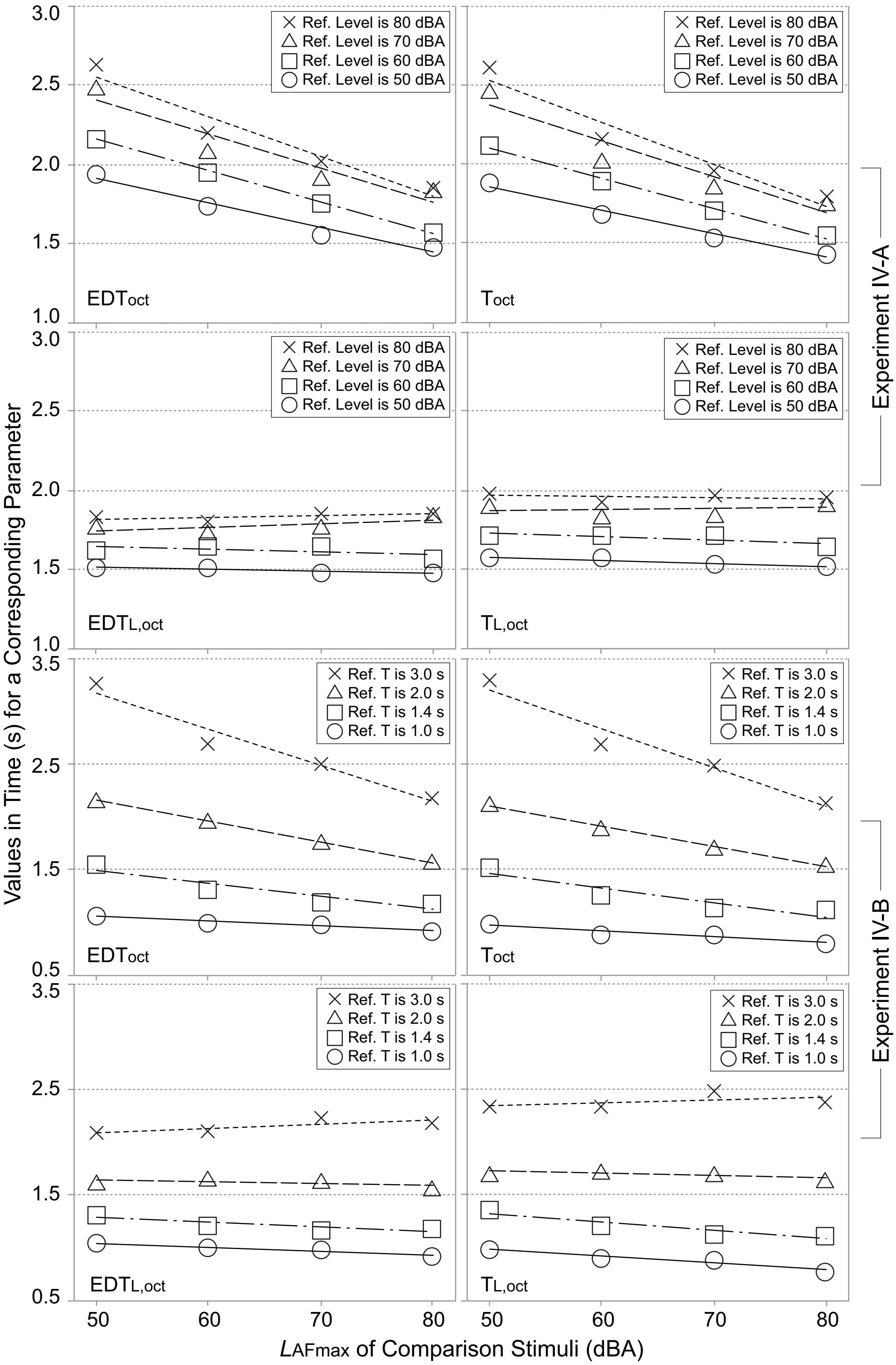

Figure 4. Equal-reverberance contours as a function of the listening level of comparison stimuli for $EDT_{L,\mathrm{oct}}$ and EDT_{oct} (the left four charts) and for $T_{L,\mathrm{oct}}$ and T_{oct} (the right four charts). The four upper charts are for Experiment IV-A, and the four lower charts are for Experiment IV-B.

as the listening level of comparison stimuli increases, the participants reduced the values of conventional parameters to match the reverberance. This implies that the participants experienced greater reverberance when the listening level increases, and this phenomenon becomes stronger as the reverberation time of reference stimuli increases (as shown in the third part of the figure). Table 4 shows the averaged slopes of the equal-reverberance contours derived from Experiments V-A and V-B for the conventional parameters and the proposed parameters. As the ideal equal-reverberance contours are flat horizontal lines, a perfect parameter would yield a value of zero in the table. The table indicates that the proposed parameters outperform the conventional parameters, and this is more substantial for Experiment V-A than Experiment V-B. In Experiment V-B, the proposed parameters exaggerate the effect of listening level on reverberance when the reference stimulus has a T value of 3 s (i.e., the values of the proposed parameters derived from the subjective responses increase with the listening level on this equal reverberance contour). This exaggeration is similar in size to the amount of reduction in the conventional values for the listening level increase. Note that the conventional parameters and the proposed parameters perform similarly when the reference stimuli had T values of 1 s and 1.4 s. When the reference stimulus has a T value of 2 s, the proposed parameters outperform the conventional parameters (for such a reference stimulus, the slope of equal-reverberance contours for EDT is -0.0145, while the slope of the contours for EDT_L is 0.0050). Hence, the proposed parameters' slightly better performance in Experiment V-B is mostly due to the subjective responses for the reference stimulus having a T value of 2 s.

Table 4. The averaged slopes over the equal-reverberance contours derived for EDT_{oct}, $EDT_{L,oct}$, T_{oct} and $T_{L,oct}$ from Experiment V. The values are time (in seconds) per comparison stimulus gain (in dB).

	EDT_{oct}	$EDT_{L,oct}$	T_{oct}	$T_{L,oct}$
Experiment V-A	-0.0148	0.0060	-0.0167	0.0032
Experiment V-B	-0.0090	0.0087	-0.0099	0.0075

DISCUSSION

The approach to modeling reverberance taken in this paper appears to be similarly effective to loudness decay modeling, and yet it is much simpler to apply. Like the loudness-based parameters, it significantly outperforms conventional parameters. The loudness-based parameters are more fundamental, in the sense that they model something of the low level auditory processing that leads to reverberance perception. The simpler approach taken here does not model auditory processing, but merely augments conventional reverberance predictors by reflecting the phenomenon that greater listening level yields greater reverberance.

The proposed models are limited to the range of listening levels and reverberation times shown in Table 1 (50 dBA $\leq L \leq$ 82.5 dBA; 1 s $\leq T \leq$ 3 s), in large rooms with the source well-beyond the near-field, and are based only on music and impulsive stimulus data (speech was not tested, and the tested music was two orchestral excerpts and solo singing). Music stimuli were not tested below 60 dBA, although there may be little practical reason to examine the reverberance of music quieter than this. Clearly, the models do not apply for reverberation times of less than 1 s, because this would invert the positive relationship between listening level and reverberance. Instead, in the absence of further experimental data it would be sensible to presume that listening level has a negligible effect on reverberance for reverberation times of less than 1 s.

The results do not provide a clear indication as to which predictor (EDT_L or T_L) is superior. In the absence of such an indication, it makes sense to choose EDT_L, because EDT is more effective than T for running stimuli (and EDT_L is simple modification of EDT). Figure 5 shows the relationship between EDT and EDT_L evaluated from Equation 3.

EDT_L combines the effects of signal and system to estimate reverberance, whereas parameters used in auditorium design tend to focus on the system alone (because the acoustician has no control over the signals subsequently emitted in an auditorium). Instead of using $L_{AF,max}$ or L_{Aeq} to represent listening level, it may be possible to generalise the approach taken in the present paper to use strength factor, G, in a modified function. Strength factor is a system response characteristic, defined as the difference between the sound pressure level measured from an omnidirectional source in the auditorium (typically with the source on stage) to a receiving position (typically in the audience area), and the sound pressure level measured from the same source (producing the same acoustic power) at a distance of 10 m in an anechoic environment [3]. For this modification to be made, some assumptions would need to be made regarding the power of a typical sound source.

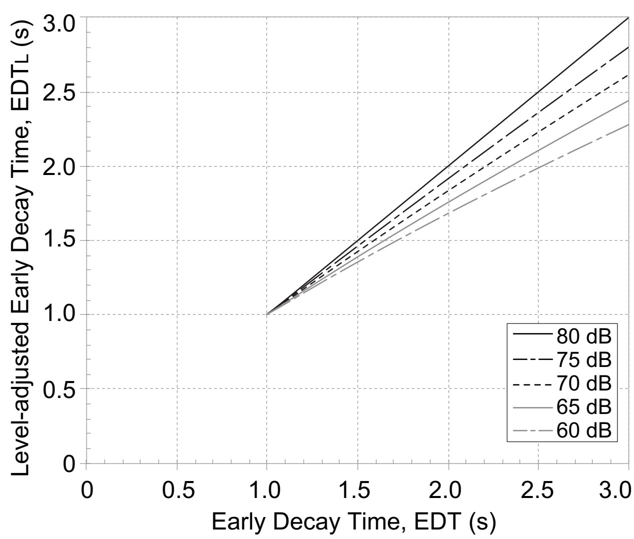

Figure 5. Relationship between the level-adjusted early decay time (EDT_L) and the conventional early decay time (EDT) for various listening levels from 60 dB to 80 dB

CONCLUSIONS

The present study proposes a simple way of more accurately estimating reverberance than offered by the conventional parameters (e.g., *T* and *EDT*) alone, by taking listening level into account. The proposed parameters work well over a range of listening levels and reverberation times commonly found in auditorium listening conditions. Previous studies show that the loudness-based parameters (which involve much more intensive calculation) obviously outperform the conventional parameters, but the present study found that the proposed listening-level modified parameters perform similarly to the loudness-based parameters. Hence parameters of the type proposed in the present paper may be of more practical value for estimating reverberance in many contexts.

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ESTIMATION OF NOISE MODEL AND DENOISING OF WIND DRIVEN AMBIENT NOISE IN SHALLOW WATER USING THE LMS ALGORITHM

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Signal transmission in ocean using water as a channel is a challenging process due to the effect of attenuation, spreading, reverberation, absorption etc., apart from the contribution of acoustic signals due to ambient noises. Ambient noises in sea are of two types namely manmade (shipping, aircraft over the sea, motor on boat, etc) and natural (rain, wind, marine fishes, seismic, etc). The ambient noises contribute more effect on reducing the quality of acoustic signal. In this paper we concentrate on denoising the effect due to wind on underwater acoustic signal using the LMS algorithm. The wind speed of the collected data ranges from 2.11 m/s to 6.57 m/s. The analysis is carried out for acoustic frequencies ranging from 100 Hz to 8 kHz. It is found that a linear relationship between noise spectrum and wind speed exists over the entire frequency range. The results of the empirical data are compared with the results obtained with the aid of the noise model developed. An adaptive model exploiting the Least Mean Square (LMS) algorithm to denoise wind driven ambient noise in shallow water has been proposed. The observation shows that the Signal to Noise Ratio (SNR) is enhanced two fold and the Mean Square Error (MSE) decreases exponentially with the aid of the LMS adaptive algorithm.

INTRODUCTION

Signal transmission underwater is a challenging task. Generally, low frequency acoustic signals are used for transmission underwater as electromagnetic signals are highly attenuated. Any modulated signal transmitted in water, undergoes various losses due to attenuation, reverberation, spreading and internal waves etc apart from ambient noise due to natural and manmade sources.

The residual noise background in the absence of individual identifiable sources may be considered as the natural noise environment for hydrophone sensors. It comprises a number of components that contribute to the Noise Level (NL) in varying degrees depending on the location of measurements [1]. The sources contributing noise include geological disturbances, non-linear wave interaction, turbulent wind stress on the sea surface, shipping, distant storms, seismic prospecting, marine animals, breaking waves, spray, rain, hail impacts and turbulence [2]. The ambient noise level spectrum is summarized in [3]. Furthermore Knudsen spectra [4] show the strong dependence of spectral power level with wind speed and sea states.

Noise measurements made in the Northern Hemisphere show self-similar wind dependent noise spectra between 100 Hz and 10 kHz [3,4], but no dependency on wind speed below 100 Hz, with noise at these lower frequencies being attributable to distant shipping. Measurements made at 40 different locations in the Southern Hemisphere showed that in regions of low shipping density the effect of wind speed is dominant in the frequency band of 22 Hz to 5 kHz [5].

The ambient noise masks the signals from underwater acoustic instruments, so the detection and cancellation of background noise is essential to enhance the SNR of acoustic based underwater instruments. This can be done by a proper adaptive filter implementation [6,7]. In this paper, an LMS based adaptive algorithm to denoise the received signal is implemented.

DATA COLLECTION AND NOISE MODEL

Data collection

The data for analysis were collected using two calibrated omni-directional reson TC 4032 hydrophones mounted in a vertical array at 5 m and 15 m depths where the depth of the sea is 25.7 m. The hydrophones have a receiving sensitivity of -170 dB over a frequency range between 100 Hz and 100 kHz. The data were acquired at a rate of 50 kHz and 500 kHz, filtered and digitised with a portable data acquisition system with 12-bit resolution. The wind speed was simultaneously measured. The measurement consists of 7 sets of data. The wind speeds of collected data range from 2.11 m/s to 6.57 m/s.

Noise model

Theoretically, the relationship between the noise levels is assumed to be proportional to the logarithm of the wind speed and this can be expressed as

$$NL = B + 20n \log(U) \tag{1}$$

where NL and U represent noise level and wind speed respectively.

The constants B and n were determined by comparing the experimental data to the model at different frequencies, where n is obtained from a 1/20 slope of regression line and the ordinate intercept of the line gives B for each empirical fit. The spectral analysis was carried out in MATLAB using the Welch method of averaging periodogram. The frequency of interest for this study ranges from 500 Hz to 8 kHz, it is inferred that wind speed and the noise level is best correlated over the frequency analyzed.

DENOISING USING LMS ALGORITHM

From the PSD of the data collected, it is noted that the noise due to wind is dominating over a range of 500 Hz to 5 kHz and extends up to 6 kHz. The effect is high at lower frequencies. Above 6 kHz, the effect due to wind is low and remains constant. An adaptive filter with the LMS algorithm is developed to denoise the effect of wind on the signal. An adaptive filter is a self-designing system that relies for its operation on a recursive algorithm which makes it possible for the filter to perform satisfactorily in an environment where knowledge of the relevant statistics is not required. The algorithm starts from some predetermined set of initial conditions, representing whatever is known about the environment. In a non-stationary environment, the algorithm offers a tracking capability in which it can track time variations in the statistics of the input data provided that the variations are sufficiently slow.

Theoretical model

The most commonly used structure in implementing adaptive filters is the transversal structure shown in Fig. 1. The transversal adaptive filter can be split into two main parts, the filter part and the update part. The function of the filter part is to calculate the filter output $y(n)$, whereas the function of the update part is to adjust the set of N filter co-efficient (w_i), $i = 0, 1, \ldots, N-1$ (tap weights) so that the output $y(n)$ reaches as close as possible to a desired signal $d(n)$.

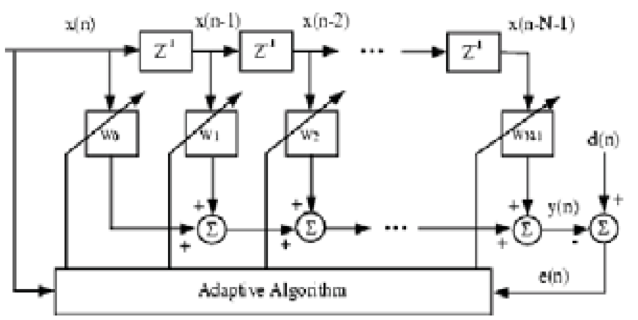


Figure 1. Structure of adaptive filter

In this paper, the filter weights are updated using the LMS algorithm which can remove the noise due to wind that gets added in the channel with the transmitted signal. Initially a test signal (training signal similar to desired signal/reference signal) is fed as input signal to the weight update of the adaptive filter for the update operation. The input to the adaptive filter will be two signals, one is noisy signal and the other is output of weight update in order to tune the filter. When the adaptive weights are tuned according to the signal fed initially, the inputs to the adaptive filter will be from the adaptive portion (update) tuned earlier and the noisy signal. The adaptive filter estimates the error due to the noisy signal. This estimated signal is compared with the reference signal and the difference between these two gives the error signal. The error signal is the exact mismatch between the reference signal and the adaptive filter estimated output. This error signal is passed to the weight update where the weight updates according to the error signal and this updated signal is now compared with the next sample of input noisy signal in the filter. This process is repeated till the error signal tends to zero which means that the weight update is perfectly tuned to the desired signal and there by the estimated output of the adaptive filter is the transmitted signal. The weight update is carried out using the LMS algorithm and the effects of non-stationary of the noise signals are eliminated.

Mathematical model

The input signal $x(n)$ to the adaptive filter at the receiver side is the sum of the desired signal $d(n)$ and interfering noise $v(n)$ in the channel

$$x(n) = d(n) + v(n) \tag{2}$$

where $x(n)$ is the input signal to the adaptive filter, $d(n)$ is the desired signal and $v(n)$ is the interfering noise. The adaptive variable filter has a Finite Impulse Response (FIR) structure. For FIR structures the impulse response is equal to the filter coefficients. The coefficients for a filter of order p is defined as

$$w_n = [w_n(0), w_n(1), \ldots, w_n(p)]^T \tag{3}$$

The error signal $e(n)$ or cost function is the difference between the desired signal $d(n)$ and the estimated signal $y(n)$.

$$e(n) = d(n) - y(n) \tag{4}$$

The variable filter estimates the desired signal by convolving the input signal with the impulse response. In vector notation this is expressed as

$$y(n) = w_n * x(n) \tag{5}$$

where

$$x(n) = [x(n), x(n-1), \ldots, x(n-p)]^T \tag{6}$$

is the input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$$w_{n+1} = w_n + \Delta w_n \tag{7}$$

and the adaptive algorithm generates this correction factor based on the input and error signals.

The LMS algorithm is a linear adaptive filtering algorithm, which, in general consists of two basic processes: a filtering process and an adaptive process. The LMS algorithm is built on the transversal filter concept. This component is responsible for performing the filtering process using a mechanism for performing the adaptive control process on the tap weights of the transversal filter. The LMS algorithm can be written in the form of three basic relations as

1. Adaptive filter output: $y(n) = \hat{w}^H(n)x(n)$
2. Estimation error or error signal is $e(n) = d(n) - y(n)$
3. Tap-weight adaptation is given by $\hat{w}(n+1) = \hat{w}(n) + \mu x(n)e(n)$

where $e(n)$ is the error signal, $x(n)$ is the input signal vector, μ is the step-size parameter, $\hat{w}(n)$ is the tap-weight vector, $d(n)$ is the desired response

RESULTS AND DISCUSSION

Estimation of power spectrum of collected data

Eight sets of data with various wind speeds of 2.11, 3.32, 4.52, 5.92, 6.03, 6.06, 6.16, 6.57m/s are used for analysis. The power spectral densities using the Bartlett and Welch methods for all wind speeds over a range of 25 kHz are shown in Figs. 2(a) and 2(b), respectively. The parameters considered for estimation are shown in Table 1.

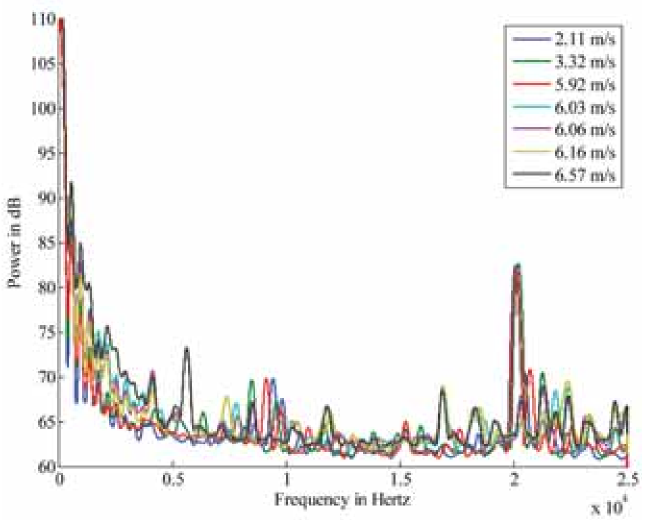


Figure 2(a). Power spectral density for various wind speeds using the Bartlett Method

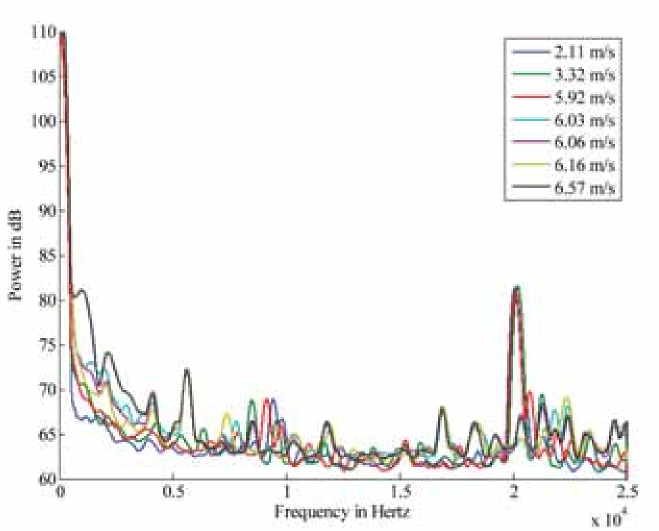


Figure 2(b). Power spectral density for various wind speeds using the Welch Method

Table 1. Parameters considered for estimation of power spectral density

Parameters	Value
Sampling frequency	50 kHz
Window type	Hanning
N-point FFT	65536
FFT window size	1024
Overlapping	50%
Hydrophone sensitivity	-170dB

For a wind speed of 2.11 m/s, the effect of wind is high at lower frequencies and it is found that the Noise Sound Level (NSL) is maximum around 76 dB for 500 Hz and decreases to 65 dB for 5 kHz. Above 5 kHz, the NSL is found to be constant. The estimation is carried out for all wind speeds mentioned above. It is inferred that as the wind speed increases the noise level also increases and the spectral level decreases with increase in frequencies. It can be noted that at 500 Hz the NSL is 76 dB for a wind speed of 2.11 m/s and it is 85 dB for a higher wind speed of 6.57 m/s. It is also evident at various frequencies that the wind speed increases the noise level. It is observed that the NSL is 73, 69, 67, 66, 65.5 and 65 dB at 500 Hz, 1 kHz, 2 kHz, 3 kHz, 4 kHz and 5 kHz respectively for a wind speed of 2.11 m/s. Similarly, the NSL is 82, 76, 73, 72.7, 68 and 66 dB for a wind speed of 6.57 m/s. The noise level of other wind speeds mentioned lies between these two levels. The analysis has been carried out to study the wind dependent ambient noise spectrum level in the frequency range between 500 Hz to 8 kHz.

Noise model analysis

The noise model has been developed and the results are presented in Fig. 3. It is noticed that there is a steep increase in the slope of the noise level as wind speed increases. It is found that above 5 kHz the NSL does not increase and remains constant, which leads to the conclusion that the effect of wind is dominating at lower frequencies.

Table 2 shows the values of *B* and *n* obtained from regression plots. The value of slope is maximun at 500 Hz and decreases as frequency increases. The values of *n* and *B* obtained from the emperical fitting are used for validation with measured noise level.

Figure 4 shows the comparison of predicted noise levels in dB using the noise model and measured noise levels for wind speed of 2.11 m/s, 3.32 m/s, 5.92 m/s and 6.57 m/s. It is observed that the predicted noise levels are as good as with the measured noise levels. As the wind speed increases the predicted noise model deviates slightly from the measured noise level.

