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Session 1A

Engineering Acoustics

The Modelling Of Radiation From An Expanded-Metal Press Using Dimensional Analysis

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Abstract

As part of a study in quantifying the noise radiated during the operation of an expanded-metal press, it has been found that the radiated noise level is dependent on the sheet material properties (bending stiffness and density), the sheet thickness, the sheet surface area and the press operating speed. For a given press, it will be convenient to be able to predict the radiated noise level based on limited measurements. In this paper, it will be shown that all the six critical variables of an expanded-metal press can be organised into three non-dimensional groups using dimenional analysis. Extensive field measurements of the radiated noise from the expanded-metal press indicate that these results when expressed in terms of these three dimensional groups, collapse onto a single curve, within the limits of experimental accuracy. Based on these results, it can be shown that the radiated noise level of an expanded-metal press can be predicted once the operating conditions of the press are known.

Introduction

This paper reports some aspects of the continued collaborative research undertaken by the Acoustics and Vibration Centre (AVC) and BHP Building Products (BHPBP). The overall aim of this research is to reduce noise on a 120 tonne Emil Bender press which has been modified to manufacture expanded-metal mesh from flat metal plate feedstock [1-8]. A schematic of the press and measurement location is shown in Figure 1. The particular research reported herein presents, using dimensional analysis, the three non-dimensional groups for the radiated noise from this press.

The explicit aim of this paper is to document how enormous amounts of experimental data can be condensed into a meaningful single curve. Eager *et al.* [4] had previously acquired acoustical radiation data at five different press operating speeds and three different feedstock thicknesses, giving a total of 15 data sets. Dimensional analysis is herein employed to reduce this somewhat complex and voluminous data into a simpler form. This in turn allows the predictions of radiated noise levels based on limited measurements.

Dimensional analysis is a useful data reduction technique. It is particularly helpful in problems where a rigorous theoretical analysis is difficult or impossible. The basis of this method is the so-called principle of dimensional homogeneity, which states that in any equation representing a physical relationship the dimensions of the left-hand side of the equation must be the same as the right-hand side. Two different methods of dimensional analysis were used to find the dimensionless groups for the expanded-metal (XPM) press. The first method was developed by Lord Rayleigh and uses the principle that any relationship can be written as a power series. The second method is more systematic and uses the so-called *Pi* (π) theorem, first attributed to Buckingham. Only the latter is presented herein.

Selection of primary control variables

The following process variables were identified during the initial inspection of the XPM press as parameters that may affect the overall sound pressure level L_p during normal production:

1. Feedstock thickness;
2. Feedstock length;
3. Feedstock width;
4. Press speed;
5. Cutting blade clearance;
6. Cutting blade profile;
7. Strand width (feedstock feed rate);
8. Condition of cutting blades;
9. Use and type of cutting lubricant;
10. Position and method of feedstock constraint; and
11. Feedstock material.

Clearly, the number and diversity of the identified variables precluded the testing of all the possible combina-

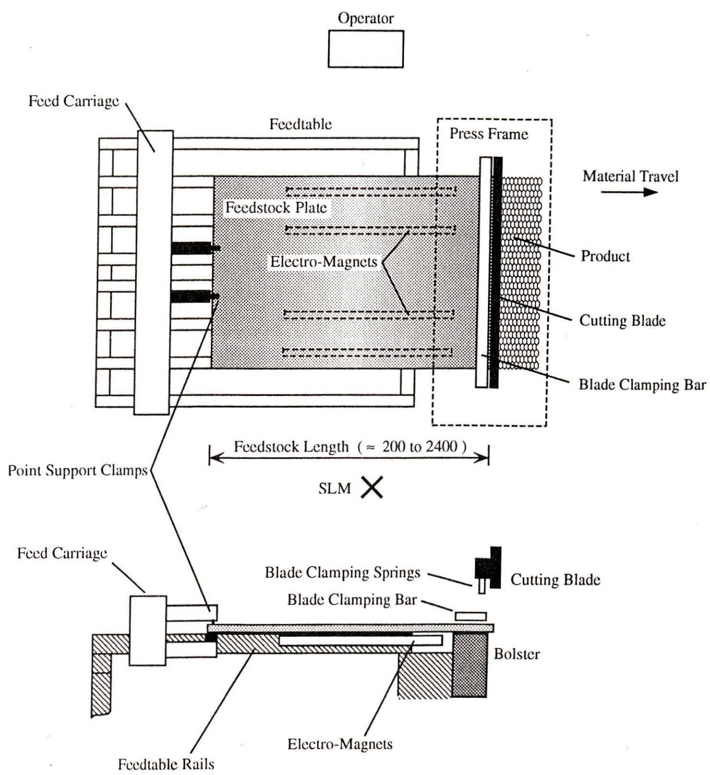

Figure 1: Expanded-metal press and measurement configuration

tions and permutations. The complexity of this task was reduced after a preliminary noise survey of the press revealed that the sound pressure level L_p on the feed-side of the press was some 10 dB(A) higher than the product-side. This allowed the variables associated with the cutting noise to be ignored until noise reduction techniques had been successfully applied to the feed-side of the press. This was possible as the feedstock radiation was masking the noise from both the cutting process and machine operation.