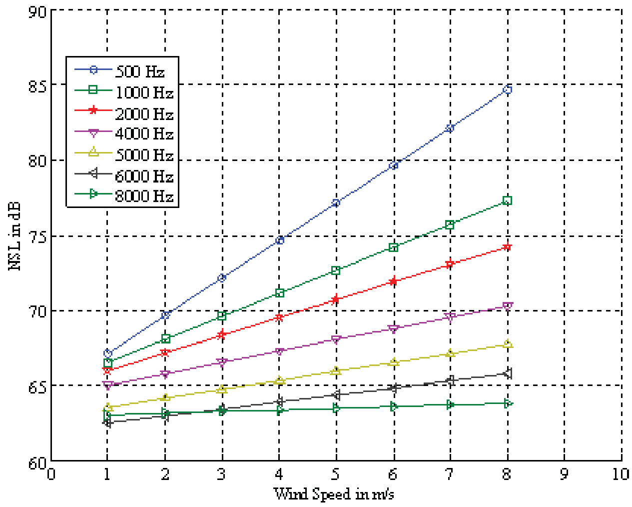
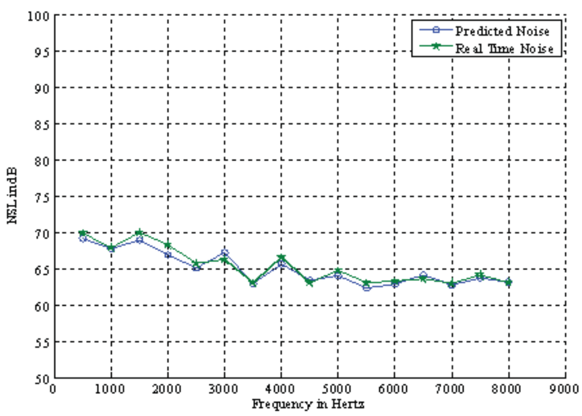


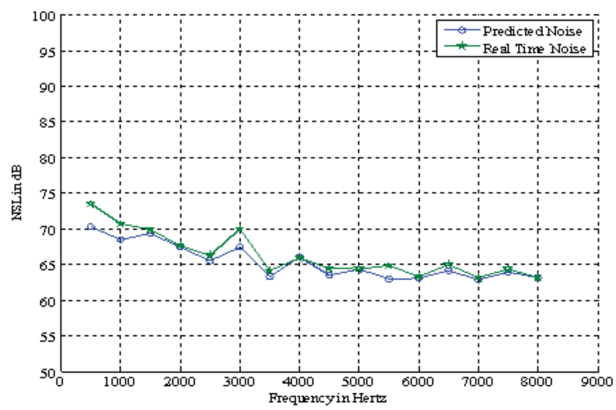
Figure 3. Effect of NSL at different wind speeds for different frequencies

Table 2. Values of *B* and *n* from regression plots

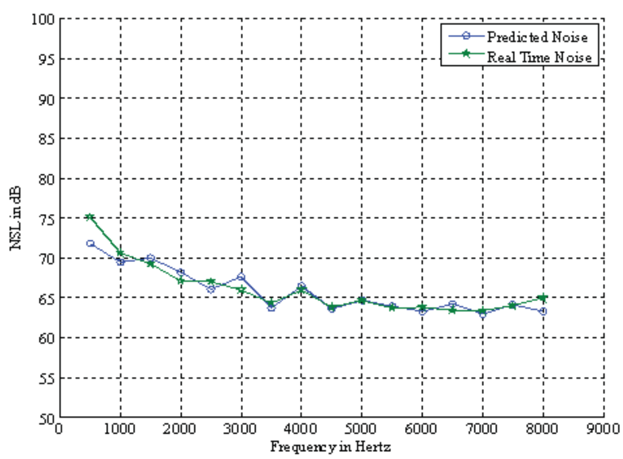
Frequency (Hz)	*B*	*n*
500	67.14	0.13
1000	66.57	0.08
1500	68.16	0.05
2000	65.98	0.06
2500	64.28	0.05
3000	66.93	0.02
3500	62.29	0.04
4000	65.04	0.04
4500	63.19	0.01
5000	63.56	0.03
5500	61.04	0.08
6000	62.5	0.02
6500	64.06	0.003
7000	62.56	0.01
7500	63.41	0.02
8000	63.04	0.006



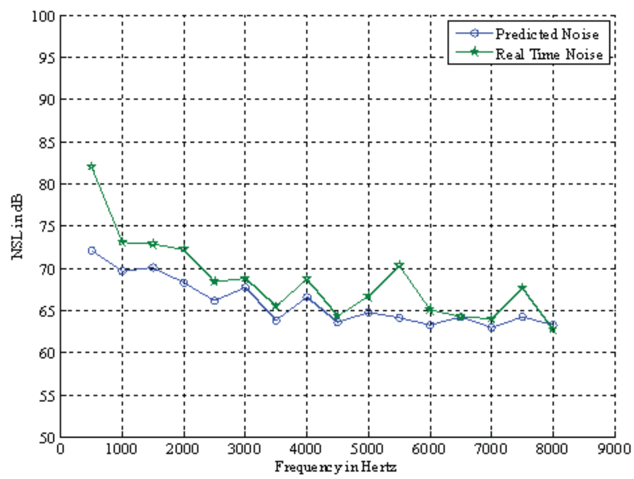
For wind speed of 2.11 m/s



For wind speed of 3.32 m/s



For wind speed of 6.92 m/s



For wind speed of 6.57 m/s

Figure 4. Comparison of predicted and measured noise levels for various wind speeds

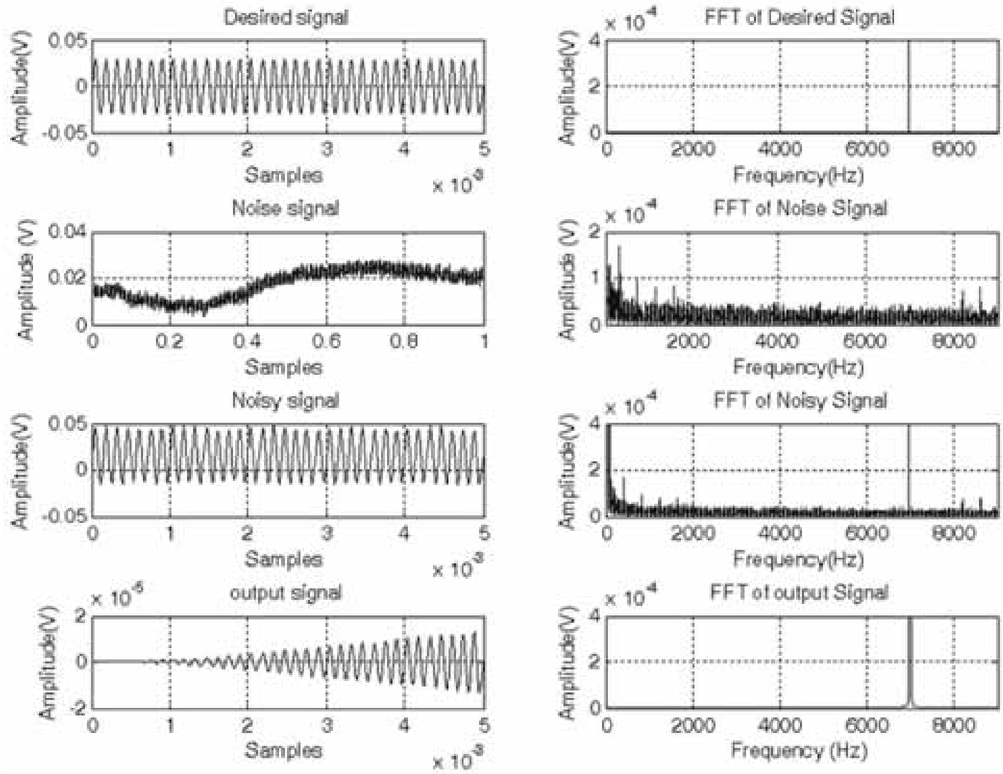


Figure 5(a). Time and FFT representation of desired signal, noise data due to wind, noisy signal 2 (desired signal plus noise) and reconstructed signal by LMS algorithm for a lower wind speed of 2.11 m/s

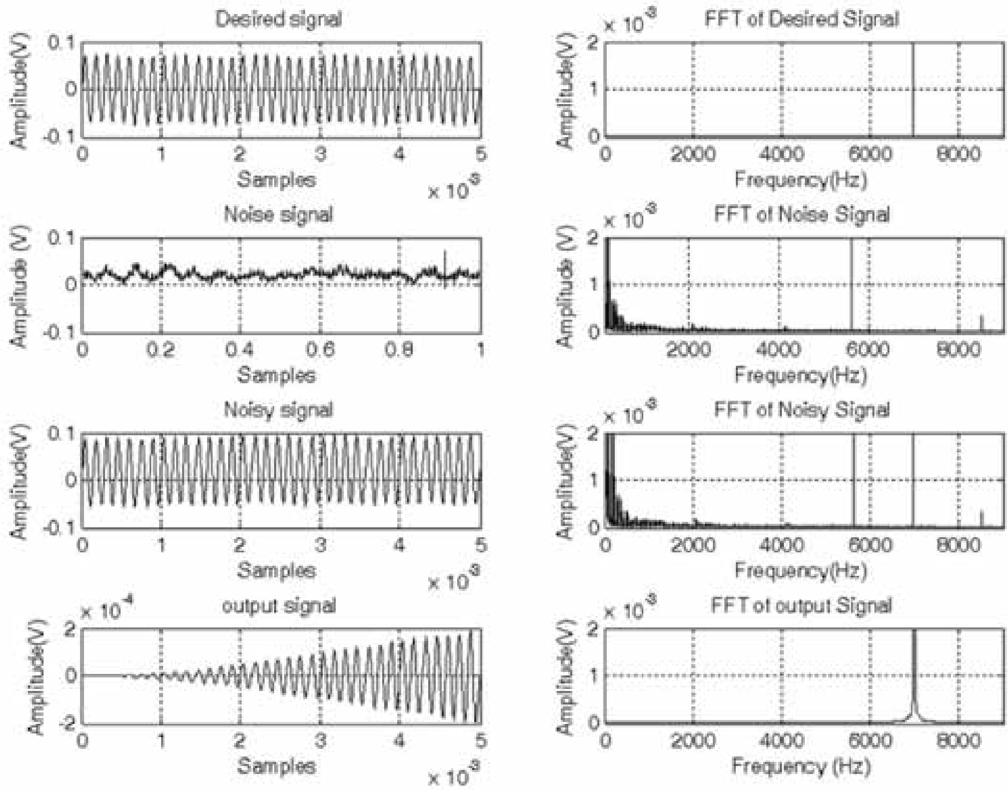


Figure 5(b). Time and FFT representation of desired signal, noise data due to wind, noisy signal 3 (desired signal plus noise) and reconstructed signal by LMS algorithm for a high wind speed of 6.16 m/s

Adaptive filter output

The data collected at 5 m depth for a wind speed of 2.11 m/s is considered as a noise signal. A desired signal $d(n)$ of 7 kHz is passed through the underwater channel where interference signal $v(n)$ with a wind speed of 2.11 m/s gets combined and forms a noisy signal $x(n)$. It can be noted that the amplitude of $d(n)$ is highly affected by $v(n)$. The $x(n)$ is the input to the adaptive filter which uses the LMS algorithm. The adaptive filter adapts to the $d(n)$ by changing the weight update equation $w(n+1)$ from the error signal obtained by comparing $x(n)$ and $d(n)$. The noise due to wind effect gets cancelled and thereby the reconstruction of $d(n)$ of 7 kHz is effectively measured. The same process is carried out for all wind speeds. The time domain and FFT of the $d(n)$, $v(n)$, $x(n)$ and reconstructed signal is obtained by using an LMS based adaptive filter for a minimum wind speed of 2.11 m/s and a highest wind speed of 6.59 m/s are shown in Fig. 5(a) and 5(b). The SNR and MSE of the adaptive filter using LMS algorithm is calculated. It is found that the output SNR of the filter is doubled when compared to the input SNR. The MSE is also reduced. The performance of the LMS based adaptive algorithm is determined for all wind speeds mentioned above and the results are represented in the form of spectrograms.

Spectrogram representation

Here, the spectrogram is a three dimensional representation based on the LMS adaptive algorithm in denoising the wind driven ambient noise. In spectrogram figures, the y axis represents the data collection time-period in seconds and the x axis represents the frequency (Hz) available at the corresponding time. The intensity of the signal available for the total time-period of the experiment carried out is represented by distinct grey patches. Figure 6(a) shows the spectrogram for wind speed of 2.11m/s with $d(n)$ of 7 kHz, $v(n)$ over a range of 100 Hz to 10 kHz, $x(n)$ and the reconstructed signal over the same ranges. The presence of noise signals are represented in dark patches whose intensity varies from 60 to 80 dB as shown in PSD plot. The hydrophone used to collect the data has a capacity to receive signals ranging from 100 Hz to 10 kHz. The high intensity at 0 to 100 Hz in the noise and noisy signals are due to turbulence. This turbulence generated noise signal is also eliminated by the filter.

Similarly the spectrogram is evaluated for all wind speeds ranging from 3.32 to 6.57 m/s and the results on the performance of the LMS adaptive algorithm are shown in Fig. 6(a) to 6(h). In Fig. 6(g) and 6(h), the noise due to marine species at 6 kHz is clearly visible. It is inferred that the adaptive LMS algorithm developed also eliminates the effect due to marine species. Hence it is found that the LMS algorithm eliminates all undesired signals in the range considered and reconstructs the required desired signal against all sources of ambient noise

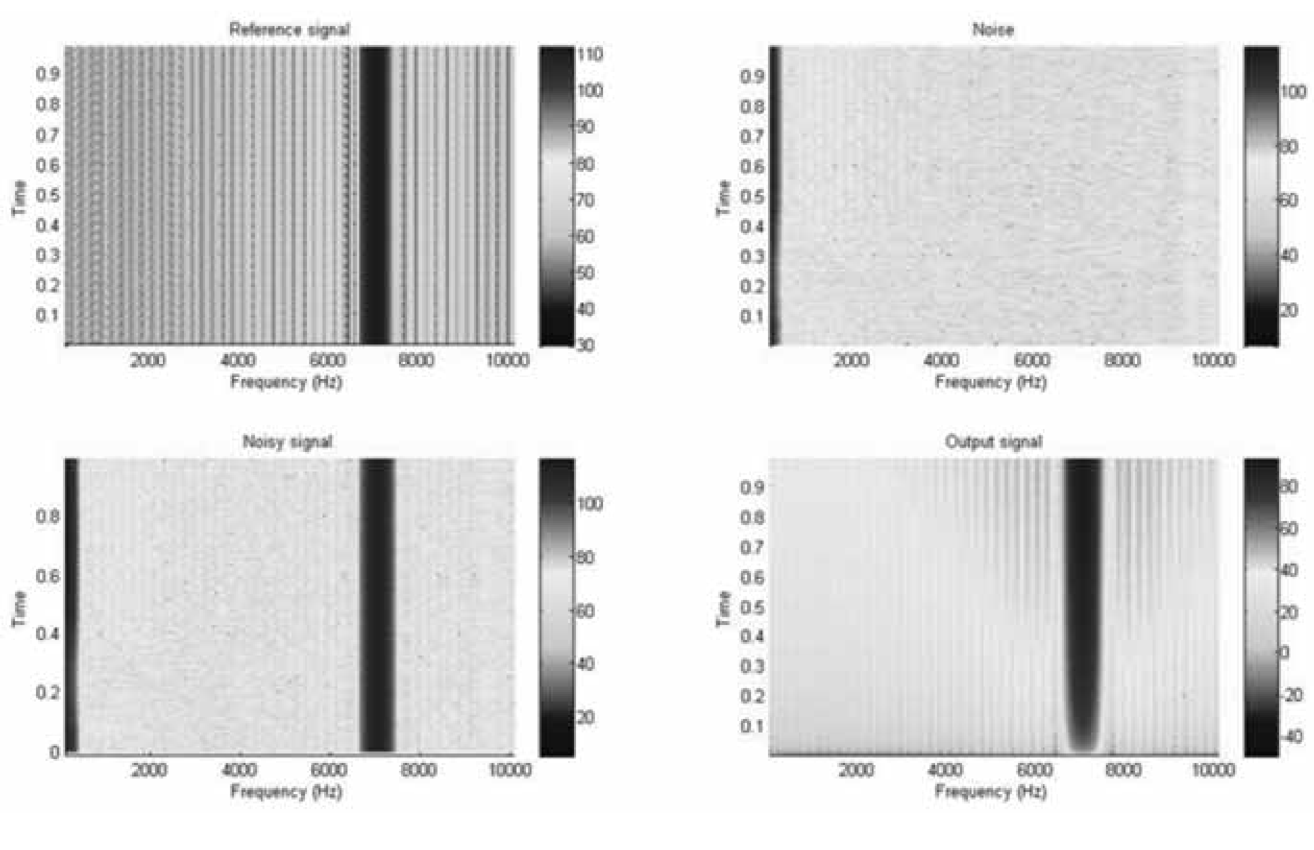


Figure 6(a). Denoising output of wind driven ambient noise by LMS algorithm (2.11 m/s)

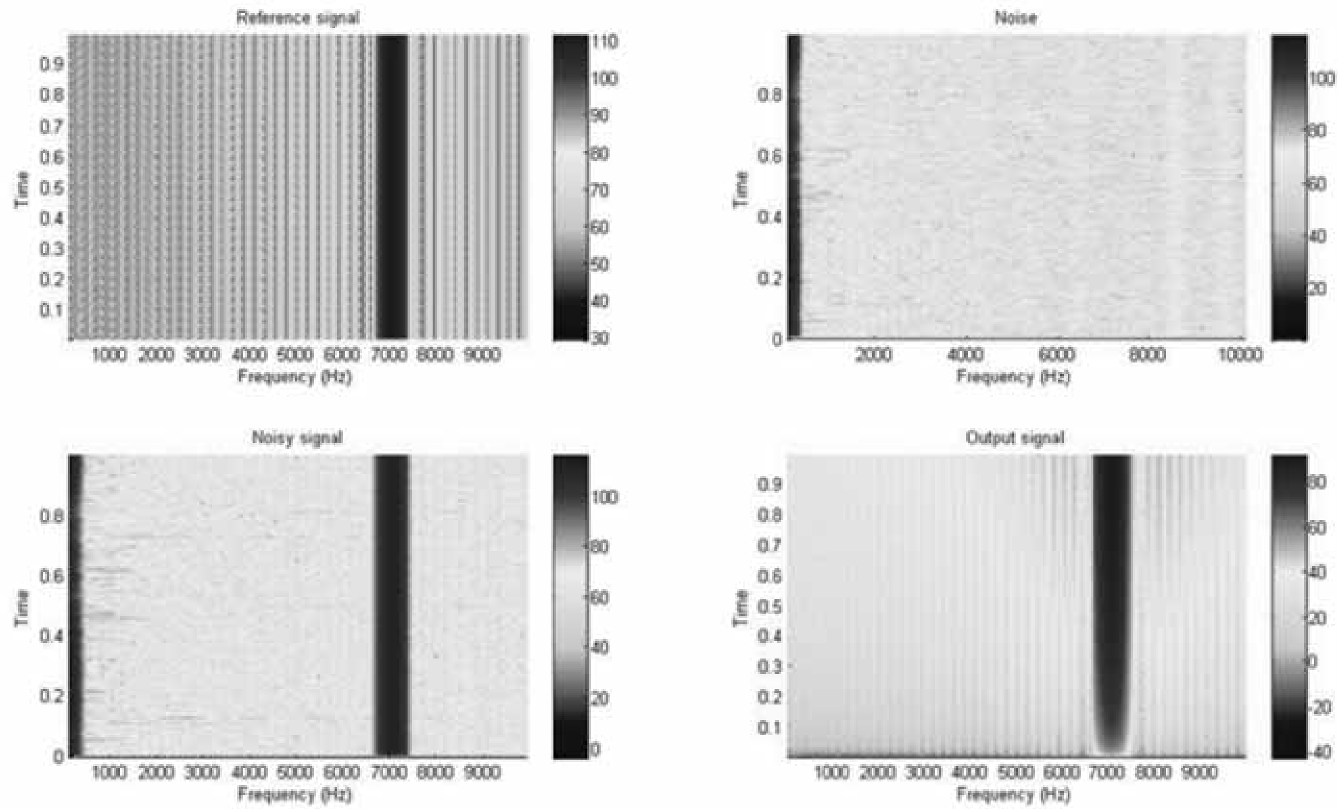


Figure 6(b). Denoising output of wind driven ambient noise by LMS algorithm (3.32 m/s)

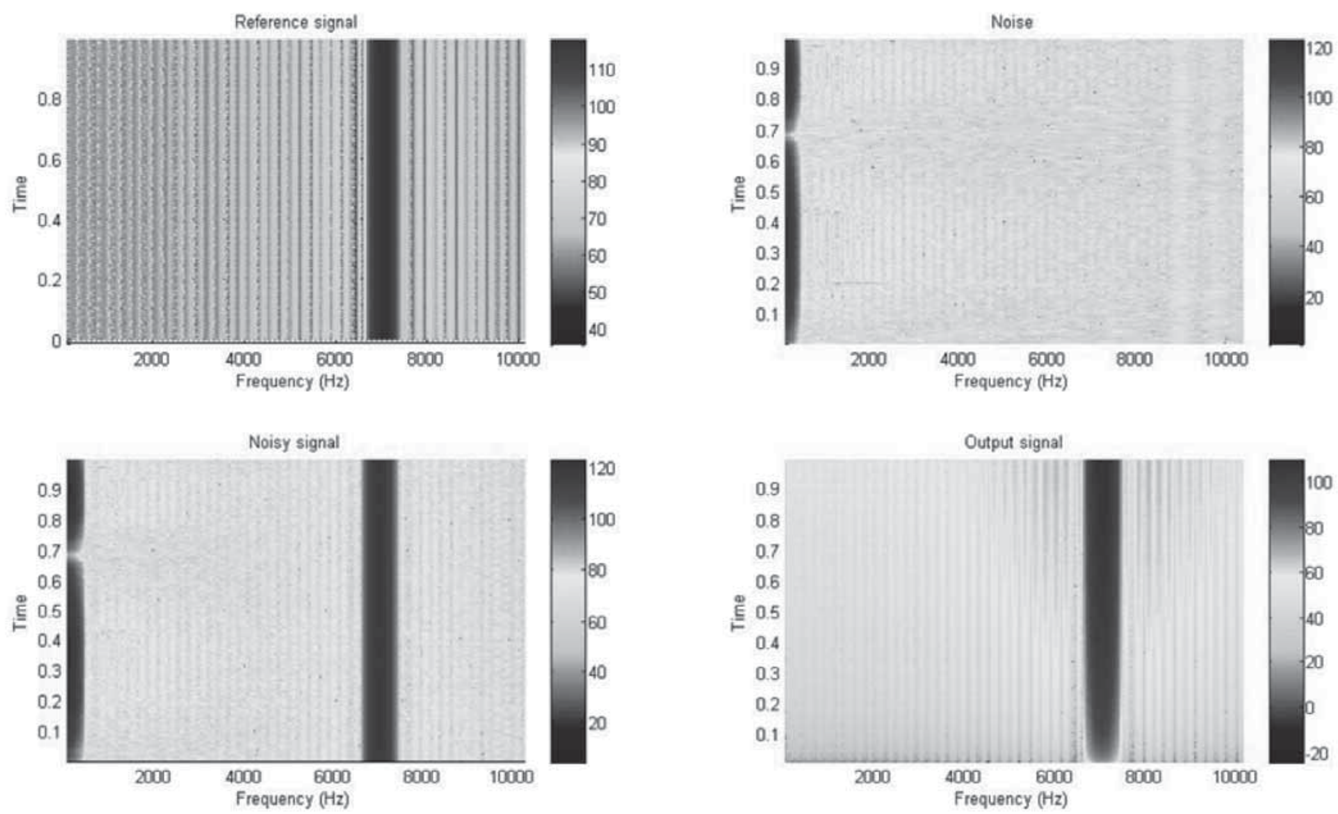


Figure 6(c). Denoising output of wind driven ambient noise by LMS algorithm (4.52 m/s)

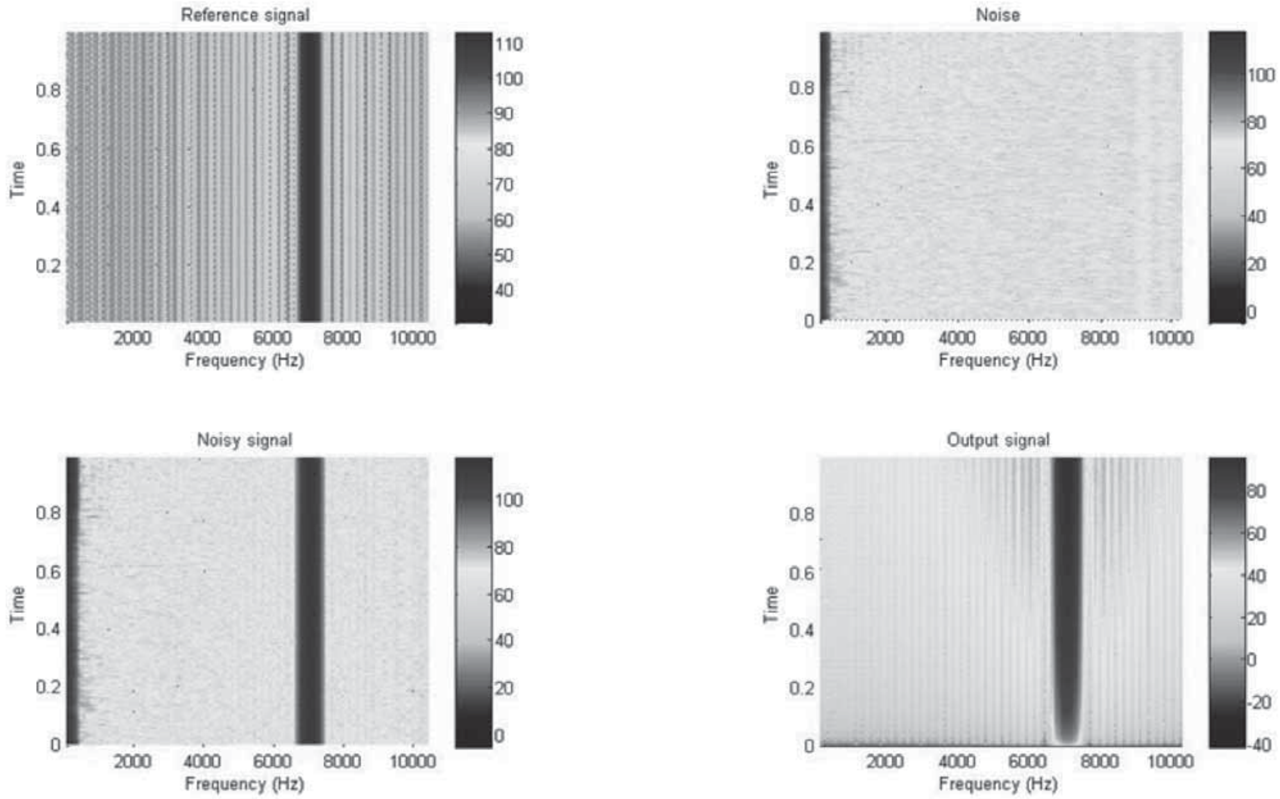


Figure 6(d). Denoising output of wind driven ambient noise by LMS algorithm (5.92 m/s)

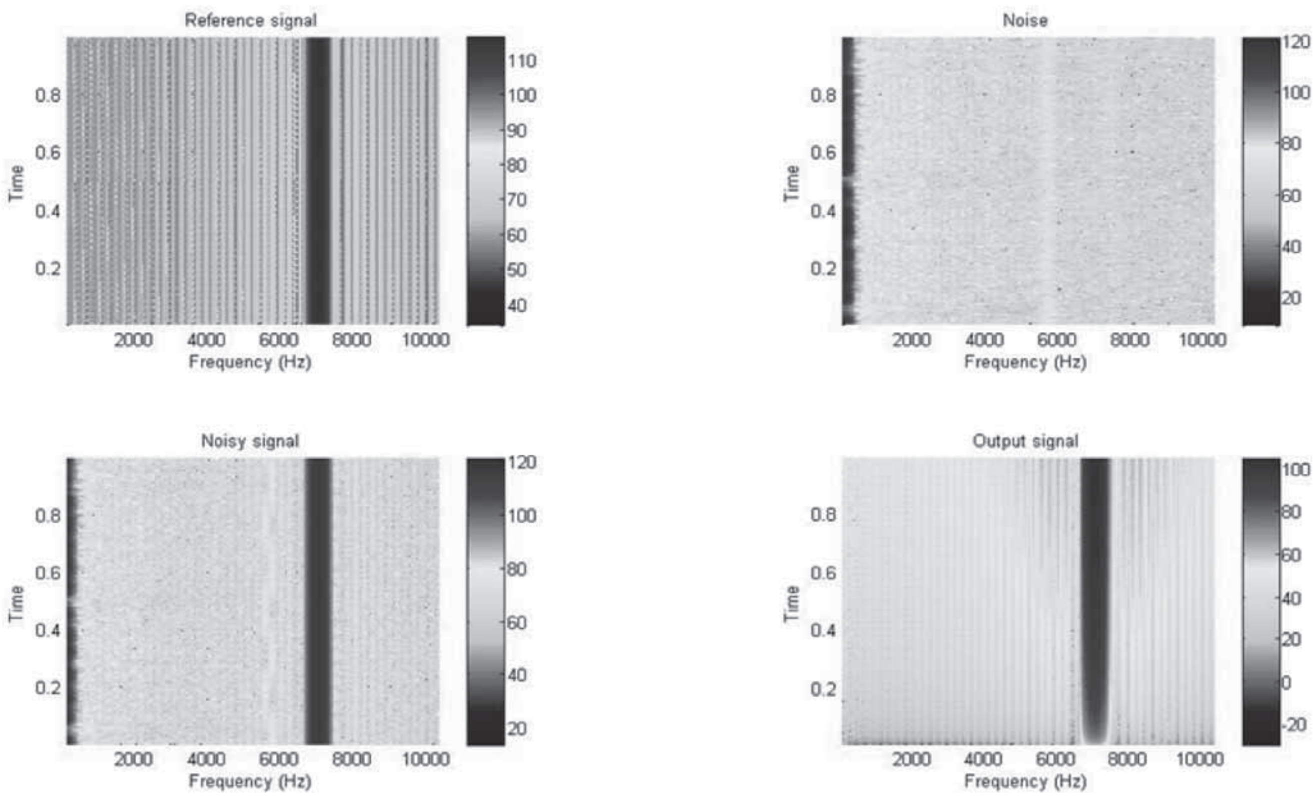


Figure 6(e). Denoising output of wind driven ambient noise by LMS algorithm (6.03 m/s)

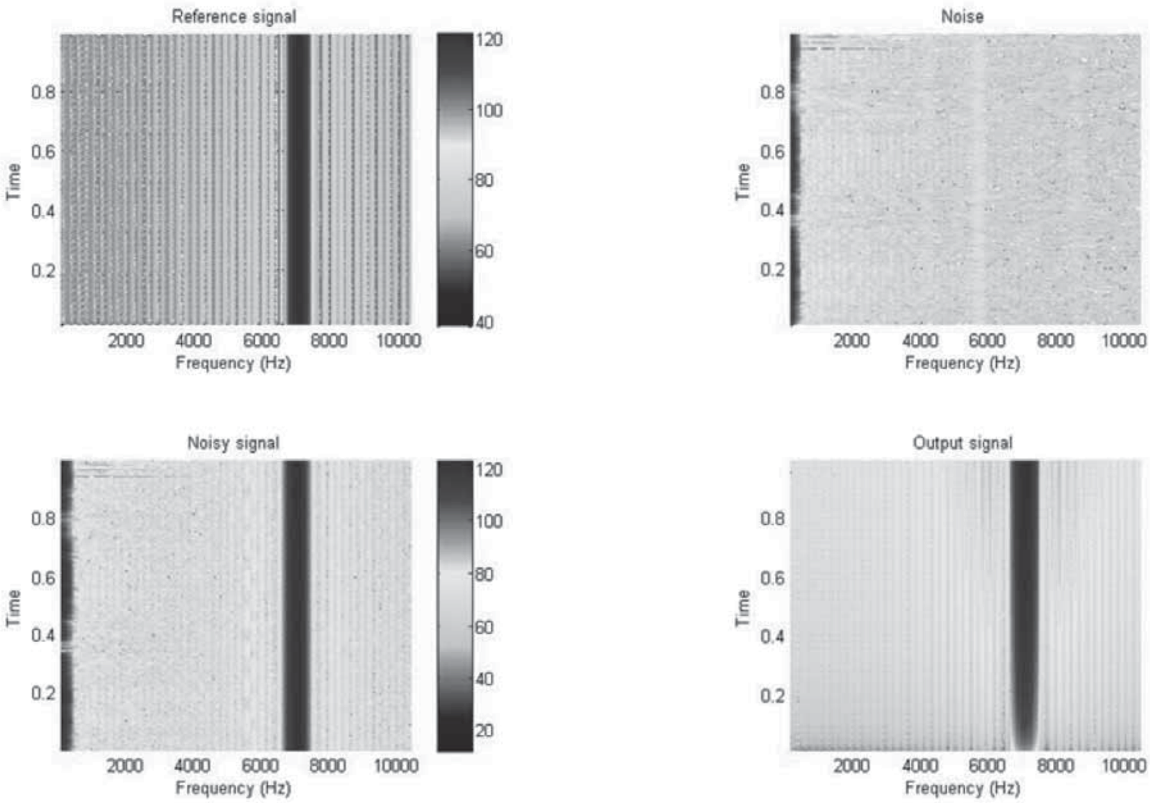


Figure 6(f). Denoising output of wind driven ambient noise by LMS algorithm (6.06 m/s)

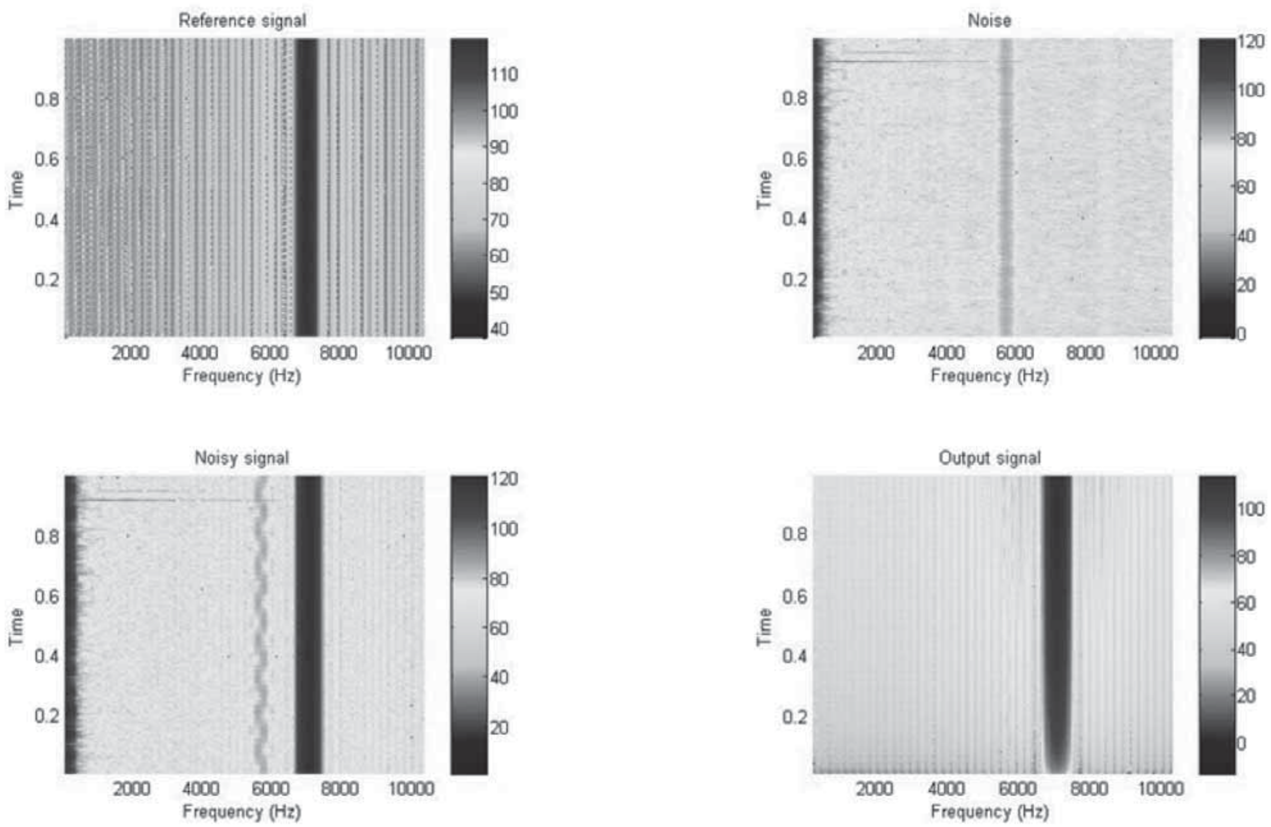


Figure 6(g). Denoising output of wind driven ambient noise by LMS algorithm (6.16 m/s)

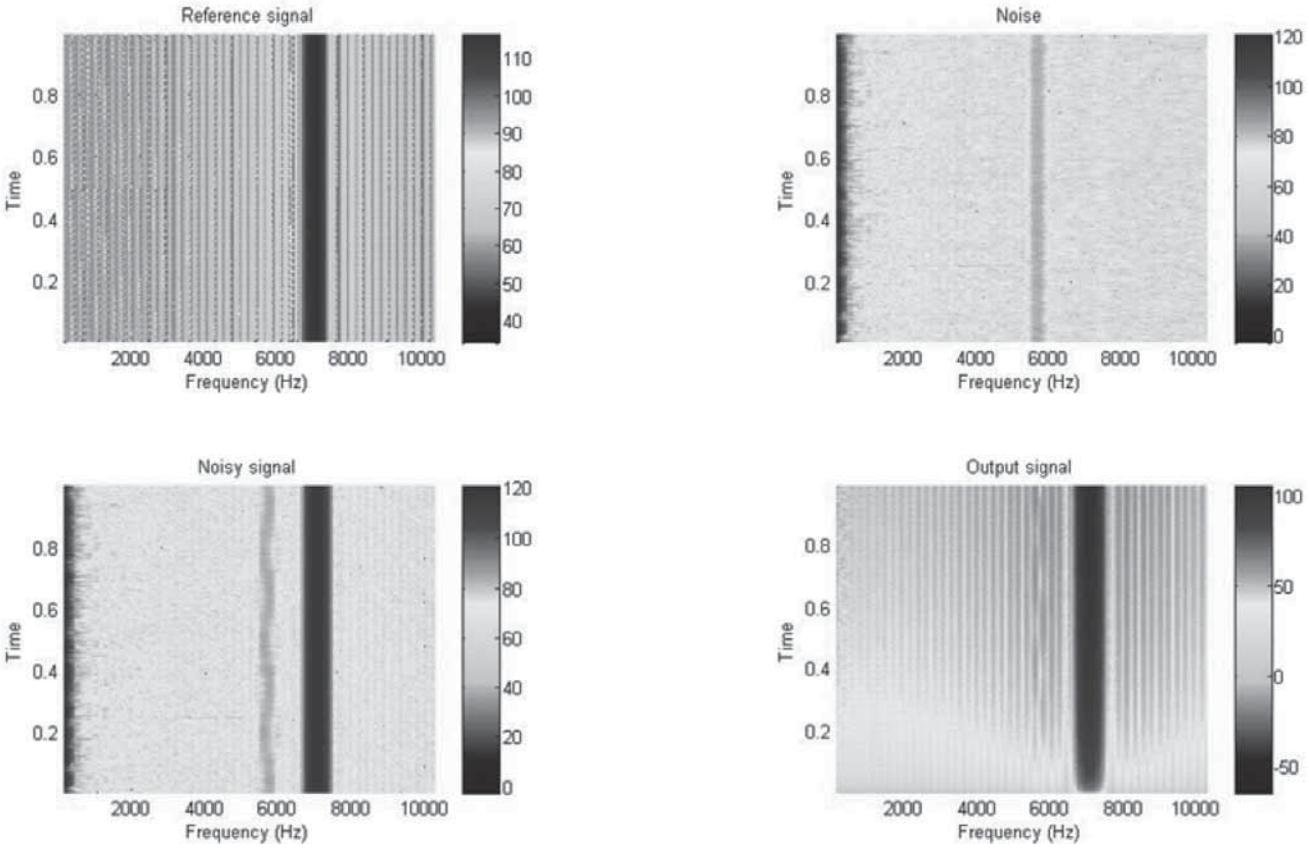


Figure 6(h). Denoising output of wind driven ambient noise by LMS algorithm (6.57 m/s)

SNR and MSE calculation

The SNR and the MSE calculated for all wind speeds considered are shown in Table 3. Using the LMS algorithm the output SNR achieved is around 53 dB for the minimum input SNR of 22 dB to a maximum of 33 dB. Hence, it is clear that there is an improvement in the SNR which ranges from 20 dB to 31 dB with an average of 25.4 dB for all wind speeds considered. Similarly the MSE is reduced from 1.8017 to 0.0195 for 2.11m/s, and similarly for all other wind speeds.

Table 3. SNR Improvement and MSE reduction using LMS algorithm for various wind speeds

Algorithm	SNR (dB) for various wind speeds							
	2.11m/s	3.32m/s	4.52m/s	5.92m/s	6.03m/s	6.06m/s	6.16m/s	6.57m/s
Input SNR	26.76	29.33	22.72	29.91	29.08	28.79	33.20	23.38
LMS	53.26	53.20	53.25	53.30	53.21	53.28	53.23	53.29
Increase in SNR	26.5	23.87	30.53	23.39	24.13	24.49	20.03	29.91

Algorithm	Mean Square Error (No unit) for various wind speeds							
	2.11m/s	3.32m/s	4.52m/s	5.92m/s	6.03m/s	6.06m/s	6.16m/s	6.57m/s
Input MSE	1.8017	1.7377	1.7511	1.7261	0.2165	0.2242	0.1437	0.3842
LMS	0.0195	0.0196	0.0195	0.0194	0.0195	0.0194	0.0195	0.0194

CONCLUSIONS

In this paper, the estimation of power spectral density for ambient noise due to wind at various speeds ranging from 2.11 m/s to 6.59 m/s is analysed and inferred that the effect of wind is dominating at frequencies from 100 Hz to 5 kHz. A noise model for estimating the effect of wind at different wind speeds for various frequencies is developed and found that it suits well with the practical data. The analysis shows that noise level increases as wind speed increases.

An adaptive LMS algorithm is developed to denoise the effect due to wind on any desired signal. The LMS algorithm implemented improves the SNR by 25.4 dB on an average for all wind speeds considered and the MSE is also reduced to an appreciable level. The spectrogram plot is presented for better understanding of the denoising effect due to wind on the signal transmitted in the shallow water region.

ACKNOWLEDGMENT

The authors would like to thank National Institute of Ocean Technology, Chennai for providing data for this analysis. The authors wish to acknowledge the support of Dr G. Latha, NIOT and Dr S. Radha, HOD ECE Department SSNCE.

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The 2012 conference of the Australian Acoustical Society will be held in Fremantle, Western Australia, from 21 to 23 November 2012. Acoustics 2012 Fremantle will be another great opportunity for Australian and International guests to get together to discuss all aspects of acoustics. Below are some updates on key presentations, workshops and dates.

Plenary and keynote presentations

The conference will include many interesting plenary and keynote presentations. Guest speakers include:

- Dr Irene van Kamp of the National Institute of Public Health and the Environment (Netherlands).
- Dr Ross Chapman of the School of Earth and Ocean Sciences, University of Victoria, Canada.

Pre-conference workshops

A variety of specialist workshops/short courses will take place prior to the event, including:

- *Active Noise Control*, University of Western Australia
- *Underwater Passive Acoustic Monitoring*
- *Advanced Machine Diagnostics and Condition Monitoring*, (2 day course), the course will be given by Em. Prof. Bob Randall from UNSW and will be held at Curtin University.

Registrations

Registrations for Acoustics 2012 Fremantle are now open via the ***RegisterNow*** website. Please visit the conference website (http://www.acoustics.asn.au/joomla/acoustics-2012.html), then click on the ***RegisterNow*** link. Alternatively, go directly to the ***RegisterNow*** website (https://www.registernow.com.au/secure/Register.aspx?ID=6324).

Papers

The submitted papers have been reviewed, and the reviews will be released on 27 August 2012. The final papers are due on 19 September.

Please refer to the conference website for all the up-to-date information regarding the conference:
http://www.acoustics.asn.au/joomla/acoustics-2012.html

SUBMARINE UNDERWATER STRUCTURE-BORNE NOISE AND FLOW NOISE DUE TO PROPELLER EXCITATION

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The current study presents the numerical prediction of the noise and vibration of a small-scaled submarine under axial excitation from a 5-bladed propeller and excitation from the flow noise induced by the pulsating pressure of the hull. Firstly, the propeller flow and submarine flow were independently validated. The propulsion of the hull-propeller was simulated using computational fluid dynamics (CFD), so as to obtain the transient responses of the propeller axial excitation and the boundary pressure of the hull. Finally, the acoustic response of the submarine under axial excitation was predicted using a finite element/boundary element model in the frequency domain, and the flow noise was predicted using Curle's analogy in the time domain.

INTRODUCTION

Propeller excitation can induce strong submarine vibration and radiated underwater noise [1,2]. As the propeller is operating in a spatially non-uniform wake of the submarine, the propeller thrust and boundary pressure of the submarine hull are fluctuating, which can generate significant acoustic signature [3]. Early research shows that the sound radiation transmitted through fluid is only 6-8% of that transmitted through shaft, while recent results show that this ratio is between 10-50% [4]. In most studies on submarine noise due to propeller excitation, the propeller excitation is assumed to have constant unit strength [4,5], and the pressure field in the fluid due to the propeller is represented by the fields due to dipoles in different directions at the propeller centre [6]. Merz [7] noted that the combination of CFD and finite element/boundary element (FE/BE) models are the trend for future studies of the propeller induced submarine hull vibration and underwater noise radiation.

In the current work, the propeller excitation and the boundary flow of the hull are simulated via CFD. The structure-borne noise and flow noise of the submarine are predicted using the BEM and Curle's analogy, respectively. The hydrodynamic fields of the submarine and propeller are simulated using CFX commercial software. The frequency response function of the submarine is simulated using ANSYS commercial software and an in-house code is developed to solve the acoustic response. The following assumptions are used: (1) only the axial excitation of the propeller transmitted from the shaft to the hull is considered, and the hull excitation via the fluid is ignored; (2) flow noise associated with fluctuating surface pressures on the propeller blades is not included in the current work; (3) the damping effect of the material is neglected; and (4) the excitations of the propeller are taken as concentrated point forces, not the distributed force on the blade.

ANALYTICAL MODEL

Boundary integral equation for acoustic problem

In a homogeneous medium, for the 3D linear time-harmonic problem for external acoustics using the Neumann boundary condition, the Helmholtz equation is

$$\Delta p(\mathbf{x}) + k^2 p(\mathbf{x}) = 0\,,\ \ \mathbf{x} \in D \tag{1}$$

The solution can be obtained using the Burton-Miller formulation [8]

$$\int_s \frac{\partial G(\mathbf{x},\mathbf{y})}{\partial n(\mathbf{y})} p(\mathbf{y}) dS(\mathbf{y}) + C(\mathbf{x})p(\mathbf{x}) + \alpha \int_s \frac{\partial^2 G(\mathbf{x},\mathbf{y})}{\partial n(\mathbf{y})\partial n(\mathbf{x})} p(\mathbf{y}) dS(\mathbf{y})$$
$$= \int_s G(\mathbf{x},\mathbf{y}) q(\mathbf{y}) dS(\mathbf{y}) + \alpha \int_s \frac{\partial G(\mathbf{x},\mathbf{y})}{\partial n(\mathbf{x})} q(\mathbf{y}) dS(\mathbf{y}) - C(\mathbf{x}) q(\mathbf{x}),\ \mathbf{x} \in \partial D \tag{2}$$

where $\mathbf{x}$ is a general field point, $\mathbf{y}$ is the source point, p is the acoustic pressure, $n(\mathbf{y})$ is the unit normal at $\mathbf{y} \in \partial D$, D is the domain of the propagation, ∂D is the boundary of D, $v_n(\mathbf{x})$ is the normal velocity, $C(\mathbf{x})$ is a geometry related coefficient, $G(r)$=-$e^{ikr}/4\pi r$ is the free space Green's function, with $r=||\mathbf{x}-\mathbf{y}||_2$, α is the coupling constant, q is the derivative of p. Once the sound pressure on the boundary is known, the pressure at any point in the exterior field can be determined by

$$C(\mathbf{x}_i)\, p(\mathbf{x}_i) = \int_s G(\mathbf{x}_i,\mathbf{y}) q(\mathbf{y}) - p(\mathbf{y}) \frac{\partial G(\mathbf{x}_i,\mathbf{y})}{\partial n(\mathbf{y})} dS(\mathbf{y}),\ \ \mathbf{x}_i \in \partial D \tag{3}$$

Curle's analogy for pulsating pressure induced flow noise

In the time domain, the flow noise induced by the pulsating pressure on the solid boundary can be obtained via Curle's analogy [9]

$$p(\mathbf{r},t) = \int_s [\mathbf{n} \cdot \mathbf{r} \,/\, (4\pi c r^2) * (\partial p_s \,/\, \partial t)\,]_\tau \, dS \tag{4}$$

where c is the speed of sound in the fluid and p_s is the boundary pressure. As the submarine is not acoustically compact, (i.e. $\omega L/c\sim1.0$, with L being the submarine length), the time derivative of the pressure is calculated at the retarded time τ. In Eq. (4), the volume source is neglected, because the noise contribution from the flow field surrounding the body is included in the quadrupole sources which is relatively small compared to the term in Eq. (4).