The next obstacle to overcome was the selection, and agreement between BHPBP and the AVC, of a product to which all tests could be referenced. It was essential that these tests be easily repeatable so that comparisons could be made as noise reduction techniques were applied or when production modifications were made to the press. The product that was finally selected was the 'Gridmesh' 50075 manufactured from mild steel, see Figure 2. This product was chosen for the following reasons:

1. Economically, 'Gridmesh' 50075 was considered the most cost effective option. It was noted that there was no standard product that allowed the testing of the proposed test variables and the manufacture of 100% saleable product. No matter how the testing was performed it would involve the manufacture of scrap product. To reduce the amount of scrap product and hence the cost of the testing it was desirable to produce, wherever possible, standard product which could be sold. We therefore chose a series of tests that limited the amount of scrap by using 5 mm feedstock and the 'Gridmesh' *x*0075 product configuration as the reference (where *x* is the feedstock thickness). Further, the raw material was identified as the major manufacturing cost, this cost being proportional to the weight of the material. By manufacturing scrap from the thinner, or lighter gauge, feedstock the cost of the trials was further reduced;
2. 'Gridmesh' 50075 was a high volume product. This was important as it meant that any product produced would not be held in stock for extended periods of time and hence would not increase the inventory stock levels. The testing could thus be scheduled more readily into a very tight and complex production schedule;
3. The machine downtime would be limited as using 'Gridmesh' 50075 produced the greatest quantity of saleable product. The time spent doing the trials was equivalent to running normal production. This allowed the tests to be programmed within the normal production schedule. This also meant that it was not necessary to retool the press before and after each series of tests, giving a potential production time saving of approximately 8 hours per test run; and
4. 'Gridmesh' 50075 had previously been identified as the nosiest standard product. Since a major aim of the project was to reduce the noise exposure to comply with present and future legislation, it was logical

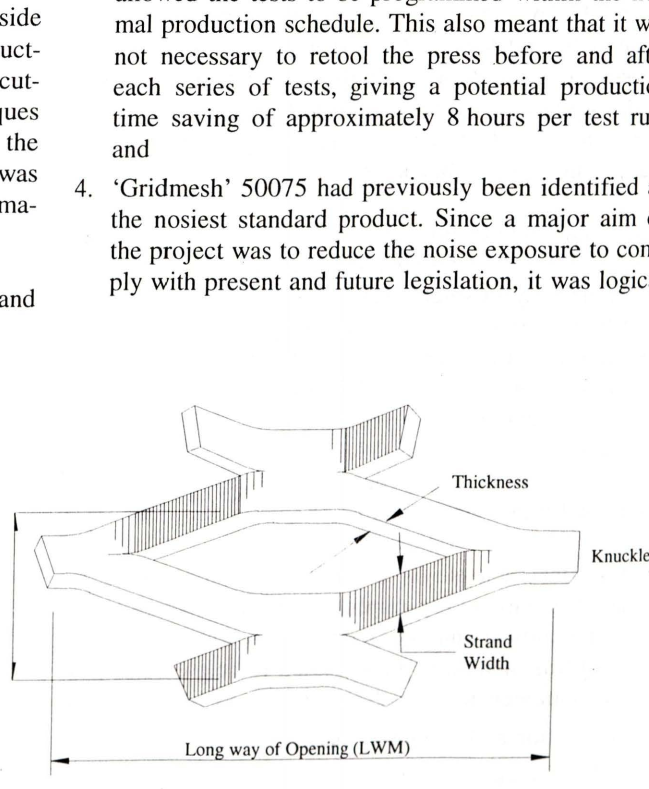

Figure 2: 'Gridmesh' 50075 product

to conduct the majority of the tests using the product that had the greatest potential to affect the operator's daily noise dose.