HYDRODYNAMIC AND ACOUSTIC CHARACTERISTICS OF THE HULL-PROPELLER

In this work, a five bladed unskewed propeller model as shown in Table 1 is chosen to match the SUBOFF submarine for two reasons. (1) Firstly, the test data of the propeller and the experimental data of the submarine can validate the numerical results. (2) Secondly, in Ref. [10], the aforementioned propeller was used to thrust a standard axisymmetric submarine hull model (DTMB model 5495-3) in the US Navy's LCC. The parameter values of the SUBOFF submarine are listed in Table 2, and the sail is located on the hull at the top dead centre. The stern appendages are attached to the hull at ϕ=0°, 90°, 180° and 270°, where ϕ is defined positive counter-clockwise as viewed from the stern. Experimental data of the SUBOFF for validation was provided by David Taylor Model Basin (DTMB) in 1988 and 1989 [11]. A number of submarine configurations, ranging from an axisymmetric body to a fully appended submarine were constructed in order to provide flow measurements for the CFD validation. Each model was placed in the Anechoic Flow Facility (AFF) wind tunnel. The flow was measured at a Reynolds number of 1.2×10^7. In the experiment, pressure taps on the hull surface connected to rotary pressure scanners provided measurements for surface pressure. A traversing mechanism was used to position hot film probes in order to measure mean axial components of the velocity profile at non-dimensionalised positions of X/L=0.978, where X is the position along the hull.

Table 1. Parameter values of the propeller

Diameter	Number of blades	Hub-to-diameter ratio	Expanded area ratio	Blade section	Design advance coefficient	skew	rake
0.25m	5	0.2	0.725	NACA66 (modified)	0.889	0	0

Table 2. Parameter values of the submarine

Submarine length L	Maximum diameter D_s	Forebody length	Midbody length	Afterbody length	Appendages
4.36m	0.51m	1.02m	2.23m	1.11m	NACA0020

Hydrodynamics of independent propeller and independent submarine

Firstly the single passage of the propeller is meshed, with the local areas such as the tip and the root refined, as shown in Fig. 1. The total number of the single passage meshes is 0.38M, the y-plus value on the boundary of the blade is about 84, herein the y-plus value represents the non-dimensional distance of the first node from the wall, which is recommended in the scope of 20 to 100, and in this case the mixed formulation of wall function (i.e. automatic near wall treatment in CFX) is selected. This mixed method is available for all frequency equations based turbulence models, which automatically switch from a low-Reynolds number formulation to wall functions based on the grid spacing provided by the user. The mixed formulation provides the optimal boundary condition for a given grid. The calculation domain of the independent propeller is shown in Fig. 2. The propeller angular speed is 650rpm in this section, and the propeller rotates anti-clockwise, looking forward through the propeller disc. Here, the advance ratio of the propeller changes from 0.1 to 1. For the computational models the inlet boundary conditions consists of the prescribed velocity profile. At the outlet $p_{ave}=p_0$ and $(\nabla\mathbf{v}).\mathbf{n}=0$ and no-slip conditions are used on the blades. Finally the thrust coefficient K_t, the torque coefficient K_q, and the open water efficiency η can be determined via CFD, with $K_t=T/(\rho N_p^2D^4)$, $K_q=Q/(\rho N_p^2D^5)$, $\eta=(J/2\pi)K_t/K_q$, where T, Q ,N_p, D are respectively the blade thrust, torque, rotating speed and diameter. The numerical model is based on the Reynolds Averaged Navier-Stokes (RANS) equations, with the body forces due to the blade's rotation being treated based on the quasi-steady Multiple-Frame-of-Reference method. The turbulent flow within the blade is formulated in a rotating reference frame, the Shear Stress Transport (SST) turbulent model is adopted in this paper [12]. As shown in Fig. 3, the K_t, K_q values of the propeller agree well with the experimental results, whilst the open water efficiency η slightly differs.

For the hydrodynamic field of the submarine independently from the propeller, a circular computational domain is used. The computational domain size is set according to Ref. [11]. The domain extends one hull length upstream of the submarine model and two hull lengths downstream of the model, thus being $4L$ in overall length. The outer diameter of the cylinder is $10D_s$. The inflow velocity is V_s=3.036m/s and the SST turbulent model is adopted. The inlet boundary condition consists of the prescribed velocity profile. At the outlet $p_{ave}=p_0$ and on the hull $(\nabla\mathbf{v}).\mathbf{n}=0$. No-slip conditions are used. The far-field boundary conditions are used at the circumferential boundaries of the computational domains, as described in Ref [11]. To capture the flow detail on the hull and appendages, the meshes near the boundary, the sail and the rudders are refined. From a mesh independent analysis, when the total mesh number reaches 2.06M, the resistance of the submarine converges at 100.2N, with 2.05% relative error compared with the experiment. To further validate the accuracy of the submarine flow, the static pressure coefficients C_p along the meridian line of the hull, at 50% of stern appendage-tip chord length, and at 10% of sail-tip chord length are compared with the experimental results from Ref. [11] in Fig. 4, with $C_p=(p-p_0)/(0.5\rho V_s^2)$, where p is the local static pressure and p_0 is the ambient pressure. Thereafter, the axial non-dimensional velocity with the reference velocity being V_s at streamwise of X/L=0.978 is also validated, as shown in Fig. 5. The contour changes from the range of a minimum value of 0.45 to a maximum of 0.9 with an increment of 0.05, and shows good agreement.

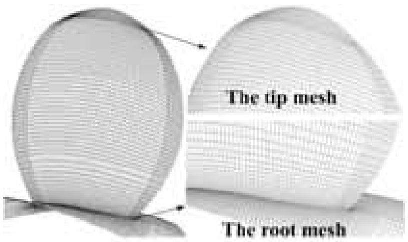

Figure 1. Meshes of the propeller

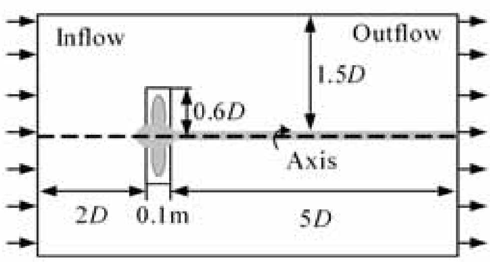

Figure 2. Calculation domain of the propeller

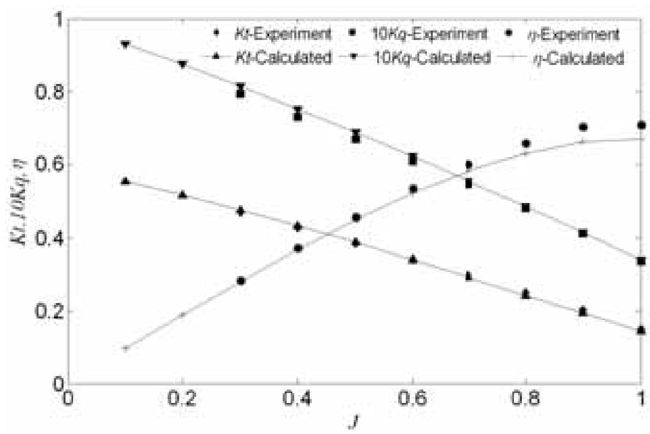

Figure 3. Open water characteristic of the propeller

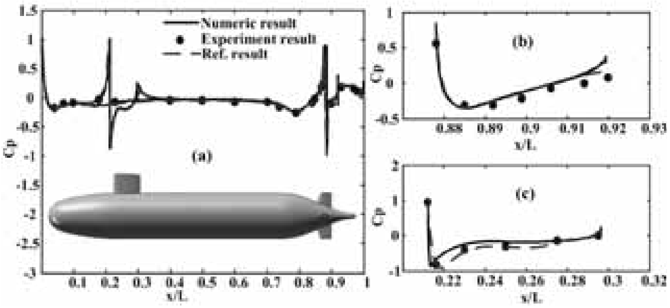

Figure 4. C_p comparison (a) C_p along the meridian line of the hull, (b) C_p at 50% of stern appendage-tip chord length, (c) C_p at 10% of sail-tip chord length

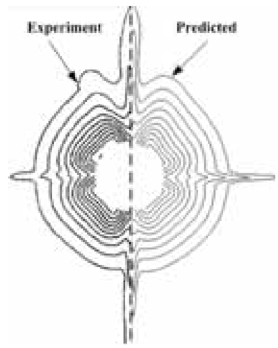

Figure 5. Axial component velocity profile at X/L=0.978

Hydrodynamics of hull-propeller system

As the independent flow of the propeller and submarine is validated, the propeller and submarine are now combined to determine the steady-state responses. The total meshes of the system are shown in Fig. 6, with the global number of nodes and elements 4.13M and 3.97M, respectively. The y-plus value is about 80, and the automatic near wall treatment is adopted. Firstly a steady flow of the system is simulated via CFD, with the advection term and the momentum equation discretized by the second order upwind scheme. Changes in the propeller rotation speed (N_p), the propeller thrust (T), and the submarine resistance (R) allow the operating point to be determined at the chosen inflow velocity of V_s=3.036m/s, as shown in Fig. 7. The steady-state response is then determined at the intersection of the two curves (T~N_p and R~N_p). The parameter values of the hull-propeller at the steady-state are listed in Table 3. Under such circumstances, the propulsion factors such as the propeller advance ratio J, the wake fraction w, and the thrust deduction t, can be determined and respectively correspond to J=0.849, w=0.225 and t=0.202.

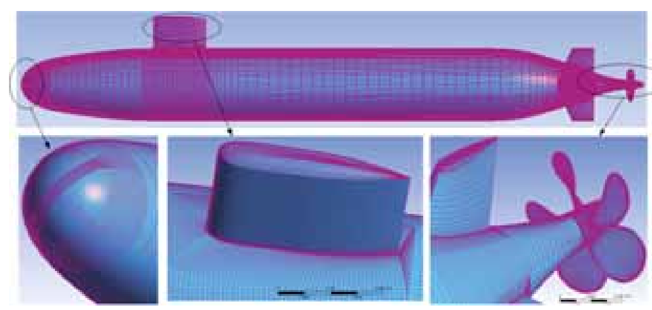

Figure 6. Meshes of the hull and propeller

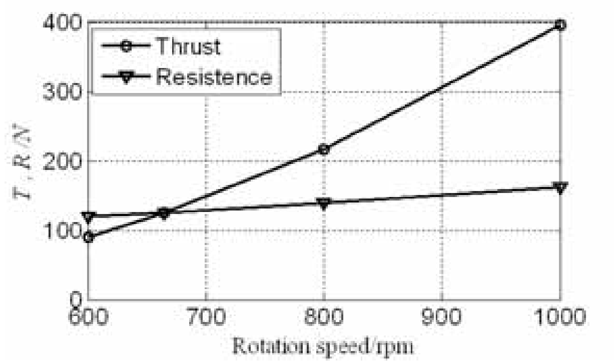

Figure 7. Propeller thrust (T) and submarine resistance (R) versus propeller rotation speed (N_p)

Table 3. Parameter values of the hull-propeller at steady-state

Inflow velocity	Propeller speed	Thrust	Resistance
3.036m/s	665rpm	125.82N	125.57N

Taking the steady flow result as the initial condition of the transient flow calculation, the propeller axial force and the pulsating pressure of hull are obtained. In the transient analysis, the meshes of the propeller are physically rotated, and the time step is 0.001s with a total simulation time of 10s. The SST turbulent model is adopted with the advection term. The momentum equation is discretized by the second order upwind scheme. The transient formulation is solved by the second order implicit scheme. Then the propeller thrust is obtained by integrating the pressure on the blade surface

$$T = \int_S p n_x dS \tag{5}$$

Figure 8 shows the propeller thrust and monitor pressure in front of the propeller. Both are tonal at the propeller harmonics in the frequency range up to 500Hz.

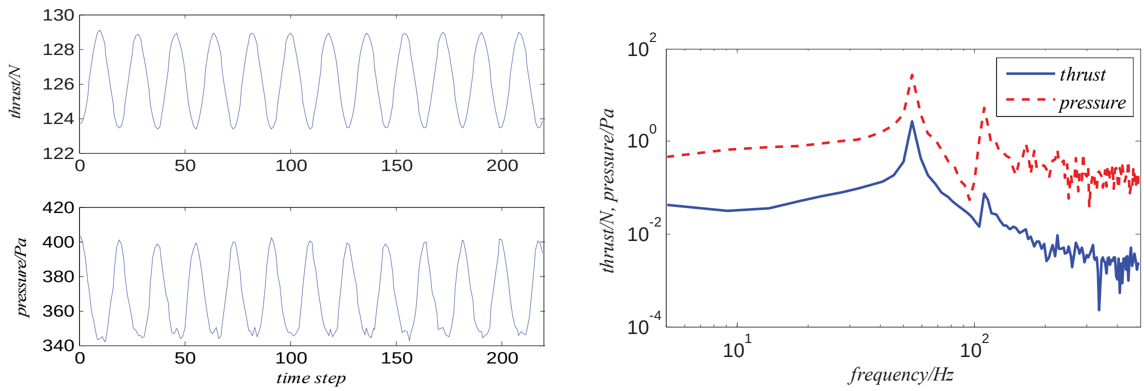

Figure 8. Fluctuations of the thrust and pressure in the time domain (left) and in the frequency spectrum (right)

Acoustic response analysis

In this section, the structure-borne noise and the flow noise of the submarine are predicted. Firstly, the finite element model of the submarine is built, as shown in Fig. 9. The structure is divided into five compartments by four bulkheads with ring stiffeners and longitudinal rib stiffeners, as shown in Fig. 9(b). To ensure the local intensity of the sail and "+" typed rudders, the appendages are also stiffened. The hull and bulkheads are represented by 26,739 SHELL63 elements, and the ribs by 5,517 BEAM188 elements. The SHELL63 is a type of elastic element with 6 degrees of freedom (DOF) which is always used to model both in-plane and out-of-plane vibration of a thin plate. The BEAM188 is a type of line element with 6 DOF which is suitable for analysing slender to moderately thick beam structures. The thickness of the hull and the bulkheads is 6mm. The rib section on the hull is of an inverted T shape, and the rib section on the appendage is of an H shape, as shown in Figs. 9(c) and (d), respectively. The material of the submarine is steel, and the structural loss factor is ignored. Considering the fluid-structure interaction, the surrounding fluid of the submarine hull is also modelled to couple the pressure and structural velocity on the boundary nodes. The axial excitation is then loaded on the bulkhead centre of the cabin near the stern, as shown in Fig. 9(a), and the structural response of the hull is calculated. Here, the shaft and the bearing dynamics are not considered in the FE model. The normal velocity of the hull was used as the boundary condition of the boundary element (BE) model. The finite elements of the hull were directly used as the boundary elements to ensure the coincidence of the FE/BE models, with no error of the data projection introduced. In this work, FEM and BEM are separately used. The fluid-structure interaction effect is modelled by FLUID30 element in FEM, which differs from the method in Ref. [7].

To predict the flow noise, the pressure and the geometry information of the submarine should be known. Here, the boundary pressure of the hull at each time step, the area, and the normal vector are exported from CFD. Finally, the sound pressure is predicted using Eq. (4) in the time domain and the flow noise is transformed to the frequency spectrum using FFT. Thereafter the structure-borne noise and flow noise of the submarine are evaluated at a field point P that is 10L downstream. The reference pressure is 1μPa and the frequency range is 0 to 500Hz. The result in Fig. 10 shows the presence of blade pass frequency (BPF) tonals at multiples of 55.4Hz in both the structure-borne noise and flow noise including the effects of propeller rotation, while the effects of resonant hull responses to broadband random excitation can be seen at 238Hz and 400Hz. Flow noise including the effects of rotation falls with increasing frequency above around 350Hz. The frequency response function (FRF) of the normal displacement of the submarine shown in Fig. 11 shows the principal bending mode of the hull at 180Hz, the breathing mode at 238Hz, the resonant mode of the rudders at 371Hz and the circumferential mode of the cabins at 400Hz. As the field point is located at the end of the submarine, the breathing mode has a significant contribution to the sound pressure.

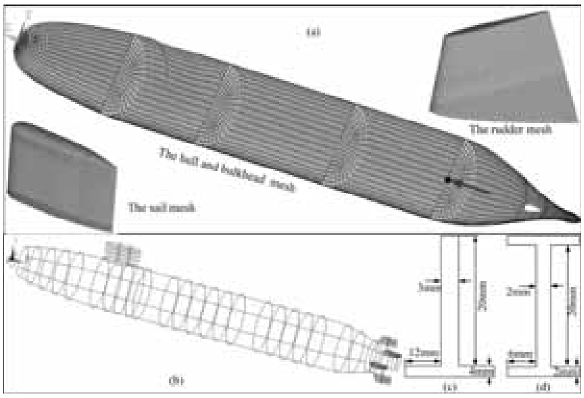

Figure 9. Finite element model of the submarine showing (a) shell elements, (b) beam elements, (c) ribs of an inverted T shape, (d) ribs of an H shape

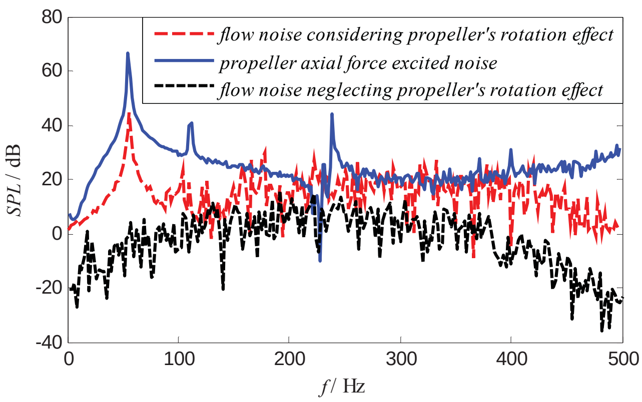

Figure 10. Submarine underwater noise at the field point P

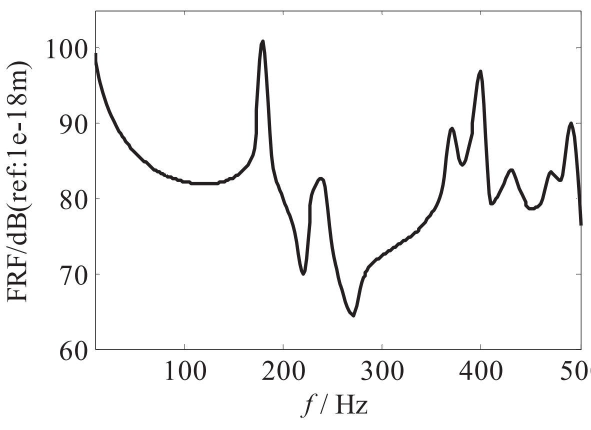

Figure 11. Frequency response of the normal displacement of the submarine under unity axial force

Figure 12 presents the time history of the flow noise considering the propeller rotating effect. A distinct fluctuation of the signature is observed. Figure 13 shows the maximum sound pressure level of the structure-borne noise and the flow noise in the horizontal plane at a distance of 10*L* from the submarine centre at the propeller harmonics. Figure 14(a) and 14(b) show the sound directivity of the two types of noise at the first harmonic of the propeller. The zero degree refers to the submarine stern part. In order to analyse the effect of the rotating propeller on the flow noise, the flow noise in the absence of the rotating propeller is also plotted in Fig. 10 and Fig. 14(c). Figure 14(c) represents the sound directivity of the flow noise at 354Hz when the propeller rotating effect is ignored. As shown in Fig. 10, at 354Hz the flow noise begins to fall. Results at the selected speed show that (1) the SPL of the structure-borne noise and flow noise differs by more than 20dB at BPF and by around 10dB at 2BPF. At 3BPF to 5BPF, the flow noise surpasses the structure-borne noise by 4dB to 10 dB. It can be concluded that the flow noise is about 10% of the structure-borne noise at propeller blade passing frequency. (2) The noise due to the axial force is mainly radiated from the conical end caps, while the noise due to the flow is mainly radiated from the cylindrical hull. As the axial force mainly excites the submarine axial mode (or breathing mode), the majority of the sound energy is radiated from the conical end caps. (3) Due to the rotating effect of the propeller, the directivity of the flow noise is asymmetric relative to the axis of the submarine. When the propeller rotates in the wake of the submarine, the boundary pressure of the hull is influenced by the propeller, and this represents the effect of rotating forces and volumes. To investigate this phenomenon, the flow noise is also calculated when propeller meshes are excluded in the flow field of the submarine, so that the boundary flow of the hull is not affected by the propeller. Under such conditions, the SPL of the flow noise is plotted in Fig. 10. The maximum SPL is about 20dB, and the sound directivity is symmetric relative to the vertical plane of the submarine as shown in Fig. 14(c), but with relatively small magnitude.

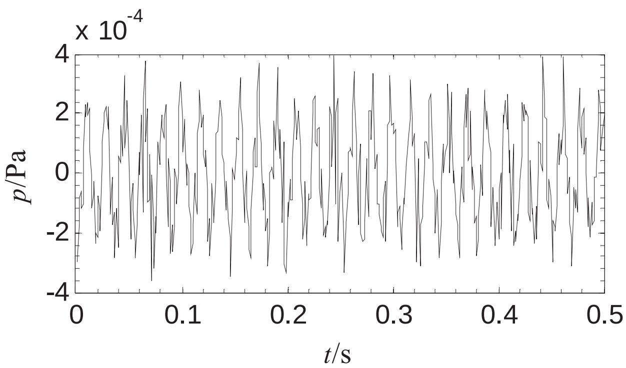

Figure 12. Time history of the flow noise

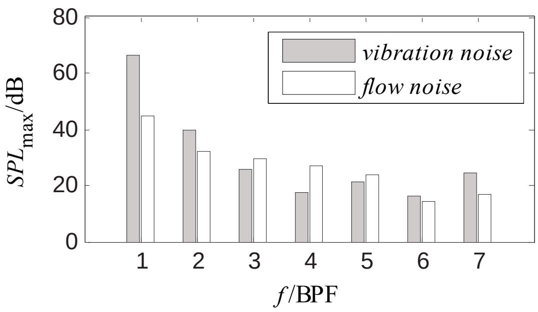

Figure 13. Maximum sound pressure level at propeller harmonics

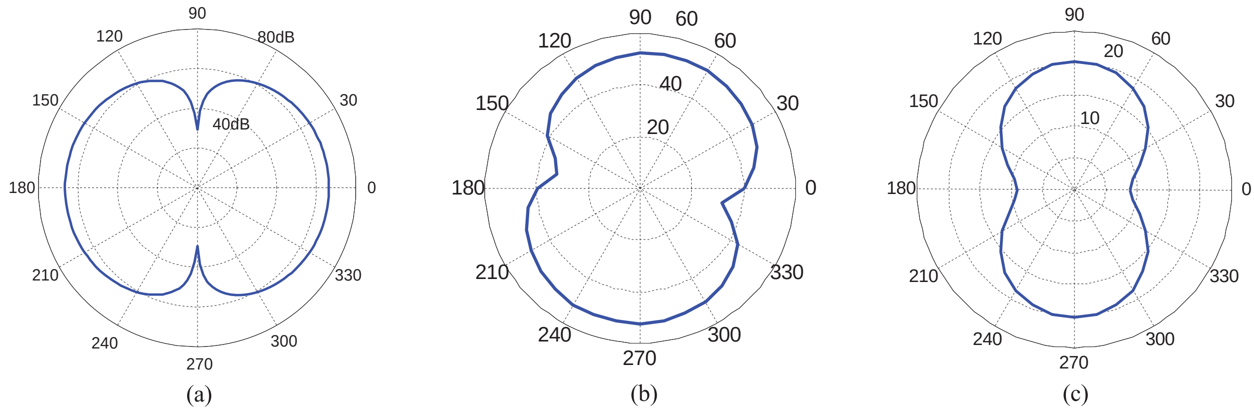

Figure 14. Sound directivity pattern in the horizontal plane at a distance 10L showing (a) structure-borne noise at BPF, (b) flow noise considering the rotating effect of the propeller at BPF, (c) flow noise ignoring the rotating effect of the propeller at 354Hz

CONCLUSIONS

This paper presents the structure-borne noise and flow noise of a submarine under axial excitation from a propeller running at low Reynolds number. The results have been derived for a specific set of model parameters with a small-scale model. In current work, only the flow noise associated with fluctuating surface pressure on the submarine hull is considered. Results show that (1) under axial excitation, the principal breathing and bending modes plus the resonance mode of the rudders and circumferential modes of the cabins generate strong structural responses. The resulting sound pressure at the field point is tonal at the propeller harmonics and structural resonances. (2) The flow noise occurs mainly around the propeller harmonics. (3) The noise due to the axial force is mainly radiated from the conical end caps, while the noise due to the flow is mainly radiated from the cylindrical hull. (4) The flow noise is lower than the structure-borne noise at the blade passing frequency (BPF), and higher than the structure-borne noise at higher harmonics of BPF. (5) Due to the propeller rotating effect, the directivity of the flow noise is asymmetric relative to the axis of the submarine.

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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.

MANIPULATION OF INTERAURAL LEVEL COUPLED WITH A PERFORMER'S HEAD MOTION FOR HEADPHONE REPRODUCTION OF PIANO SOUND

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INTRODUCTION

Modern electronic instruments not only provide high quality musical sound reproduction, but also allow performers to select public or private monitoring (via loudspeakers or headphones). While the tonal quality of the musical sound can be matched relatively well between these two monitoring choices, headphone-reproduced sound has a potentially objectionable spatial aspect in that no change in the sound typically results when a performer's head moves. While a straightforward solution would be to filter the sampled instrument sounds using dynamic update of Head Related Transfer Functions (HRTFs), coupled with a motion-tracking sensor that enables the headphone reproduction of spatially stabilized virtual sound source [1], such a system faces a significant challenge to performer acceptance since the HRTFs induce an apparent change in the instrument timbre. Since maintaining authentic character of the timbre is considered to be more important than enhancement of spatial attributes for most electronic instruments, a relatively simple cue, such as Interaural Level Difference (ILD), might provide a better solution for dynamic spatial sound processing in response to the performer's head movement (as proposed in [2]). In order to find such effective yet simplified means for both spatially and timbrally stable headphone piano sound, an empirical study of head-related responses was executed in which measures were made of the level variation in broad frequency bands that occurred at a performer's average head position as a function of two variables: one variable was the angle of rotation of the performer's head in the horizontal plane (i.e. yaw angle) and the other was the pitch of the note played on the piano (i.e., the variation in the piano sound as each of the 88 keys were depressed).

ACOUSTICAL MEASUREMENT

In order to provide an empirical basis for a parametric manipulation of headphone-reproduced interaural level at a piano performer's ears, a set of acoustical measurements were made using a sphere microphone (SCHOEPS KFM360) that was designed to capture signals in a manner that resembled human interaural differences in level and delay. The sphere microphone was placed at a location approximating the average location of a performer's head in front of the piano and aligned such that the center of the microphone system faced the piano's center, above which an additional cardioid microphone was positioned (this microphone was used to provide a reference for the signal captured by the sphere microphone). The sphere microphone was rotated ±40° variation, with higher angular resolution for yaw angles at which the performer's nose was pointed nearer to the center of the piano. To capture consistent performance of the 88 piano notes for the 17 yaw angles tested, we utilized a MIDI-controlled acoustic piano (the YAMAHA DISKLAVIER) with notes played at a constant MIDI key velocity (100) and constant duration (3 seconds). The recorded data, comprising 1496 notes in total (17 yaw angles by 88 notes), was collected in a studio space that was not overly dry (i.e., not anechoic, as might be preferred for isolating the direct sound from the piano).

PARAMETERISATION

While the variation in level of the signal received at ear position that occurred with changes in the yaw angle of the receiver (the sphere microphone) was captured separately for all 88 notes of the piano, there was a practical consideration that required a parametric representation of these variations. Therefore, a simplified gain function was developed using a quadratic polynomial fit to data level obtained for a $\pm$40° variation in yaw angle. The gain values were calculated relative to the RMS level of the same note captured by the reference microphone. This data was reduced to a representation of ILD that remained constant for subsets of adjacent notes to be reproduced by an electronic piano. It was decided that all frequencies below A4 (with fundamental frequency 440Hz) would have the same gain function applied, since the observed ILD variation was in fact quite similar at these frequencies. Next, five more subsets of notes were chosen, with note ranges that were initially determined through visual inspection of the ILD functions. Then the boundary MIDI-notes at which changes in the gain function would occur was iteratively adjusted to minimize the difference between the original level changes over all 88 notes and the parameterized level changes.

Table 1 shows for the resulting six ranges of note numbers and the two polynomial equations (for +yaw and -yaw, respectively) that parametrically represent the gain variation within each subset.

Table 1. Equations representing the gain variation for each ear

Note Number	Ipsilateral Equation (+yaw Θ)	Contralateral Equation (-yaw Θ)	Frequency Region (Hz)
1 to 48	$0.001\Theta^2+0.1213\Theta$	$-0.0008\Theta^2+0.1298\Theta$	< 440
49 to 56	$0.0025\Theta^2+0.2367\Theta$	$-0.0014\Theta^2+0.2144\Theta$	440 to 700
57 to 62	$0.0059\Theta^2+0.4222\Theta$	$-0.0021\Theta^2+0.2924\Theta$	700 to 1000
63 to 72	$-0.0044\Theta^2+0.1502\Theta$	$-0.0014\Theta^2+0.2348\Theta$	1000 to 1760
73 to 78	$0.005\Theta^2+0.5041\Theta$	$-0.0046\Theta^2+0.3513\Theta$	1760 to 2500
79 to 88	$0.0027\Theta^2+0.4284\Theta$	$-0.0026\Theta^2+0.2882\Theta$	> 2500

CONCLUSION

This paper derived a simplified means for controlling the level variation in a head-tracking, headphone-based reproduction of piano sound. The resulting gain functions fit well empirical data observed at a piano performer's ears for 1496 recorded notes. System users experience a satisfying externalized image of the piano sound that is stable both in spatial position and in timbre.

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COMPETITION

Frame Your Physics

The Australian Institute of Physics is running a competition to promote the communication of physics to a general audience. Physics is an important science that helps us learn about the world around us, and if we are lucky, allows us to build useful technologies.

The aim of the competition is to present physics in a new light by making a video that presents a topic in physics to a general audience in an entertaining and informative way in under 3 minutes. The top videos will get an invite to a special presentation night at Questacon and will go in the draw for a share of $3000!

This competition is open to everyone. Entries must be submitted by 15th October 2012. So get filming, be creative and have some fun with it! And if you do submit an acoustics related video please let the AAS General Secretary know so we can use it for promotion.

Read all about it at:
www.act.aip.org.au/Frame_Your_Physics/Home_-_Frame_Your_Physics.html
[or just do a google search on Frame your Physics]

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MATERIALS AND MUSICAL INSTRUMENTS

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Many of the musical instruments with which we are familiar today have derived their basic structure from “natural” objects, and their detail has depended upon the materials from which they can be constructed. This brief exploration of the musical instrument scene shows how this both underlies and perhaps limits their future development.

INTRODUCTION

The evolution of musical instruments over the ages has depended in large manner upon the materials available from which they could be made. It is interesting, therefore, to examine the dependence of the behaviour of musical instruments upon these materials and to see in what ways they restrict or enhance the instrument performance.

PERCUSSION INSTRUMENTS

Music probably began in the humanities as a form of song, imitating to some extent the songs of other animals but with the addition of meaningful words so that the song told a story. The oldest musical instruments were probably made to enhance these songs by the addition of rhythmic sounds and began the evolution of percussion instruments. The first of these were simple sticks of wood clapped together, like Australian Aboriginal clap-sticks, but other cultures had hollow logs that could be beaten with heavier sticks, and these instruments were gradually evolved to more sophisticated forms. The peak of the evolution of these wooden percussion instruments was probably the xylophone and its partners, in which wooden planks of graded length were carefully supported so that they produced some form of musical scale when struck by a wooden hammer. These planks were further refined by thinning the central section so that several modes were in nearly harmonic relation, thus giving a more pronounced musical pitch.

These developments depended upon many of the intrinsic properties of wood, particularly that it was fairly consistent in elastic properties from one piece to another of the same species and quite low in internal damping so that the sound would not decay too quickly. It also needed to be easily shaped by the available woodworking instruments and long lasting enough for the efforts to be worthwhile. Since these early times, wood has been used in a large variety of other musical instruments, as will be discussed later, but it has always stood out as a readily available, simply workable, and long-lasting mechanical material.

Later in the evolution of humanity came metals, and these were of greater variety. Setting aside the “precious metals” gold and silver, those that were readily available from mining operations were copper, tin, lead and zinc, with iron entering the scene much later. Pure metals are too soft for most uses, so alloys were developed for use in making domestic or military objects. The most useful of these alloys were pewter (typically 85% tin and 5% lead), brass (typically 60% copper and 40% zinc) and bronze (typically 88% copper and 12% tin). Pewter had a low melting point and was rather soft, so that it was good for making domestic items, while brass and bronze were largely used by the military for armour and weapons. Brass was a “tough” alloy that would bend under stress, while bronze was more brittle and likely to crack, but both had melting points low enough (900–950°C) to make them useful casting alloys. When iron later became common it was widely used, but had a higher melting point which made manufacture more difficult.

Brass and bronze were the obvious candidates for making various percussion instruments since they could be melted and cast into shape in a fairly simple way. Bronze was the preferred material for large heavy objects such as bells, since it was very dense and had low internal loss, giving a sustained sound. Many bell shapes were developed, with different traditions in different countries. We are most familiar with the West European church bell, the shape of which is designed to produce a mode sequence close to 0.5, 1.0, 1.2, 1.5, 2.0,... with the 1.0 mode being the nominal pitch. One significant thing is the mode 1.2, which is a minor third above this nominal and gives the bells their particular sound. Bells from other cultures have different shapes and different sounds, but bronze casting is straightforward and gives sustained sound, and bronze bells last for thousands of years. Without bronze we would not have the bells we know today.

For percussion instruments with rather thin walls, such as gongs and cymbals, special bronze alloys are required, since brass will often deform under the impact and normal bronze might break. The sound of these instruments is very much different from that of bells, largely because of the fact that their vibration amplitudes are comparable with or larger than their material thickness, so that nonlinearity leading to harmonic production and even chaotic oscillation is common, giving impressive sounds to highlight important events in musical performance.

The final form of percussion instrument to be mentioned is the drum, which originally consisted of a piece of animal skin stretched tightly over some sort of retaining edge. Apart from the development of tuned instruments such as the tympani and the use of elastic polymer sheets instead of animal skin, very little has changed over the centuries.

STRING INSTRUMENTS

Much more important for modern music than are the percussion instruments are the string instruments, which include plucked stings such as the guitar, bowed strings such as the violin, and hammered strings such as the piano. In all cases the string itself is unable to radiate appreciable sound intensity because its diameter is so small compared with the sound wavelength involved, so that it is necessary for the vibrational energy of the string to be transferred to some much larger structure that can radiate efficiently. In nearly all cases this radiating structure is made of wood.

Let us begin with the violin family, the most important of all string instruments in a modern orchestra. The strings are traditionally made from animal gut, which had a considerable influence on tone quality since it had large internal losses at high frequencies. Modern violins mostly use synthetic polymers, which have less loss at high frequencies and so produce a brighter sound. The higher strings are even made of metal, which further reduces the high-frequency losses.

The violin body is made from wooden plates with strong edge support, and the most important part is the slightly domed top-plate to which string vibrations are conveyed through a bridge with one of its posts supported by a peg running through to the back plate, this converting the sideways forces produced by the bow-excited vibrating strings into vertical forces that excite plate vibrations. A very important thing about wood is that it is elastically very anisotropic, with the bending modulus being nearly ten times as large along the grain as it is in the transverse direction. This means that the lowest vibration mode should ideally have a shape about three times as long in the grain direction as in the transverse direction, and this is approximately the shape of a traditional violin. This lowest mode can therefore be efficiently excited and gives a strong fundamental sound to the instrument. Of course things are much more complex than this, for the back plate is also excited into vibration, and the enclosed volume, vented through the “f-holes” contributes another important resonance. The shape of the violin is also, of course, influenced by the need to allow access to all the strings by the bow, which gives the overall “figure-eight” form.

The wood used for the violin has been the subject of extensive study, particularly in relation to the excellent violins produced by Stradivari and Guarneri in Cremona, in the early eighteenth century. Was there something special about the wood used – grown in the “Little Ice Age” of the seventeenth century? The answer is not yet clear, but modern violins can now be made that surpass the perceived quality of these “old masterpieces” as judged in “blind” playing and listening tests. Modern experiments with different kinds of wood show that this does have a pronounced influence on tonal quality, as is indeed to be expected from the variations in elastic anisotropy and vibration losses. Much study is still in progress since modern makers want to produce the best possible instruments.

All these principles apply in equal measure to the viola, cello and double-bass, the larger instruments of the standard string group. In a famous development, American violin maker Carleen Hutchins and Harvard physicist Frederick Saunders expanded the scope of the violin family to a total of eight members using the acoustical principles underlying the classical Italian violins, and this “New Violin Octet” covers a range from that of the standard double bass to one octave above the violin. Quite a number of these octets now exist around the world.

Since elastic anisotropy is important in reproducing the quality of classical violins, this rules out many materials for their construction – one could make a violin body out of thin metal sheet, for example, but it would sound very different! The most likely contender is fibre-reinforced plastic composite material, since the elastic properties, and particularly the anisotropy, can be adjusted in the design. Quite good violins have been made using such composites for the body, but they do not possess the beautiful appearance of high-quality wood.

Very much the same principles apply to the materials from which the bodies of guitars are made, and for very much the same reasons. One major difference is that the top-plate of a violin is curved to support the stress of the string tension, and this raises its vibrational mode frequencies, while a guitar top plate is flat and stiffened by a set of carefully arranged braces. There are thus somewhat different criteria to be used in choosing appropriate materials.

Taking one step further, instruments such as the harpsichord and piano also rely upon a wooden soundboard to translate the vibrations of the strings into radiated sound. Many more compromises are needed here, however, because of the large size of the soundboard and the fact that the great tension produced by the large number of metal strings is supported by a frame made either of wooden or steel beams. The soundboard is stiffened by a pattern of braces but there is still an influence of the material properties upon its vibration and consequently upon the sound produced.

WOODWIND AND BRASS INSTRUMENTS

Wind instruments became popular long ago because of the sustained loud sounds they could produce. Their introduction into music appears to have been achieved because of the natural occurrence of structures that could make loud tonal sounds when blown. An early example is the conch shell, which has the structure of a conical tube wound in an elongated spiral. Blowing through a hole made near the narrow end of this spiral could produce a loud trumpet-like sound, and this is still widely used in ceremonies in some Buddhist temples in Asia, only one or at most two different pitches being produced. In this case it is the structure rather than the material that is important and the derived musical instruments generally used metal when evolving into their modern form.

A related but very different case evolved in Australia, where termites hollowed out the centre material of the trunks of small Eucalypt trees. The hollow trees could be detected by tapping on the trunk and then cut down to produce tubes typically about 150 cm in length and with slightly flaring internal diameter, typically about 4 cm at the narrow end. Figure 1 shows a picture of such a tube called a didjeridu.

Figure 1. A didjeridu is crafted by termites that eat out the centre of the trunk of a small Eucalypt tree to produce a slightly flaring tube

The third, and perhaps most important, shaped material giving rise to wind instruments was bamboo, which grew smooth uniform tubes of various lengths and diameters with blocking partitions at intervals along their length. Once again the material here was not of great importance, but the existence of such varieties of tube lengths and diameters led to the development of instruments with a cluster of pipes of graded length which could be used to play tunes, as in the panpipe as shown in Figure 2, or single pipes with finger holes as in the Japanese shakuhachi or the middle-European flute.

Figure 2. A set of panpipes made from bamboo. These pipes are sealed at the lower end by a natural partition in the bamboo

Materials come into importance when these traditional instruments were formalized for modern use. Lip-blown instruments derived from the conch shell were mostly made using brass, because this material was readily available and, because of its non-brittle nature, it could be mechanically worked into flaring tube structures. These techniques have persisted until the present day with only minor variations to improve appearance and stop corrosion. The sound of a brass instrument is determined largely by its shape and that of the mouthpiece, but there is some minor influence from vibration of the thin walls of the flaring horn, determined more by shape and thickness than by material properties.

Wood is also widely used for the class of "woodwind" instruments, particularly the flute, oboe, clarinet and bassoon. The instrument is machined from the wood and desirable material properties here relate mainly to the smoothness of the wood surface inside the tube, for this influences acoustic losses. Durability and appearance are also important and hardwoods from rainforest environments such as ebony are particularly favoured. One other feature of some of these instruments is the fact that they are excited by vibration of a reed valve held between the lips, and the material properties of the cane used to produce this reed are of great importance. Reed-making is essentially a hand-crafting process and, while working reeds can also be made out of plastic materials, they are generally considered to be of low musical quality.

A relatively recent development in woodwind instruments was the mid-nineteenth century development of flutes made from silver alloy tubing by Theobald Boehm (Figure 3). His major contributions were actually a mechanism involving coupled soft-padded keys that was later transferred to other woodwind instruments, and the conversion of the tapered wooden flute tube to a cylindrical metal tube with a tapered head-joint. Because the flute tube is cylindrical there is little opportunity for wall vibrations to influence sound quality so flutes are typically made of silver-plated copper-nickel alloy or of silver with 5 to 10% of added copper to harden it. Silver flutes are generally superior to those made from copper-nickel alloy because they are made and adjusted by hand, not on an assembly line, but this has little to do with the materials involved. Despite this, top-quality flutes are often made of gold and perform rather better than silver flutes, not because of the material but because they are made and finished by the best maker in a top company. This progression has even been carried further to platinum, with a dedicated musical composition by Edgar Varese entitled "Density 21.5".

Figure 3. A classical wooden flute with simple finger holes and a modern silver flute with complex Boehm finger keys (not to quite the same scale)

The other "woodwind" instrument that uses metal instead of wood is the saxophone, developed by Rudolf Sax in the nineteenth century using the coupled-key system developed by Boehm. The instrument is made from a metal alloy, typically brass or bronze, and produces a loud mellow sound, but this is due to the widely tapering enclosed air column rather than to the material. Because of the shape of the saxophone, it would be difficult to make it from anything except metal, or perhaps plastic.

PIPE ORGANS

Pipe organs have been on the musical scene since the time of the Romans, two thousand years ago, and can be regarded as derived as a mechanical version of the panpipes. It is by far the largest present-day musical instrument, with perhaps the exception of the carillon. The Sydney Opera House organ, (a small fraction of which is shown in figure 4), contains nearly 10,000 pipes which are controlled by five keyboards and a separate pedalboard, while the Sydney Town Hall organ has a "square conical" longest pipe of 64 feet (20 metres) which produces a fundamental of 8Hz from the bottom key of the pedalboard. The sounding of the pipes in an organ is controlled by mechanical, pneumatic or electric links between the keyboards and the valves that supply air to the pipes, and varies from one organ to another. In the present context, however, the matter of interest is the materials from which the organ pipes are made.

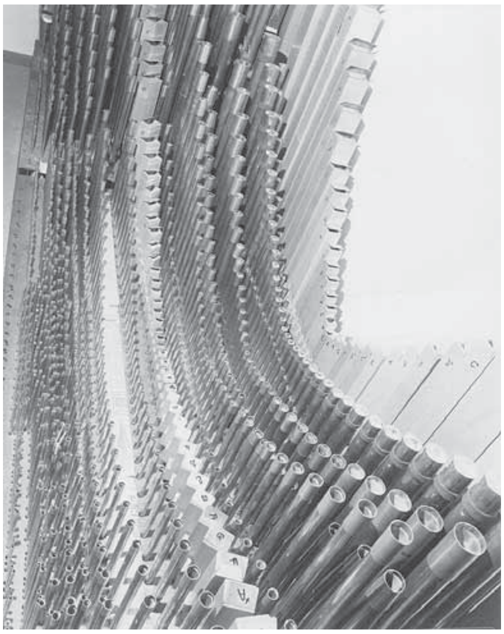

Figure 4. A small fraction of the ten thousand pipes making up the Sydney Opera House organ

From the discussion above, it is clear that, provided the material used has a smooth surface, it has little effect upon the sound of a cylindrical pipe. Some of the pipes of an organ, however, are square rather than circular in cross section and are made of wood. The walls of these pipes can be made to vibrate under the influence of internal acoustic pressures and so may have an effect upon tone quality when the playing frequency or one of its harmonics is a near match for a vibration frequency of the wall panels. The walls of these wooden pipes are generally sufficiently thick that this is not a significant matter, but the effect can be detected by acoustic measurements.

In the case of the cylindrical or conical pipes that constitute most of the organ, the material has little effect upon sound quality, though this might not be true if the walls were very lightly damped because the cylindrical symmetry is broken at the pipe mouth where there is an aperture across about half of the pipe diameter. Such resonances would be regarded as intrusive, so there is no desire except to ensure that they do not arise. The main problem with organ building, rather than design, is therefore to ensure that thousands of pipes of different sizes and shapes can be made as simply as possible and with opportunity left for detailed adjustment after the organ is assembled.

Many different metals could be used to make the pipes, but the alloy of choice is a tin-rich tin-lead alloy similar to pewter. It has the advantages of a low melting point, 170 to 230°C, and is both strong enough to stand upright for hundreds of years without distortion and soft enough that the pipe mouth can be easily adjusted by the organ builder using simple hand tools. To make the pipes, a rectangular box of liquid metal, with one of the sides having a gap of about 1mm along its bottom edge, is slid along a flat table of marble or other similar material and the result is a uniform layer of the tin alloy that solidifies in less than a minute. This layer can then be lifted up and cut into pieces that are wrapped around wooden rods and soldered closed to constitute organ pipes. These cylindrical sections are then soldered onto the smaller structures constituting the pipe mouth and supporting section. These techniques can be used over dimensions ranging from millimeters to meters.

Very few difficulties have been experienced with these methods or the resulting pipes, one of the few being interestingly known as "tin pest" in which the shiny tin pipes develop white powdery surfaces and may even corrode away. Interestingly this is not a true case of corrosion but rather of phase change in a tin-rich alloy at the sub-freezing temperatures that may be experienced in Scandinavian churches!

Since the metal from which the pipes are made has little effect upon tone quality, it is possible for some organs to be designed with display pipes made from copper or some other material of different appearance, or for them to be painted with appropriate decorations, none of this having any significant effect upon tone quality. Figure 5 shows a variety of organ pipes all sounding the same note but with differing sound quality because of their differing shapes.

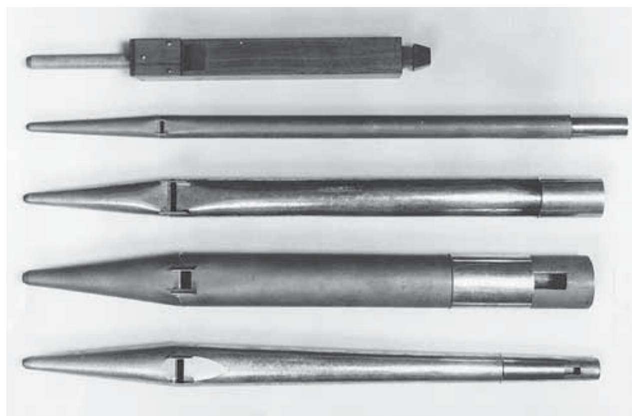

Figure 5. A variety of organ pipes all sound the same note. The pipe at the top is made from wood, the two shiny pipes are made from tin-rich tin-lead alloy and the two dull pipes are made from lead-rich alloy

CONCLUSIONS

The relationship between musical instruments and the materials from which they are constructed has a long history, and many instruments have evolved because of the prior existence of specialized materials or natural structures. Some aspects of modern musical culture have opted to divorce from this relationship and to produce "musical" sounds by purely electronic means that require no real instruments. While this certainly gives "musical freedom" to the composer, the absence of an identifiable instrument and performer detracts from the influence upon the listener. The next step, perhaps, is to bypass the electronic equipment and have simple "brain to brain" transmission of musical compositions through neural couplings. But perhaps not!

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.

THE SPEECH TRANSMISSION INDEX AFTER FOUR DECADES OF DEVELOPMENT[1]

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This year, the Speech Transmission Index (STI) celebrates its 40th birthday. It has been four decades since Houtgast and Steeneken first published their objective method for predicting speech intelligibility in Acustica [1]. Since then, the STI has evolved into a versatile and mature method, used in a diversity of applications. It is now more popular than ever, with record numbers of STI users as well as manufacturers of STI measuring solutions. We mark the occasion by looking back at the development of the Speech Transmission Index throughout the decades, while also presenting an overview of current developments and challenges.

ORIGINS OF THE SPEECH TRANSMISSION INDEX

What inspired Houtgast and Steeneken to develop the STI was their desire to save time and to eliminate the dull work associated with subjective intelligibility tests. Or, in the words of Houtgast: their "laziness". Their work back then, at TNO in the Netherlands, consisted largely of carrying out lengthy evaluations of speech intelligibility, mainly of military communication systems, using large numbers of human test subjects. The need was there for a faster, and more diagnostic, alternative to subjective listening tests. The primary design objective was that it should be a physical measuring method (ie. based purely on physical principles without humans in the measuring loop), which could produce results fast. Moreover, a measuring method was required that could use a test signal in order to obtain direct and immediate results. This sets the Speech Transmission Index apart from the Articulation Index (AI), which was already around at the time. The STI owes several of its key characteristics to the work done by French and Steinberg [2] on which the AI is also based. However, the AI (and later on its successor the Speech Intelligibility Index, SII) is basically calculated from measured sound pressure levels, theoretical data or measured impulse responses. Among other things, this means that the AI and SII are inherently "blind" to non-linear effects, whereas the STI incorporates these effects.

The Speech Transmission Index concept also incorporated insights crossed over from research in the visual domain in the early 1970s. Optical system engineers back then already used the concept of the Optical Transfer Function (more generally named the Modulation Transfer Function) to quantify the transmission quality of optical systems. Houtgast and Steeneken realized that similar principles in the time domain should apply to transmission of speech signals.

[1] Originally published in the IOA Acoustics Bulletin, May/June 2012 and reprinted with permission.

KEY CONCEPT

Houtgast and Steeneken designed their STI test signals based on modulated, speech-shaped noise. The basic principle underlying the STI is that preservation of speech intelligibility during transmission is achieved by preservation of the natural intensity fluctuations in speech spectra. The design of test signals was such that they mimicked these natural modulations, but in such a way that measurements could be carried out quickly, precisely and within the constraints of calculation (computer) power of the time. After four decades of evolution, the basic principles remain unchanged – although the computer power is now available in handheld devices, whereas the necessary equipment originally required several people to lift.

INITIAL USE OF THE STI METHOD

In the 1970s, the STI was very much a niche method. The inventors themselves used the STI in various real-life applications, but use by others was limited to a few studies done out of scientific interest only. The publication of Steeneken and Houtgast's JASA paper in 1980 [3] marked the beginning of more widespread use of the method. The growing group of STI users forked into two separate (but overlapping) communities almost from the very beginning. On the one hand, there is a scientific community, attracted to the way the STI predicts speech intelligibility based on a near-universally applicable model with only few design parameters. On the other hand, there is the engineering community, interested mostly in the practical advantages that the STI was designed for: fast, objective and accurate predictions of speech intelligibility.

To the engineering community, standardization of the STI method by successive IEC-committees (in successive editions of IEC 60268-16 [4]) turned out to be of key importance. The version of the STI described by Steeneken and Houtgast [3] was standardized as the original, first edition of IEC 60268-16. TNO already had a variety of test signals available, but the

RASTI test signal (Room Acoustical STI), designed specifically for application of the STI in room acoustics) saw the most widespread use. This was largely due to the availability of RASTI measuring hardware from HYPERLINK "http://www.bksv.com/" Brüel & Kjær, based on TNO's earlier RASTI device (figure 1).

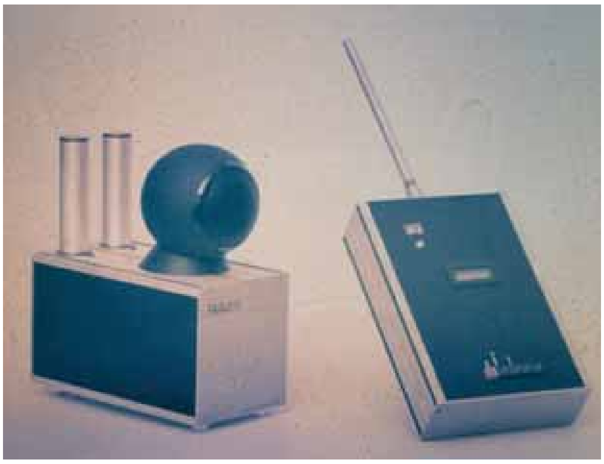

Figure 1. First handheld implementation of the STI (RASTI hardware, 1980)

Over the years, a lot of criticism towards the STI came from users having experiences with RASTI outside its intended scope of use. RASTI measurements are accurate measurements of the STI, if applied to pure room acoustics; ie. transmission chains featuring electro-acoustic components should never be measured using RASTI. Words to this effect in the RASTI manual have not stopped people from attempting to do so anyway – and even publishing criticizing accounts of how RASTI failed to yield accurate predictions.

IEC 60268-16 SECOND EDITION (1998)

There was also a certain amount of justified criticism towards the "original" STI, which triggered a significant amount of research at TNO in the 1980s en 1990s to improve on the method. Several major improvements were standardized in the second edition of IEC 60268-16, which was released in 1998. The original STI did not account for the fact that speech perception is aided by synergistic effects between adjacent frequency bands. Among several other improvements, additional model parameters were added to take these between-band interactions into account. The 2nd edition of the STI was named STI_r ('r' for revised), but the subscript was dropped later on. It is now customary to simply refer to any version as "STI," indicating which revision of the IEC standard applies in accompanying text (if relevant).

The STIDAS IID device produced by TNO was capable of measuring the STI according to first and second editions, using a host of different test signals, including full STI modulated noise test signals and STITEL (specifically for telecommunication measurements). This device was sold worldwide, but its specific hybrid analog-digital design made it too expensive for many users. Some of these units remain in service to date, mostly at military research facilities. A photo of the STIDAS STI device is shown in figure 2.

Figure 2. STIDAS I (STI Device using Artificial Signals) device based on a PDP-11/10 computer and custom analog hardware (1971)

A trend in the 1990s was that many acousticians started to use estimations of the STI based on measured impulse responses. Affordable PC-based software for impulse response measurements was becoming commonplace. If certain conditions are met (among which linearity, no back ground noise or band-pass limiting), then the STI may be precisely derived from the impulse response. This is what many users were doing (or rather, what their software was doing for them). Unfortunately, the conditions for this approach to work do not generally apply. In fact, much like RASTI, impulse response-based STI estimates can only be relied upon in evaluations concerned purely with room acoustics. A need was widely felt for a test signal (and a version of the STI method) that was applicable to electro-acoustics transmission chains, and could be measured quickly and directly. This led to the development of STIPA.

IEC 60268-16 THIRD EDITION (2003)

The third edition introduced two major changes. Most importantly, it introduced the STIPA test signal (sometimes referred to as STI-PA), which is a test signal optimized for PA systems. Compared to RASTI, STIPA has the advantage that all octave bands are covered (125 Hz – 8 kHz), although only two modulations frequencies are tested per octave band. This means that STIPA can be used reliably in nearly all cases involving electro-acoustics as well as room acoustics. STIPA can be used in any condition that RASTI was previously intended for, with the possible exception of rooms featuring pronounced, individual echoes. Since RASTI is inherently unsuitable for any condition involving electro-acoustics, the introduction of STIPA made RASTI completely obsolete.

The 3rd edition also introduced the concept of level-dependent masking. Earlier versions of the STI ignored the fact that auditory masking curves flatten out at higher sound levels, effectively reducing intelligibility. The resulting mismatch sometimes observed between the STI and subjective intelligibility at high sound levels no longer exists from the 3rd edition onwards. The price for this added accuracy is that measurements need to be calibrated in terms of the (A-weighted) sound pressure level. This was already common practice, but not specifically required before. If acoustic calibration is not feasible (e.g., when evaluation intelligibility of purely electronic devices that may be used at arbitrary speech levels), level dependent masking may be disabled. The resulting STI is then only valid for comfortable listening levels.

The design and release of STIPA had the intended effect. A photo of the first STIPA-capable device to reach the market is shown in figure 3. Measuring devices by several manufacturers reached the market, and the last users that had been holding on to their now-obsolete RASTI equipment made the transition. Although STIPA is just one of several standardized test signals in the 3rd edition, it turned out to be virtually the only one used in practice. Many users still using indirect (impulse-response based) measurements also decided to obtain STIPA-capable devices. Some (local) regulations specifically requiring STIPA helped to speed up this process. In practice, situations for which the STIPA test signal is insufficient, and "full STI" measurements are required, are rare; this is the case mainly when strong discrete, single echoes occur.

Figure 3. The first STIPA-capable device to reach the market, made by Gold Line (2002)

IEC 60268-16 FOURTH EDITION (2011)

Even if the STI method itself had some room left for future improvement in its third edition, it was mostly the text of the IEC standard itself that now became criticized. With more equipment manufacturers implementing STIPA, it became apparent that it was not easy to build a STIPA-capable device when using the standard as a single source of information. The standard was therefore completely overhauled and much information was added.

The fourth edition of the standard [4] outlines not only how to design direct STI measurement (using modulated test signals such as STIPA) but also how to implement indirect (impulse response-based) measurements. Limitations of different approaches and test signals are now clearly indicated in the standard. In other words, for different types of application, the standard now prescribes which methods may, and which ones may not be used safely.

The fourth edition features only a single (minor) change to the STI algorithms itself: the calculation of level-dependent masking was changed from a discrete lookup-table to a smooth continuous function. Also added is information on interpretation of the STI relative to true speech intelligibility. Whereas the STI quantifies the impact of the transmission channel on intelligibility, there is also an influence of talkers and listeners. There are fixed and well-known relations between STI and intelligibility for "normal" populations. The 4th edition of the standard also assists in interpreting the STI for populations of non-native talkers and listeners, as well as certain categories of listeners with hearing loss.

THE MAJOR CURRENT CHALLENGE: VALIDATION AND CERTIFICATION

Every successive update of the STI method was validated at TNO, using a reference system called COMCHA. This reference system simulated a wide variety of representative test channels (78 channels based on band-pass limiting and 68 channels for communication channels). TNO also maintained reference versions successive generations of measuring devices. Besides validation of new additions to the STI framework, these tools were also used to provide third-party validation and certification services, for instance for STIPA measuring devices from various manufacturers.

Today, validation services based on these assets are no longer be offered by TNO. In practice, there is no other institute or company capable and willing to take over this service that has the same level of confidence, expertise and (especially) independence. This is perhaps the major current challenge for the future of the STI: making sure that all STI devices measure consistently and correctly according to the standard and produce identical results. Likewise, all STIPA signals (and also other STI test signals), should be interchangeable and compatible with each IEC-compliant measuring device.

For the moment, the best solution appears to be to create an open-source validation database. TNO and Embedded are collaborating in creating such a reference database of degraded STIPA test signals using the original COMCHA conditions, verified with "golden standard" software from TNO. This set of signals will represent the various types of conditions for which STIPA is sensitive, such as noise, reverberation, peak clipping, etc. This database will be made available through the internet under an open licensing regime, such as (for instance) GPL. Not only will developers be able to test and validate their devices using these signals; their users (and competitors) will be able to check compliance using the very same database. In our view, this provides for a system of checks and balances that eliminates the need for an impartial certifying authority.

CURRENT AND FUTURE RESEARCH

The STI has been a tool in many scientific studies, but it is also itself the subject of scientific investigation. In the past, the focus was often to improve the method, in terms of solving known inaccuracies and issues with the method. Nearly all of these issues have been thoroughly investigated and are now closed chapters; examples are the interaction with gender, non-linear auditory masking and variations in the modulation spectrum. Right now, the focus of research is more on *extending* the scope of the method rather than just generally improving it.

One very interesting field of research is measuring the STI using real, recorded, speech instead of artificial test signals. This was actually considered from the very beginning; in the early years however, there was simply a lack of processing power for this to be practically feasible. First accounts of speech-based STI measurements were published in the 1980s. A difficulty with speech-based STI measurements is that useful, natural modulations are present (such as in the artificial test signals), but detrimental components, such as nonlinear distortion components, tend to have similar modulation spectra. Alternative approaches were proposed, among others, by Drullman [5] and Payton [6], but their approaches were only partially successful in separating between useful and detrimental modulations. The concept of weighing modulations frequencies within an MTF based on the question whether or not phase shifts occur was explored by van Gils and van Wijngaarden [7], and proven promising. Speech-based STI measurements were, among other applications, shown useful to evaluate digital voice coders. An open question at the moment is to decide on optimal phase weighting functions. Also, further validation in a wider range of realistic conditions is needed.

Another field of research is the study of binaural STI measuring methods. The STI has always been a monaural model. This means that the STI cannot be used to distinguish between conditions in which binaural listening benefits are significant. Specific model additions have been proposed by van Wijngaarden and Drullman [8] to incorporate binaural listening. Similar work has been done by Beutelmann et al. [9] in the context of the Speech Intelligibility Index (the successor to the Articulation Index). This work needs to be consolidated into a robust addition to the STI model, that may optionally be used to refine STI-based studies in which binaural listening plays a predominant role. Such an addition also needs to be validated.

MEASURING THE "FULL" STI WITH MODULATED NOISE CARRIERS

Another relevant current research topic is concerned with improving and extending the current array of test signals. At the moment, the STIPA test signal is used nearly exclusively. This means that only two modulation frequencies per octave band are tested. A "full" STI measurement involves modulation frequencies sampled in 1/3 octave bands from 0.63 Hz to 12.5 Hz. In practice, a sparsely sampled MTF matrix (such as the one offered by STIPA) suffices for most applications – but not all. As mentioned above, care should be taken when using STIPA in rooms with discrete echoes. All current commercially available methods for measuring the full MTF matrix make use of inverse calculation of the MTF based on impulse response measurements. This is not permitted if nonlinear distortion components may occur. Only the TNO reference system currently features a fully IEC-compliant measurement mode for full STI measurements. The drawback of the TNO system is that it is based on obsolete hardware, takes up to 10 minutes for a single measurement point, and requires the test signal generator and the STI analyzer to be synchronized.

Embedded Acoustics has initiated a research project that is intended to result in an advanced full STI measuring scheme, based on modulated noise carriers, that does not need to be synchronized. In practice, a measurement will appear to be similar to a STIPA measurement, except for the measurement time (which will probably need to be 1 to 2 minutes).

ON TO THE NEXT FOUR DECADES…

When the 4th edition of IEC-60268-16 was published last year, hardware and software vendors proved quick to update their products. This is encouraging; it shows that the market is quick to respond to changes. Several companies will launch new STI products in 2012, from STIPA modules for existing hardware to completely new devices and mobile apps (figure 4). Also, the STI is finding its way into new standards and regulations every year, replacing now-obsolete subjective intelligibility tests and less advanced metrics. This ranges from the national NEN-2575 standard for certification of Voice Evacuation systems in the Netherlands, to the NFPA-1981 standard in the US for testing speech intelligibility of face masks.

In conclusion, there is a community willing and able to support the STI, and the number of users is also consistently growing. Keeping the method up to date for another forty years will be an effort that requires this community of individuals and companies to actively cooperate. We predict that in the next few years we will see this community pulling together, and starting to prepare work for the 5th update of the IEC standard, somewhere around 2016.

Figure 4. iPhone apparatus for performing 4th edition-compliant STIPA measurements (2011)

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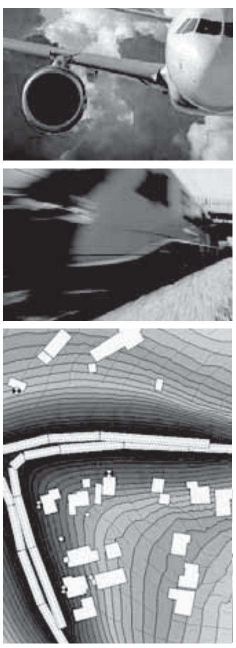

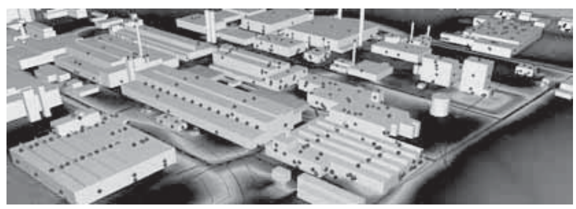

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WIND FARM NOISE – AN ETHICAL DILEMMA FOR THE AUSTRALIAN ACOUSTICAL SOCIETY?

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Not since the opening of the Third Runway at Sydney Airport has there been so much publicity in Australia concerning noise – in this case wind farms. Putting aside the issue of noise versus inaudible noise there is a question being raised as to Members of the Society breaching the Code of Ethics. This is not the old question of Professional versus Learned Society. Reliance upon criteria contained in Guidelines or Standards may be an excuse by consultants that in turn places the "fault" on the SA EPA and the New Zealand Standard. However, if people making complaints to no avail and leave their homes because of the wind farm "noise" what is the responsibility of Members of the AAS to the community?

INTRODUCTION

The April 2012 edition of the Australian Acoustical Society's journal (Acoustics Australia – Vol 40, No. 1) provided a series of papers and technical notes relating to wind farm noise [1]. However, the articles supporting wind farms did not discuss the acoustic impact of the wind farms. The articles referred to criteria and compliance with the criteria. The articles did not identify the basis of the criteria or the acoustic impact of wind farms even when they complied with the nominated criteria.

It is evident from the recent public forums conducted by Senators Madigan and Xenophon in South Australia, Victoria and New South Wales that wind farm "noise" is an issue in the community [2,3]. The degree of claims for and against wind farm noise is reminiscent of the aircraft noise debate (with the introduction of jet aircraft to Australia) [4] and the third runway at Sydney Airport [5].

Examination of the aircraft noise debate finds acoustic and socio-acoustic research undertaken in Australia by Members of the Society. Examination of the wind farm noise issue finds a different position.

Members of the Society had been at the forefront of preparing acoustic and vibration Guidelines and Standards in Australia [6] to protect the community from a wide range of noise sources and invariably rely upon overseas experience/standards that are then compared or evaluated with Australian situations.

For example with respect to road traffic noise, we had Standards/Guidelines that originally followed the UK Department of Environment [7] recommendations (rather than US Department of Transport criteria). Work undertaken by the ARRB and Dr Stephen Samuels (and others) lead to a modification of the British criteria to account for Australian road conditions.

AIRCRAFT NOISE IMPACTS IN AUSTRALIA

In the initial stages for aircraft noise assessment Australia adopted the US NEF system [8]. As a result of community concerns about aircraft noise, and a Commonwealth government inquiry (HORSCAN report) [4] led to the noise study by the National Acoustics Laboratory [9] to then result in the ANEF system used for aircraft noise assessments in Australia. Changes have been proposed to the aircraft noise standard, citing the community's response to aircraft noise and the need for supplementary acoustic metrics. However the use of the N60, N70 or N80 descriptor [10] has not been presented in terms of any socio-acoustic surveys and therefore there is a fundamental problem of implementing N60/N80 criteria without any basis to support that criteria.

In the original NAL report on aircraft noise there is the dose response curve for ANEF versus affected people which is slightly different to the curve in Australian Standard AS 2021 [11]. Contained in the NAL report is a dose response for the N70 that can be placed in the context of the unacceptable/acceptable limits for the ANEF system and in turn the building site acceptability tables in AS 2021.

The NAL report does not provide any regression curves showing a basis for N60 or N80. Therefore, as presented previously [12-15], there are issues as to substantiating the number of events that may be applied to the N60 and N80 for an acceptable aircraft noise impact.

In undertaking research work with Fergus Fricke at Sydney University [16] most postgraduate students became aware that Fergus pulled/pushed you sideways to look into different aspects of your subject which required further investigation and a broadening of the material that was the subject of the research. It is such an approach that students of acoustics (of which all members of the Society can still said to be students) can benefit in their daily use of acoustics to have in the back of their mind when there is a problem the quote of Professor Julius Summer Miller "Why is it so?".

This is the exact situation when faced with the challenge of measurements from helicopter operations not agreeing with the international computer modelling led to investigating the matter of lateral attenuation. Investigation found that the attenuation algorithms in the computer model [8] were wrong, had been wrong for many years, and people were unaware of that fact. Investigations, including going back to the original reference documents [17,18] to uncover the problem, which was verified with additional testing leading to that material being presented to the US Aircraft Standards Committee in 2003 [19], accepted and two years later INM was amended to overcome that issue.

Similarly in seeking to validate military aircraft operations with the computer model we kept on getting incorrect results for high frequency noise which under the same investigative concept lead to querying the results. Testing over a number of years led to identification that the original model for determining atmospheric attenuation coefficient per hundred metres was not carried out in any vast chamber or airfields, ovals or similar. The attenuation coefficients were determined from a stainless steel sphere of 1.68 m diameter on a theoretical basis [20].

Utilising measurement data for aircraft operations under different atmospheric conditions found the universal attenuation coefficients [8,21] did not agree with field measurement for aircraft [22] and monitoring at industrial sites.

These results revealed that if one utilises the atmospheric attenuation contained in various International and American standards in computer models there can be errors. And in particular there can be significant errors if one is dealing with high frequency noise, particularly with respect to the discharge of high velocity steam where there is a significant component of the noise source occurring above 2000 Hz.

It is in light of the above background material and the fact that throughout Australia there are hundreds of residents in proximity to wind farms who claim to be adversely affected, and in some cases so affected that they leave their properties, that must be of concern to members of the Society where there are repeated responses that these people are imagining the problem.

It would appear that the reaction of the community to wind farms is not that dissimilar to communities that were subject to the aircraft noise following the introduction of the jet engine that ultimately led to the famous NAL study. The number of people affected by wind farms is not as great as that affected by airports simply because wind farms are not located in suburban areas. However, in taking into account the percentage of people affected in the area covered by the nominated noise level criteria it would seem to be more than 10% of the population are seriously affected.

MEASUREMENT OF WIND FARM NOISE FOR THE COMMUNITY

I and a number of acousticians in Australia have been requested to undertake reviews of wind farm applications and/or conduct measurements of wind farms. This is not dissimilar to requests for peer reviews of acoustic reports for Development Applications or Compliance Tests for a range of typical noise sources, domestic, road, rail, air traffic, and industrial developments.

These reviews and testing have raised a number of issues as to the adequacy of the original assessments, the accuracy of the measurements and question the acceptability of noise limits which are simply matters that an appropriately qualified and experienced acoustic engineer/consultant would undertake.

Such investigations and assessments have raised concerns as to the adequacy of the guidelines and also the results of compliance testing undertaken by various organisations that include Members of the Australian Acoustical Society.

As a result of undertaking the assessments and providing those reports in the public domain I and other consultants have been labelled by wind farm power entities as being “anti-wind farm” or having close ties to “anti-wind farm lobby groups”.

Having discussed this very fact with other Members of the Society who have been so labelled and do not accept such accusations, I have stated a number of times that I am not anti-wind farm but have been simply presenting the facts as to what has been generated by such installations that requires further investigation.

If one is to be labelled as anti-wind farm when simply presenting the facts of what is occurring as a result of undertaking work for the community, then it must be the case that the acoustic consultant/engineer who undertakes work for wind farm applicants should equally be labelled by the wind farm industry as “pro-wind farm”.

Both the “anti-wind farm” and “pro-wind farm” acousticians who are Members of the Society would undoubtedly disagree with such labelling and should identify the fact that they are truly independent in carrying out such assessments. Furthermore, if those persons are Members of the Society then they could bring to their defence that there is an obligation to abide by the Code of Ethics of the Australian Acoustical Society [23].

So how can persons undertaking assessments “for or against” wind farms of the noise impact of wind farms be a dilemma for the Australian Acoustical Society you may ask.

CODE OF ETHICS

From the Code of Ethics, that appears on the Society’s website, one can see there is the Responsibility for the members of the Society:

The welfare, health and safety of the community shall at all times take precedence over sectional, professional and private interests.

The explanatory notes in the Code of Ethics in referring to Responsibility requires members of the Society to:

- conform to acceptable professional standard and procedures, and not act in any manner that may knowingly jeopardise the public welfare, health, or safety.
- endeavour to promote the well-being of the community, and, if over-ruled in their judgement on this, inform their clients or employers of the possible consequences.
- contribute to public discussion on matters within their competence when by so doing the well-being of the community can be advanced.

The explanatory notes in the Code of Ethics in referring to Work within Areas of Competence requires members of the Society to:

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

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

NOISE IMPACT

A significant number of wind farm assessments follow a generic format. Whether there is identification of primarily the South Australian EPA Wind Farm Guidelines [24,25] or the New Zealand Wind Farm Standard [26,27], the assessment in terms of those guidelines uses the ambient noise level to provide regression line curves, use of a criterion of 35, or 40 dBA and background +5 dB, whichever is the greater value.

The acoustic assessment generally provides the results of computer predictions using the A-weighted value to then indicate compliance with the criteria contained in Guidelines/ Standard.

The noise assessment in relation to the application provides predicted levels in terms of the substation and construction activities that are related to relevant guidelines, and may include an assessment of noise from power lines to indicate significant separation distance to residence to not present at an issue. In some cases there is identification of the acoustic impact of the substation, construction activities, and power lines [28-31].

However in the generic wind farm assessments there is no actual noise assessment of the wind farm, i.e. the assessment simply states compliance with the relevant guidelines and that is it.

The generic wind farm "noise assessment" considers the noise outside residences and does not identify to the community the audibility of the wind farm, the relationship of the guideline criteria with respect to the acoustic environment of the area, the percentage of time in which there will be audible noise as a result of weather conditions, or conversely a reduction in noise as a result of weather conditions.

The generic wind farm "noise assessment" does not report the situation of residents hearing the noise inside their homes or having sleep being disturbed or that some residents experience disturbance even when there is compliance with the guidelines noise limit. The "noise assessment" does not indicate situations in Australia where residents (host and non-hosts) leave their homes to live elsewhere.

The question is now being asked in the community, and invariably will be asked in courts of law, whether the absence of that material in the "noise assessment" is a Breach of Code of Ethics.

The Association of Australian Acoustical Consultants (AAAC), of which firms become members of that Association, have a Code of Professional Conduct [32] which goes one step further than the AAS in the section on Professional Standards:

- To maintain the standards of business and personal conduct reasonably expected of a professional
- To act with professional responsibility and integrity in my dealings with the community and clients, employers, employees and students
- To provide professional opinions in an objective and truthful manner, avoiding statements that may be demeaning, misleading or unethical
- Not to misrepresent one's skills and experience
- To undertake work only in areas of competence, unless the client is informed of the member's limitations
- To maintain a proper sense of responsibility to the client, broader community, employees, the profession and the environment.

In attending various rural dwellings to undertake wind farm noise measurements questions have been raised by the occupants as to the conduct of members of the AAAC and the AAS in relation to monitoring and reporting of the results/impact.

RURAL NOISE ENVIRONMENTS

Acousticians in Australia that are aware of the origins of Australian Standard AS 1055 [33,34] will be well aware that it follows that the general scenario outlined for other standards and its primary function as per its original title was "Noise Assessment in Residential Areas".

Accordingly AS 1055 is not really a document that is appropriate for rural areas and the background levels that were suggested for various categories may be appropriate in suburban areas. However for areas removed from traffic the lowest background level in AS 1055 would not necessarily apply in such areas.

Rural areas removed from main roads and the like, and being areas nominated for wind farm developments can experience background levels less than 20 dBA in the day and night, and can also experience ambient L_{eq} levels less than 30 dBA during the day and less than 25 dBA at night.

A fundamental question that communities exposed to wind farms raise is how can the guidelines substantiate 35, or 40 dBA as an acceptable base level at night in rural areas?

The SA EPA Guidelines refer to an indoor sleep disturbance level of 30 dBA by reference to a WHO Guideline [35]. However there is a failure to correctly identify that the WHO guidelines were referring to suburban areas impacted by traffic noise and did not provide criteria for rural areas or consider wind farm noise. The draft New South Wales Wind Farm Guidelines [36] specifically clarified the WHO guideline sleep arousal related to noise in suburban areas from traffic [37].

The situation of background levels in residential bedrooms which are between 10 dBA and 20 dBA, even with turbines operating, must be a fundamental issue of concern for the Members of the Society for a guideline that suggests 40 dBA is an acceptable level at night (as an external level) or 30 dBA as an internal level.

If the "pro-wind farm" acoustician's defence to inadequate reporting assessment or consideration of the community's health relies upon Guidelines or Standards that have been issued for wind farms, then apparently blame may be to the authors of the Guidelines or the Standards committees which include Members of the Society.

It could well be argued that when the first version of the guidelines were prepared by the South Australian EPA they did not have the benefit of an existing wind farm to undertake measurements and determine the appropriateness of the draft guideline and then the guideline.

It would appear historically that the original SA EPA guidelines were based upon overseas material in part. However, there does not appear to be any reference in the document to identify where the base criteria have been substantiated for use in Australian rural communities, i.e. socio-acoustic study to support the limits.

OUTCOMES

The current public debate as to noise impact from wind farms would appear to be more complex than just the "Learned Society of Professional Institution" question raised by Fergus Fricke [38] in the same 1982 AAS Bulletin that reported on the NAL 1982 Aircraft Noise Report.

If further work finds there is a health issue as a result of "noise" generated by wind farms and there are "acoustic assessments" that state there are no health impact no sleep impacts, and no infrasound, then what happens?

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AAS CODE OF ETHICS

1. Responsibility
The welfare, health and safety of the community shall at all times take precedence over sectional, professional and private interests.

2. Advance the Objects of the Society
Members shall act in such a way as to promote the objects of the Society.

3. Work within Areas of Competence
Members shall perform work only in their areas of competence.

4. Application of Knowledge
Members shall apply their skill and knowledge in the interest of their employer or client, for whom they shall act in professional matters as faithful agents or trustees.

5. Reputation
Members shall develop their professional reputation on merit and shall act at all times in a fair and honest manner.

6. Professional Development
Members shall continue their professional development throughout their careers and shall assist and encourage others to do so.

EXPLANATORY NOTES

1. Responsibility
In fulfilment of this requirement members of the Society shall:

- Avoid assignments that may create conflict between the interests of their clients, employers, or employees and the public interest.
- Conform to acceptable professional standard and procedures, and not act in any manner that may knowingly jeopardise the public welfare, health, or safety.
- Endeavour to promote the well-being of the community, and, if over-ruled in their judgement on this, inform their clients or employers of the possible consequences.
- Contribute to public discussion on matters within their competence when by so doing the well-being of the community can be advanced.

2. Advance the Objects of the Society
Appropriate objects of the Society as listed in the Memorandum of Association are:

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

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

Object (g)
To encourage research and the publication of new developments relating to acoustics.

3. Work within Areas of Competence
In all circumstances members shall:

- Inform their employers or clients if any assignment requires qualifications and/or experience outside their fields of competence, and where possible make appropriate recommendations in regard to the need for further advice.
- Report, make statements, give evidence or advice in an objective and truthful manner and only on the basis of adequate knowledge.
- Reveal the existence of any interest, pecuniary or otherwise, that could be taken to affect their judgement in technical matters.

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

- Strive to avoid all known or potential conflicts of interest, and keep employers or clients fully informed on all matters, financial or technical, that could lead to such conflicts.
- Refuse compensation, financial or otherwise, from more than one party for services on the same project, unless the circumstances are fully disclosed and agreed to by all interested parties.
- Neither solicit nor accept financial or other valuable considerations from material or equipment suppliers in return for specification or recommendation of their products, or from contractors or other parties dealing with their employer or client.

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

- Pay nor offer inducements, either directly or indirectly, to secure employment or engagement.
- Falsify or misrepresent their qualifications, or experience, or prior responsibilities nor maliciously or carelessly do anything to injure the reputation, prospects, or business of others.
- Use the advantages of privileged positions to compete unfairly.
- Fail to give proper credit for work of others to whom credit is due nor to acknowledge the contribution of others.

6. Professional Development
Members shall:

- Strive to extend their knowledge and skills in order to achieve continuous improvement in the science and practice of acoustics.
- Actively assist and encourage those under their direction or with whom they are associated to advance their knowledge and skills.

Technical Note

EFFECT OF A 35 dB(A) MINIMUM CRITERION ON A WIND FARM DEVELOPMENT

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INTRODUCTION

Environmental noise criteria for wind farms in Australia are normally determined individually for nearby receiver locations. The criteria take the form of a minimum criterion or the background $L_{A90,10min}$ noise level plus 5 dB(A), whichever is the greater, for each integer wind speed between turbine cut-in and rated power.

At low wind speeds, the minimum criterion typically applies due to the lower background noise levels than during periods of higher wind speeds. A minimum criterion of 40 dB(A) is specified in the following standards and guidelines that are typically applied in Australia:

- New Zealand Standard 6808:1998 *Acoustics – The assessment and measurement of sound from wind turbine generators* (NZS 6808:1998) [1]
- New Zealand Standard 6808:2010 *Acoustics – Wind farm noise* (NZS 6808:2010) [2]
- South Australian *Wind Farms Environmental Noise Guidelines* 2009 (2009 SA Guidelines) [3].

In Western Australia and New South Wales, the 2003 version of the South Australian *Wind Farms Environmental Noise Guidelines* (2003 SA Guidelines) [4] has been adopted for the majority of recent wind farm projects, and this version applies a 35 dB(A) minimum criterion.

It is important to note that both NZS 6808:2010 and the SA Guidelines also consider a 35 dB(A) minimum criterion, although the application of this is limited to particular situations. NZS 6808:2010 states that a 35 dB(A) minimum criterion may be applied in "high amenity areas" which is to be considered only where a district plan promotes a higher degree of acoustic amenity protection to an area, and where the wind speed and measured background noise levels are low enough to justify the application. The SA Guidelines apply a minimum criterion of 35 dB(A) to receivers located in areas primarily intended for rural living, as defined by the relevant Development Plan. However, this is not commonly applied in our experience, as most wind farms are located in zones intended for primary production.

This technical note investigates the effect of applying a minimum criterion of 35 dB(A) based on AECOM's database of background noise measurements at 60 separate receiver locations adjacent to 10 different wind farm developments. Noise criteria are determined for both a 35 and 40 dB(A) minimum criterion, and the difference in criteria between the two cases investigated for three different wind turbine models.

This is also relevant to the recently released *Draft NSW Planning Guidelines: Wind Farms* (Draft NSW Guidelines) [5]. These guidelines propose a minimum criterion of 35 dB(A) and suggest that, because of the 5 dB(A) reduction in the minimum criterion, turbines will be sited approximately twice as far away as would be required in other Australian states.

BACKGROUND NOISE LEVELS

Background $L_{A90,10min}$ noise level measurements undertaken by AECOM at over 60 sites have been collated to determine a mean background noise level at hub height wind speeds for the measurement set. All of the noise level measurements are correlated against hub height wind speeds at the meteorological mast at the proposed wind farm site (a height of approximately 80 metres), and periods of rain and extraneous noise have been removed from the data set. After removal of these data points, the majority of the measurement sites include over 2000 data points, with 12 sites including between 1400 and 2000 data points.

The average background noise level at each integer wind speed for each site was determined by a best fit regression analysis. A mean background noise level for the entire dataset was then determined by averaging the background noise levels at each integer wind speed across the sites. Finally, a best fit regression analysis was conducted on the average background noise levels to determine a background noise level at each integer wind speed for the 60 sites.

The above analysis has been conducted in accordance with the method prescribed in the 2009 SA Guidelines, with the exception that all wind speeds have been considered and not just those between turbine cut-in and rated power. This has been done intentionally to provide an indication of the lower wind speeds at which the 35 dB(A) criterion may affect the end compliance result. This method is similar to the 2003 SA Guidelines except that it considers wind speeds at hub height rather than at 10 metre height. The use of hub height wind speeds is preferable as it minimises the potential effects of air stability which can result in variations in the relationship between wind speeds measured at hub height and those at 10 metres.

Figure 1 presents the mean background noise levels (with bars shown corresponding to one standard deviation), the best

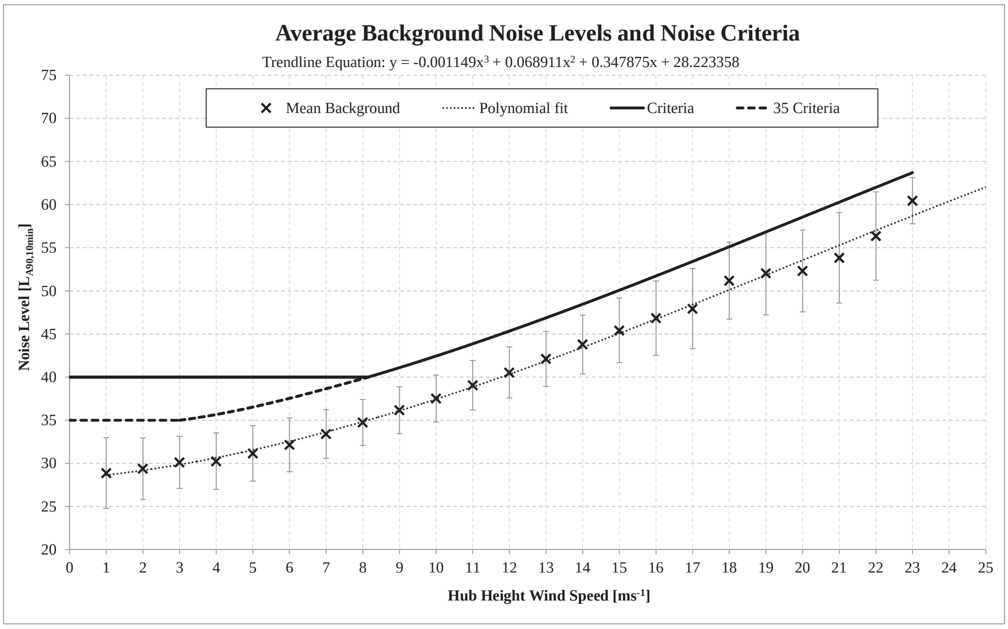

Figure 1. Average background noise levels and noise criteria

fit regression curve and the corresponding noise criteria for a both a 35 dB(A) and 40 dB(A) minimum criterion.

The results indicate that a 35 dB(A) minimum criterion would typically control the noise criteria at hub height wind speeds of approximately 3 to 4 m/s before the background noise level starts to increase with higher wind speed. The criteria determined under both situations would typically be identical at wind speeds of approximately 8 m/s or greater.

WIND FARM NOISE LEVELS

Wind farm noise levels will also increase with increasing wind speed, as the turbine sound power levels increase between cut-in and rated power. Evans and Cooper [6] found that the increase in turbine noise level against hub height wind speed at a receiver location closely matched the increase in the sound power level of the turbines at the wind farm.

Therefore, to approximate the wind farm noise level at a receiver location for comparison with the noise criteria

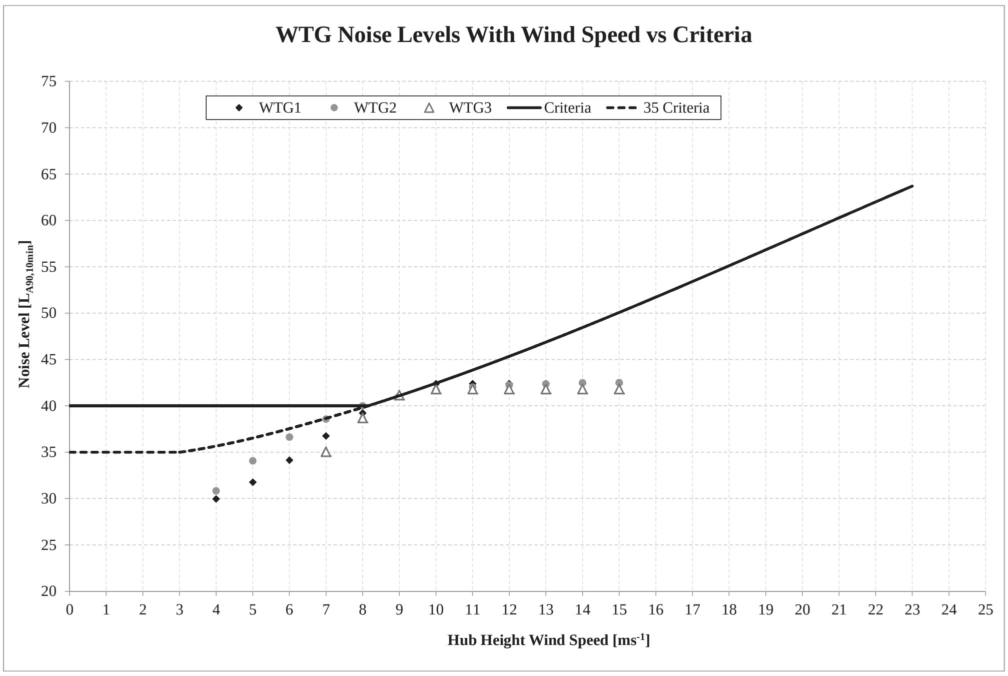

Figure 2. WTG noise levels with wind speed vs criteria

presented in Figure 1, the sound power levels of three modern wind turbines against hub height wind speed were sourced from manufacturer data available online. The three turbines, each from a different manufacturer, are:

- WTG 1: Vestas V112-3.0MW [7]
- WTG 2: Nordex N90 2.5MW [8]
- WTG 3: GE 2.5MW-103 [9].

The sound power versus wind speed profiles for the three turbines were scaled so that compliance with a 40 dB(A) minimum criterion would just be met, emulating noise levels at a location where the wind farm had been designed to comply with a 40 dB(A) minimum criterion. The turbine noise levels are plotted against the previously determined noise criteria in Figure 2.

The results in Figure 2 indicate that the application of a 35 dB(A) minimum criterion has minimal effect on the compliance of a proposed wind turbine layout with the noise criteria, for the turbine models considered. This is as the peak noise levels occur at hub height wind speeds above 8 m/s.

Table 1 summarises exceedances of the criteria that would occur when incorporating a 35 dB(A) minimum criterion at each of the 60 measurement sites based on a turbine noise level just compliant with the 40 dB(A) criteria. It can be seen that the majority of the receivers remain compliant with the more stringent criteria. For 90% of the receiver locations, there would be no noticeable reduction in noise levels (i.e. 2 dB(A) or less) due to the application of the 35 dB(A) minimum criterion, whichever of the three turbine models were selected.

Table 1. Percentage of receiver sites at which exceedance of criteria with 35 dB(A) minimum criterion would occur

Exceedance	WTG 1	WTG 2	WTG 3
0 dB(A)	80%	60%	78%
1 dB(A)	12%	22%	12%
2 dB(A)	3%	7%	5%
3 dB(A)	2%	8%	2%
4 dB(A)	3%	3%	3%
5 dB(A)	0	0	0

DISCUSSION

Based on an analysis of background noise measurements at 60 sites adjacent to 10 different wind farm developments, and manufacturer's data for three different wind turbine models, it appears that a turbine layout designed to comply with a 40 dB(A) minimum criterion would still comply with a 35 dB(A) criterion in the majority of cases. At 90% of the considered receiver locations, there would be no noticeable reduction in noise levels required to achieve compliance with the more stringent criteria (i.e. 2 dB or less). This appears to contradict the assumption that a 35 dB(A) minimum criterion would result in turbines being sited significantly further away from residences.

A further suggestion from this analysis is that Regulatory authorities that currently apply the 2003 SA Guidelines could consider the adoption of the updated 2009 SA Guidelines, with minimal changes to noise levels at residential locations. The 2009 SA Guidelines provide other advantages such as updated noise level measurement, prediction and assessment techniques. The use of hub height, rather than 10 metre height, wind speeds is one example.

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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.

DRAFT NATIONAL AIRPORTS SAFEGUARDING FRAMEWORK - REVIEW OF AIRPORT NOISE GUIDELINES

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The draft framework calls for 'new' noise metrics and associated criteria to supplement the long established ANEF system documented in Australian Standard AS2021. One of the main reasons cited for the suggested changes was that the majority of noise complaints come from residents living outside the 20 ANEF contour. It is widely acknowledged that complaints are a very poor indicator of annoyance. In fact the latest available Sydney Airport Operational Statistics (for November 2011) demonstrate this (as found on Airservices Australia's website). In particular the Noise Complaint Section (p 15 onwards) and the tabulated Complaint History vs Number of Complainants (p 19) highlights that a high number of complaints from a given area does not necessarily mean there are a high number of complainants. For example, the November data shows 1660 complaints from 1 complainant in Kellyville, NSW, which is well outside normal aircraft noise impact zones of any description. Similarly, there were 1239 complaints from 2 complainants in Eastlakes in November 2011. This pattern of complaints is evident during other months of data collected and published by Airservices Australia in 2011. To that end, the draft guideline report does not acknowledge how many resident complaints would come from outside of the proposed more stringent criteria.

If residents outside the 20 ANEF contour have been given an expectation that they will not be affected by aircraft noise, the problem is how noise information is presented and communicated, not the technical means of assessing it. This is where the proposed number above metric (eg N70) can help. Such metrics have been used by practitioners for decades in Australia and are very useful in providing a more comprehendible way of understanding noise impacts and exposure. However, it cannot alone be the sole measure of impacts, just like the current ANEF system is not the sole criteria for aircraft noise. The AS2021 that provides the ANEF criteria requires that in addition to ANEFs, maximum (L_{max}) noise level events are to be assessed in determining effects on land uses. The L_{max} noise level is the basis for the 'number above' metrics (eg N70) and like the N70, the AS2021 also relies on maximum noise levels from aircraft for impact assessment, but importantly not for planning purposes.

The most relevant aspect in the document that is contentious is paragraph 15 of the Guideline:

There should be no new designations or zoning changes that would provide for noise sensitive developments within a 25 ANEF where that land was previously rural or for non urban purposes. Zoning for noise-sensitive development should be avoided where ultimate capacity or long range noise modelling for the airport indicates either:

- *the area is within the 20 ANEF;*
- *20 or more daily events greater than 70 dB(A);*
- *50 or more daily events of greater than 65 dB(A); or*
- *100 events or more daily events [sic] of greater than 60 dB(A).*

The first sentence in the quote above is consistent with the current AS2021 recommendations and there has been wide acceptance of this to date. However, the criteria presented in the second part of the quotation above are not founded on credible scientific studies or information. The first issue is how one reasonably quantifies ultimate capacity or long range operations of an airport. Secondly, the presented criteria appear to be combining traditional planning metrics for new homes near existing airports (ANEF) with more recent 'annoyance' based metrics for new aircraft noise on existing homes on an ad hoc basis. No new data are presented, with reliance placed on a relatively small sample taken some 30 years ago in a study by the National Acoustics Laboratories (NAL). If the new metrics are adopted, then a significant amount of land around airports which is currently available for rezoning for noise sensitive purposes will become sterilised for that purpose.

Whilst the number of movements exceeding 70dB(A) during a 24-hour period and the number of movements exceeding 60dB(A) over the night time period is useful information to allow residents within the community to understand what their reaction to the noise might be, there is no technical justification for setting the number of movements at these levels as criteria to assist in preparing planning guidelines and legislation. These metrics have so far only been used as information to assist the community in understanding the airport noise environment.

These metrics cannot be justified by analysing complaints, since complaints do not correlate well with noise annoyance. The use of criteria around the new metrics for planning purposes is not supported, but the use of information on maximum noise levels under flight paths to assist the community in its understanding of likely noise impacts is useful. For planning purposes, the ANEF system should be retained along with the current AS2021 approach to maximum noise level assessments. Presenting maximum noise level events using N70 and N60 contours should become a formal requirement for information purposes (as it has been used to date) only.

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.

PROTECTING RECREATIONAL USERS OF FIREARMS FROM IMPULSIVE NOISE[1]

The elongated barrel of a silenced weapon is an icon of danger. But can they make us safer?

Remember the movie cliché of the silenced weapon – one dull squeak, followed by bouncing brass that makes a similar level of noise, and the bad guy drops down dead. What does science tell us about the reality behind that story? Just how quiet are silenced weapons? From the recreational shooter's perspective, sound levels are important and the long-term effects of repeated noise exposure can be damaging. Few professional shooters, or doctors for that matter, would doubt the necessity of protecting their ears from the noise that weapons produce.

Matthew Parker Branch, a doctor of otolaryngology, studied the effectiveness of different approaches to the problem of sustained exposure to dangerous noise levels [1]. Specifically, he set out to compare the noise reduction capabilities of commercially available ear-level hearing protection in the form of earmuffs or earplugs, to that of firearm muzzle suppressors (commonly known as silencers).

Muzzle suppressors work by inhibiting the progress of the rapidly expanding gases, so they lessen the noise from the burning propellant and expanding gases on any weapon – supersonic or not. They can even bring great benefits in sound reduction to large, long-range rifle calibres. This makes 'suppressor' a more accurate term than 'silencer'.

In the US, where there are over 250 million privately owned firearms, recreational shooting is a popular sport. Recreational use of firearms is a significant cause of noise and related ear injury in the United States [2]. Unlike industrial exposure, hearing protection during recreational firearm use is not regulated or enforced. This represents one of the largest neglected areas of advocacy for prevention of ear injury. According to Branch's paper [1], approximately 15 percent of Americans between the ages of 20 and 69 (or 26 million people) suffer from hearing loss.

The problem stems in part from the fact that ear-level protection rarely gives the level of protection or noise reduction ratio (NRR) that they advertise. Recommendations from the US National Institute of Occupational Safety and Health (NIOSH) suggest that earmuffs have 25% less NRR than stated and earplugs have 50% less NRR than stated [3].

NRRs are determined using laboratory tests in continuous noise rather than impulsive sounds such as that of gunfire, and consequently do not determine the level of protection given in a particular environment. In addition, lab tests are conducted under ideal circumstances where care can be taken to ensure a good fit to the receptor's head.

Hiram Maxim first introduced and marketed muzzle suppressors in the US in the 1920s. These devices either attach to the muzzle (by way of threading the barrel or by proprietary quick attachment mechanisms) or are integrated into the barrel. Muzzle suppressors allow the heated gases from the barrel to expand into a series of chambers or baffles, cooling and slowing the gas's exit from the barrel. The result is a shorter, quieter sound signature. The basic design of suppressors has changed little over the years, but modern design and manufacturing have improved their sound reduction effectiveness. Unlike ear-level protection, muzzle suppressors are relatively easy to use in a consistent, repeatable fashion. They offer protection for the shooter and bystanders alike and allow interpersonal conversation and situational awareness of sounds not afforded by ear-level devices.

Matthew Branch tested common rifle and pistol calibres with and without muzzle suppression, using strict military/industrial standard sound measurements for impulse noise. The study used a Brüel & Kjær 2209 sound level meter with a B&K 4136 microphone, and a B&K 4220 pistonphone for calibration before and after the testing. The equipment was placed in accordance with the appropriate Military-Standard 1474D protocol. Five shots were fired to establish the unsuppressed sound level, after which 10 shots with the suppressor attached were fired under consistent environmental conditions [4]. The following chart compares the resulting attenuation of the sound level using a muzzle suppressor with the unsuppressed sound level, which would be experienced by ear-level protection alone (without a muzzle suppressor attached).

The suppressors are clearly effective and the attenuation levels reduced the sound levels to under 140 dB in all tests. This is significant because according to NIOSH, the safe threshold for single-impulse sound exposure is 140 dB. However, all rifles and pistols still produced significant noise levels, with the high-powered Remington Model 700 almost reaching 140 dB even with the suppressor in place. The Remmington's bullet, travelling at around 840 m/s, leaves the barrel at approximately two and a half times the speed of sound.

Recognising the difficulties of ensuring that ear-level protection is effectively fitted and unaffected by real-world circumstances, such as interference from eye-protection, NIOSH recommends that shooters use double ear-level protection – both earplugs and earmuffs. However, the best form of hearing protection is that which people will actually

[1] This piece includes excerpts from the paper by Matthew Branch [1] and is reproduced with permission from the Brüel & Kjær Waves article at www.bksv.com/NewsEvents/Waves/Waves_2012_issue9/Muzzle_suppressors.aspx

use. As specialists continue to point out, shooters often don't use any form of hearing protection, hence suggesting double hearing protection is perhaps not the most effective solution. Due to their use at the source of noise production, muzzle suppressors are much more effective at reducing noise. Ideally, both hearing protection and muzzle suppressors should be used by recreational shooters to avoid hearing damage.

Calibre	Weapon	Ammunition	Unsuppressed sound level (mil. std.)*	Unsuppressed sound level (at the ear)	Attenuation with muzzle suppressor (mil. std.)	Attenuation with muzzle suppressor (at the ear)
9 mm	Sig Sauer P226	Remington UMC 147 gr.	160.5 dB	157.7 dB	33.1 dB	28.1 dB
.45 ACP	Glock 21	Remington UMC 230 gr.	162.5 dB	162.5 dB	30.7 dB	33.9 dB
5.56 mm	Colt M4	M855 NATO 62 gr.	164 dB	155 dB	26.6 dB	29.8 dB
7.62 x 41 mm/.308	Remington Model 700	Remington 168 gr.	165.7 dB	157.3 dB	26.8 dB	26 dB

**The military standard (mil. std.) measurement is measured at 1 metre left of the gun's muzzle*

REFERENCES

[1] M.P. Branch, "Comparison of muzzle suppression and ear-level hearing protection in firearm use", *Otolaryngology -- Head and Neck Surgery* **144**(6), 950-953 (2011)

[2] W.W. Clark, "Noise exposure from leisure activities: A review", *Journal of the Acoustical Society of America* **90**, 175-181 (1991)

[3] E.H. Berger, J.R. Franks and F. Lindgren, *International review of field studies of hearing protector attenuation.* In: *Scientific basis of noise-induced hearing loss*, Editors A. Axelsson, H. Borchgrevink, R.P. Hamernik, P. Hellstrom, D. Henderson, R.J. Salvi, New York, Thieme Medical Publishers, pp. 361-377, 1996

[4] P.H. Dater, *Firearm sound level measurements: Technique and equipment*, 2nd edition, ATI Star Press, Idaho, 2000

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 proposed 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 tentatively been set as*:

Delegate	$840	$720 (early bird)
Student	$320	$255 (early bird)
Accompanying person	$140	

**An additional GST applies to Australian based delegates*

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 CD 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 both the technical exhibition and poster display.

The Congress Banquet is not included in the registration fee.

Technical Program

After the welcome and opening ceremony on Sunday 16 November, the following three days will involve 10 parallel sessions covering all fields of noise control. Major areas will include Community and Environmental Noise, Architectural Acoustics, Transport Noise and Vibration, Human Response and Effects of Low Frequencies and Underwater Noise. A series of distinguished lectures and workshops are planned to cover topics such as:

- Noise impact on high density living
- Impact on dense living
- Wind turbine noise
- Active noise control
- Aircraft noise
- Power station noise

Organising and Technical Committee

- Congress President: Dr Norm Broner
- Technical Program Chair: Adjunct Professor Charles Don
- Technical Program Co-Chair: Adjunct Professor John Davy
- Technical Program Advisor: Mrs Marion Burgess
- Proceedings Editor: Mr Terry McMinn
- Sponsorship and Exhibition Manager: Dr Norm Broner
- Congress Treasurer: Ms Dianne Williams
- Social Program Chair: Mr Geoff Barnes
- Congress Secretariat: Ms Liz Dowsett

Further details are available on the congress website

www.internoise2014.org

News

2011 Australian Young Professional Engineer of the Year goes to AAS member

One of the QLD Division members, Emma Charlton, was announced as the winner of the 2011 Australian Young Professional Engineer of the Year. Emma Charlton is an Associate Director of AECOM and manager of the Acoustical group in Brisbane. She has been working in the engineering profession for ten years and has played a key role in a wide variety of projects in the fields of noise and vibration, with a particular focus in the areas of architectural acoustical design, environmental noise modelling, vibration control and road and rail noise impact assessments. Emma has held a key leadership position for AECOM for over five years. She has also managed the Acoustics discipline in Queensland since January 2008. One of the biggest challenges and opportunities arising during her time at AECOM has been responding to the merger of Bassett and AECOM in 2004. She has been responsible for overseeing the implementation of a number of changes to the reporting and management systems since the merger. Emma also acts as project director on multiple projects in the Buildings group, providing high level leadership to the project team. This includes high level project direction, identification and management of risk and management of the client interface at a senior level. In recognition of her role as a leader in the business, Emma was recently selected to attend the inaugural Senior Leadership Development Program developed by AECOM in conjunction with the Australian Graduate School of Management. She was one of only 25 people from the 4500 staff across Australia and New Zealand selected for the course and the only person under 35 years of age. Emma is an active member of the engineering community and participates on a number of committees including the Standards Australia committee AV-003 on Acoustics Human Effects and the UDIA Women in Development committee.

AAS Awards

In addition to the AAS Excellence in Acoustics Award and the annual Education Grant, there are a number of annual prizes and awards made by the Divisions of the AAS. The QLD Division has establish the Colin Speakman Travel Bursary which provides up to two travel awards and the NSW Division which provides up to three travel grants for research students to attend the annual conference of the AAS. The SA Division has the David Bies prize for meritorious contribution to acoustics and the WA Division has a prize for the best student project in acoustics and vibration. For more information on all these prizes and awards see www.acoustics.asn.au/joomla/notices.html

The Colin Speakman Travel Bursary

The QLD Division has established student travel bursaries. The awards, which will be offered annually, are established in honour of the late Colin Speakman FAAS. The two bursaries of up to $1200 each are provided to assist with travel and registration costs associated with attendance at the annual conference of the AAS, or in years when Australia hosts or co-hosts an international acoustical conference, the relevant international conference (this includes conferences in the Inter-Noise, ICA and Wespac series). To be eligible for travel assistance through these awards, applicants must be a student at a university in Queensland, engaged in research in an area relevant to acoustics, towards the award of an Honours, Masters or higher degree and must be presenting a paper at the conference.

In addition, the QLD Division has determined that for 2013 and subsequent years, monies available through our final year undergraduate/1st year postgraduate, student research bursaries, (the Acoustic Bursary and the RJ Hooker Bursary) are to increase, to $2000 each. Research bursary awardees, who have a paper accepted for presentation at the annual conference in the year in which the award is made (or in the year following) are also eligible to apply for a further grant of up to $250, as a contribution to their registration as a student delegate. The Queensland Science Contest Acoustic Bursary will also increase, to $600. The latter is awarded to primary and secondary school students participating in the Science Contest.

More information on all these prizes and awards can be found at www.acoustics.asn.au/joomla/notices.html

ICA call for early career nominations

The AAS are invited to consider nominating one of their members for consideration for the prestigious the ICA Early Career Award to be presented at the ICA 2013 in Montreal, 2-7 June 2013. This award is presented to an individual who is relatively early in his/her professional career (about 10-15 years of active career), who has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics or both and who has been active in the affairs of acoustics through the AAS. The Award consists of an Award Certificate, a Medal, and an Honorarium. The honorarium for the Early Career award to be announced at the ICA 2013 Congress will be Euro 1,000 plus up to Euro 1,000 for travel to the congress. The nomination deadline is 1 October 2012 and that all nominations should be sent by email to ICAECGrantChair@icacommission.org.

Details about the award, the nomination process and the required documentation to support the nomination are given on the ICA web at www.icacommission.org/eaward.html

Relocation for Wilkinson Murray

Some important milestones at Wilkinson Murray have happened this year. We celebrate 50 years of consulting since Roger Wilkinson first established Carr and Wilkinson in 1962 before being joined by Barry Murray in 1976. After 20 years with the Sydney office in Crows Nest (Willoughby Road) we have recently moved to.... Crows Nest and now have a wonderful office at Level 4, 272 Pacific Highway with sweeping views over the harbour and out to the Blue Mountains (when there are no temperature inversions which trap the pollution). Wilkinson Murray also has offices in Queensland and Orange as well as an office in Hong Kong established by Barry Murray in 2006. Barry recently returned to Sydney in July to join the management team of John Wassermann, Neil Gross and Ben Lawrence. With John Wassermann's experience, Wilkinson Murray is also now providing consulting in air quality as well as acoustics.

Health of Australian Science Report

Australia's Chief Scientist, Professor Ian Chubb has produced a report on the Health of Australian Science. This provides a comprehensive overview of Australia's science system, outlining our strengths and vulnerabilities. Professor Chubb has said that overall, ours was a healthy and robust system, but that some identified challenges would lead to long term issues for Australia if no action is taken and "the future prosperity of Australia is dependent on having a strong supply of graduates in the right areas coming through the education system. There are some areas of expertise that are crucial to our national interest which are lacking what they need to prosper".

Agricultural sciences, physics, mathematics and chemistry are highlighted in the report as being vulnerable and all are crucial for Australia's future. The total numbers in Engineering don't meet demand and there are shifts between disciplines. Opportunities outlined in the report include developing a more strategic funding system and improving the relationships between science and industry. Chubb also remarked that we need to develop a culture that appreciates a science education, both the students and the teachers of it. "The

Health of Australian Science Report is not a story about rebuilding after a train wreck. We do not have a train wreck. But the Report is a signal: it encourages us to be alert; to be prudent while willing to take bold action when we need to." The full report is available from www.chiefscientist.gov.au

Revision of AS/NZS Occupational Noise Management - Part 4: Auditory Assessment

Work on revision of AS/NZS Occupational Noise Management - Part 4: Auditory Assessment will commence soon. If anyone has experience using the present standard and has suggestions for improvements please contact Warwick Williams <Warwick.Williams@nal.gov.au>, Chair of Committee AV-003.

US NIOSH report on damage risk criterion for impulse noise

This is an important report giving confidence in the use of $L_{Aeq,8h}$. The National Institute for Occupational Safety and Health (NIOSH) has analysed several audiometric databases of human and animal impulsive noise exposures and concluded that $L_{Aeq,8h}$ is preferable to using MIL-STD 1474D or the Auditory Hazard Assessment Algorithm.

More information at www.cdc.gov/niosh/surveyreports/pdfs/350-11a.pdf

Safe Work Australia to progress work on vibration

Fact sheets on hand-arm vibration (HAV) and whole body vibration (WBV) will soon be published on the Safe Work Australia website and the following work has been agreed to:

- Development of a code of practice for vibration based on European material.
- Development of health monitoring guidance based on UK HSE material.
- Approach Standards Australia to update the HAV and WBV standards to be direct adoptions of the current ISO standards.
- Investigate the merits of the development of competencies/training for vibration assessment and measurement.

New Products

Acoustic camera from HW Technologies

Nearfield acoustic mapping measurements can be made with the portable handheld paddle that features 48 double-layered microphone channels allowing for real-time Acoustic Holography measurements. The double-layer structure enables the measurement of acoustic pressure signals while particle velocity/acoustic intensity on the measurement plane is calculated and mapped simultaneously. Furthermore, this microphone layout facilitates a differentiation between noise sources on the measurement plane or in the field behind the paddle. The double-layered microphones act like intensity probes, delivering a vector used to calculate an acoustic map. The paddle weighs 2 kg with an array body diameter of only 30 cm. The recommended measurement distance is 0.1 to 0.15 m and it can be used to distinguish airborne from structure-borne sound. More information from

www.hwtechnologies.com.au

Fantech faster fan selection

Faster fan and silencer selection is now possible using the Fantech Product Selection Program. Fantech first launched the 'Interactive Product Suite' (as it was first known) in 1998. Over the next 10 years, Silencer selection and an acoustic analysis module were added. Other functions such as real-time linking with the Fantech website and the ability to export fan schedules to Excel were also included. Users can now see all the technical data in one easy view and have the ability to recall it at any time from a schedule. They can also search by any product code, export to Excel and PDF, and save all 2D DWG, DXF and 3D Revit files simultaneously. Energy efficiency calculations have also been incorporated into every selection. Another key advantage of the new program is the introduction of a "Basic" mode to make it easier for first time and less-technical users. This mode is ideal for architects, builders and draftsmen as it allows them to make quick selections of fans or silencers and get the detail they need in five simple steps. The "Advanced" mode was designed to be used by engineers, contractors and consultants who required selections based on more specific criteria. More information and a copy of the program can be obtained from www.fantech.com.au

Meeting Reports

NSW Division

On 25th July, Andrew Parker and Joon-Pil (JP) Hwang from SLR Consulting Australia Pty Ltd gave a talk entitled *Vibration mitigation of a bridge by use of a tuned mass damper*. Their case study highlighted the measurement, modelling and analysis techniques that were implemented in the design of a tuned mass damper (TMD) for reducing footfall vibration on a pedestrian/cycle bridge that was recently opened in Sydney.

The NSW Division has awarded travel grants to six (6) research students to attend the Acoustics 2012 Fremantle conference in November. The amount of each award is $1200. Each student will be presenting their research work at the conference.

QLD Division

The Queensland Division hosted a technical meeting on 11th April entitled *Update on the application of environmental criteria and the Queensland development code to proposal development with transport noise corridors*. The guest presenter was Arthur Hall, Principal Advisor (Road Traffic Noise Management) Geospatial, Road Assets & Design Engineering & Technology at the Department of Transport and Main Roads. Arthur is well known to the acoustic community in Queensland through his time with TMR, and he informed the meeting at the beginning that he has been with the department since he was 18! The MP4.4 policy is very topical in Queensland at the moment and this was highlighted by the attendance at the meeting with standing room only available. Arthur presented an overview of the policy, the TMR policy position statement as well as information on how the transport noise corridors were developed. The content presented by Arthur was very informative and provided a regulatory insight in the development of this new policy. There was plenty of debate and strong opinions provided during discussions however the meeting was held in good humour. The Queensland Division would like to thank Arthur for taking the time and presenting at the technical meeting.

Standards Australia

The AAS has the opportunity for a representative on all relevant Standards Australia committees. At this time there are two important committees for which there is no specific AAS representative:

AV-002 - Acoustics Instrumentation and Measurement Techniques - this committee deals with instrumentation and measurement excluding architectural acoustics (which is dealt with by another committee)

AV-003 - Acoustics Human Effects - this committee deals with Standardisation in the field of hearing conservation of persons exposed to noise in the course of their occupation

If any member of the AAS would be interested to become a member of either these committees please contact the AAS General Secretary at GeneralSecretary@acoustics.asn.au.

Future Conferences

ACOUSTICS 2012 Fremantle

The annual conference of the Australian Acoustical Society will be held at the Esplanade

Hotel in Fremantle, Western Australia, from 21-23 November 2012. The theme for this conference is “Acoustics, Development, and the Environment”, which is very relevant in the Western Australian context given the significant urban, mining and infrastructure development being undertaken at present.

Acoustics 2012 Fremantle will include sessions addressing the acoustical and vibration aspects of major developments, and will disseminate up-to-date methodologies and practices. The conference will also include sessions and workshops on acoustical topics that fall outside of the main theme. The overview program is available. There is a welcome function on the Wednesday and the conference opening on Thursday morning. Papers continue through to Friday afternoon. The abstracts that have been submitted can be viewed from the link on the website. More information at www.acoustics.asn.au/joomla/acoustics-2012.html

ICA 2013 Montréal, Canada

The 21st International Congress on Acoustics, ICA2013, will be held 2-7 June 2013 at the Palais des Congrès in downtown Montréal, Canada. This meeting is hosted by the Acoustical Society of America (ASA) and the Canadian Acoustical Association (CAA). The high standard technical program will include plenary, distinguished, invited, contributed and poster papers covering all aspects of acoustics. There will be an extensive technical exposition highlighting the latest advances in acoustical products. The call for papers will be going out early October.

Important dates: Abstracts will be due 15 November 2012 and full papers 22 January 2013.

Several satellite meetings on specialised topics will be held in conjunction with ICA2013. The International Symposium on Room Acoustics (ISRA) will be held 9-11 June 2013 in Toronto, immediately following the ICA. More information on ISRA can be found at www.ISRA2013.com.

More information on ICA2013 can be found at www.ica2013montreal.org

ICSV20 Bangkok, Thailand

The 20th International Congress on Sound and Vibration (ICSV20) will be held 7-11 July 2013 in Bangkok, Thailand. The conference will be held at the Imperial Queen’s Park Hotel which is strategically located in the city centre and the important commercial district, with direct access to a lush public park. The skytrain, shopping and entertainment complex are within a short walking distance from the hotel. The expressway and subway are also located nearby.

Important dates: Abstracts will be due 1 December 2012 and full papers 1 April 2013

More information at www.icsv20.org

Inter-noise 2013 Innsbruck, Austria

The 42nd International Congress and Exposition on Noise Control Engineering will be held in Innsbruck, Austria from 15-18 September 2013. The Congress is being organised by the Austrian Noise Abatement Association for the International Institute of Noise Control Engineering. Innsbruck, the capital of the Tyrol, is located in the Alpine region of Austria, in the valley of the river Inn, at 580 metres above sea level. It is surrounded by mountain ranges and numerous peaks which reach an altitude of 2700 metres above sea level. The city has 140,000 inhabitants and hosts one of the oldest universities in Europe, founded in the year 1562. The conference will be held at the award winning Innsbruck Congress Centre.

More information at www.internoise2013.com

Diary

2012

9 – 13 September, Portland, USA
nternational Conference on Noise and Vibration Engineering (ISMA 2012)
http://www.isma-isaac.be/conf/

17 – 19 September, Leuven, Belgium
ISMA Noise and Vibration Engineering Conference (ISMA2012)
http://www.isma-isaac.be/conf/

21 – 23 November, Perth, Australia
ACOUSTICS 2012 Fremantle
http://www.acoustics.asn.au/joomla/acoustics-2012.html

2013

1 - 4 May, Singapore
International Congress on Ultrasonics (ICU 2013)
http://www.epc.com.sg/PDF Folder/ICU 2010 Phamplet v1 (12 Jul 2010).pdf

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

2 – 7 June, Montréal, Canada
21st International Congress on Acoustics (ICA 2013)
http://www.ica2013montreal.org

9 - 11 June, Toronto, Canada
International Symposium on Room Acoustics (ISRA 2013)
http://www.isra2013.com

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

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

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

2014

17 – 19 November, Melbourne, Australia
Inter-Noise 2014
http://www.internoise2014.org/

2015

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

2016

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

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

VIBRATION ISOLATORS FOR NOISY ELEVATORS

In a new luxury 20-storey residential tower in downtown Taipei City, Taiwan, vibration from the motor driving four Mitsubishi elevators was causing significant structure-borne noise. This noise transferred through walls and into living units whenever the lifts were running, which was very often.

The problem was extremely noticeable on the upper levels of the building, including the most expensive apartment. The residents of the building had complained loudly to Mitsubishi about the effect of lift noise on their sleep and quality of life.

In accord with common industry practice, the original design of the elevator relied on straight rubber to isolate the motor from its mount. However this material didn't adequately handle the vibration in this case.

In an attempt to remediate the problem, Mitsubishi spent substantial time and money trying two other kinds of rubber isolator, without success. A vibration damping product was required to provide good isolation from 50 to 160 Hz. Further noise and vibration tests revealed a peak frequency around 400 Hz. Thus, an isolator material that would flex at low loads but remain strong at high loads was needed.

In consultation with Pyrotek Noise Control in Taiwan and Pyrotek's Product Development in Melbourne, an isolation material called Sylomer was selected. Sylomer is an elastic polyurethane material that deforms under tension and compression loads, but always returns to its original form. The results illustrating the improvement in noise levels before and after installation of the Sylomer vibration isolators in the four Mitsubishi elevators are given in Table 1.

The acoustic performance of the Mitsubishi elevators was significantly improved, much to the delight and relief of the building tenants.

Table 1. Noise levels before and after installation of the Sylomer isolator pads

	Frequency Range					
	50 – 400 Hz			Overall		
Lift	Before dBA	After dBA	Improvement dBA	Before dBA	After dBA	Improvement dBA
1 & 2	39.0	34.2	4.8	48.7	39.5	9.2
3 & 4	42.9	28.9	14.0	45.8	38.6	7.2

Sustaining Members

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

3M AUSTRALIA
www.3m.com

ACOUSTIC RESEARCH LABORATORIES
www.acousticresearch.com.au

ACRAN
www.acran.com.au

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

ADAMSSON ENGINEERING
www.adamsson.com.au

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 COLPRO
industrialacoustics.com/australia

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

PEACE ENGINEERING
www.peaceengineering.com

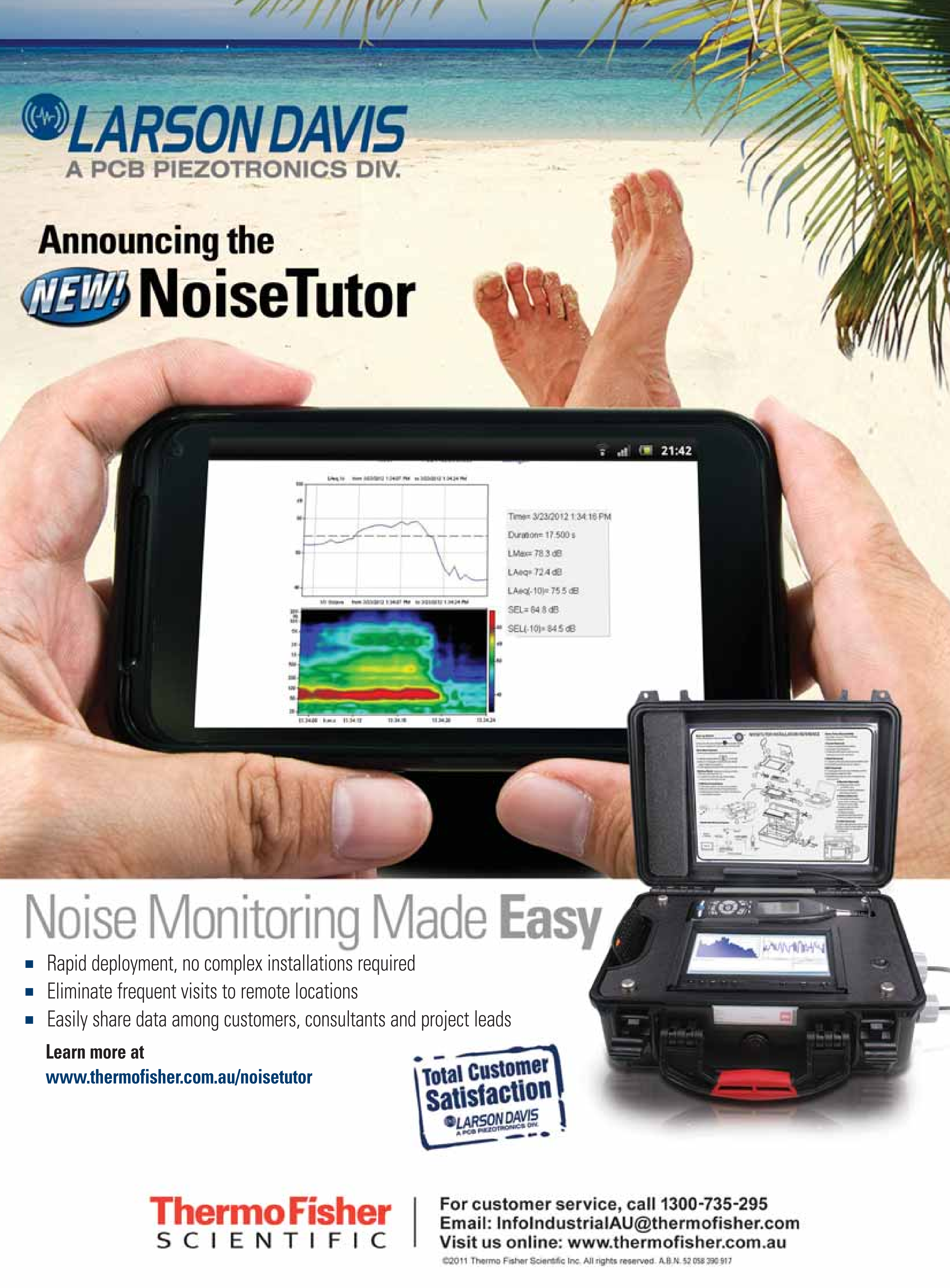
PYROTEK NOISE CONTROL
www.pyroteknc.com

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


LARSON DAVIS
A PCB PIEZOTRONICS DIV.
Announcing the
NEW! NoiseTutor
21:42
Time= 3/23/2012 1:34:16 PM
Duration= 17.500 s
LMax= 78.3 dB
LAeq= 72.4 dB
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