Selection of variables for dimensional analysis

We identified the following issues which suggested that dimensional analysis would be of benefit in the study of the XPM press:

1. The number of tests conducted would be reduced by using dimensional analysis to select the most appropriate variables. These could be used as control variables during the experimentation phase of this investigation;
2. Establish the dimensionless parameters for the XPM press which would reduce the measured data into a more compact and meaningful form. Through such incisive and uncluttered presentations of information, researchers are able to discover new features and missing areas of knowledge;
3. Produce a generic dimensionless graph that could be applied widely throughout the XPM industry for predicting L_p at various standard operating conditions for specific parameters. Dimensional analysis would make it possible to generalise the limited experimental results to situations involving different feedstock material properties, feedstock material dimensions, and press variables. The consequence of this generalisation is manifold, since one is now able to describe the phenomenon in its entirety and not be restricted to the actual experiment that was performed. Thus it was possible to conduct fewer highly selective experiments to achieve the same result and thereby achieve savings in both time and cost; and
4. Dimensional analysis allows management to predict the L_p of the press for various production situations. For example, a machine speed for a particular product can be chosen which minimises the noise exposure of the press operator.

In selecting the variables to be used in the dimensional analysis of the XPM press we considered the following issues:

1. Since the measured sound pressure level L_p is in dB which is dimensionless, it has to be converted into rms sound pressure (p) in Pa using Equation 1:

$$p = 2x10^{-5} * 10^{L_p/20} \tag{1}$$

   This relationship was applied to all the measured data to obtain a dimensional value of the rms sound pressure (p). This was then used as the primary independent variable in the dimensional analysis.
2. Feedstock radiation had previously been identified as a major noise source. It is well documented that L_p is proportional to the surface area of the radiating body. It was also noted during the preliminary measurements that L_p dropped with decreasing feedstock length. It was therefore postulated that the rms sound pressure (p) was a function of the feedstock length (a);
3. It is also well documented that the stiffness of a vibrating plate affects the sound radiation. Of all the types of waves that can travel in plates bending waves are by far the most important for sound radiation. It was postulated that the feedstock radiation would depend on the bending stiffness or flexural rigidity (D) of the feedstock material. Recalling [9] the bending stiffness (D) for a rectangular plate is:

$$D = \frac{Eh^3}{12(1-\nu^2)} \tag{2}$$

   where E is the Young's Modulus of the feedstock material, h is the thickness of the feedstock and ν is the Poisson's ratio of the material. It was therefore postulated that the rms sound pressure (p) would be a function of the bending stiffness (D) of the feedstock;
4. During the preliminary measurements it was also noted that L_p increased with both increasing feedstock thickness (h) and machine operating speed (spm). The feedstock adjacent to the cutting-blade was visually observed during the cutting phase to bend and lift both itself and the blade-clamp. The force from the cutting-blade impacting upon the feedstock was greater than the force constraining the feedstock. As the cutting-blade retracted, the gravitational and strain energies stored in the deformed feedstock were subsequently released and the feedstock slapped the feedtable. It was this slapping that produced the difference in L_p between the feed and product sides of the press. To keep the units consistent for dimensional analysis the machine operating speed data was converted to sps as follows:

$$sps = \frac{spm}{60} \tag{3}$$

   It was therefore postulated that the rms sound pressure (p) would be a function of the feedstock thickness (h) and the press operating speed (sps).
5. It was noted by inspection that the feedstock thickness (h) and feedstock length (a) combination would yield a dimensionless shape factor.

   To derive a third π group it would be necessary to introduce another variable. Feedstock density (ρ) was chosen as it is a material dependent variable and introduced the mass dimension that would help to account for the gravitational forces acting on the strained feedstock. It was further postulated that the rms sound pressure (p) would be a function of the feedstock density (ρ).

In summary it was postulated that the rms sound pressure (p) would be a function of the feedstock thickness,

Table 1: Dimensional analysis, units & dimensions

Variable	Symbol	Units	Dimensions (MLT)	Comments
Sound pressure	p	Pa	$ML^{-1}T^{-2}$	Measured variable
Feedstock thickness	h	mm	L	3, 4 and 5 mm
Feedstock length	a	m	L	≈ 0.3 τo 2.4 m
Press speed	sps	s^{-1}	T^{-1}	65, 80, 100, 120 & 140 spm
Bending stiffness	D	N.m	ML^2T^{-2}	$D - f(E, h^3, v)$
Density	ρ	kg/m^3	ML^{-3}	7,860 kg/m^3

feedstock length, press speed, feedstock bending stiffness and the feedstock density, as given in Equation 4:

$$p = f(h, a, sps, D, \rho) \tag{4}$$

The density was held constant throughout the investigation by conducting all the tests using mild steel feedstock. Also, the effects of the feedstock boundary conditions were eliminated by conducting batches of tests with the feedstock constrained in exactly the same manner. The feedstock was constrained by pinch-grip clamps symmetrically located on the trailing edge of the feedstock at 450 mm separation (Figure 1). The feedstock width was held constant at 1200 mm so that the feedstock area was proportional to the feedstock length. The machine speed was varied in increments from the minimum press speed, 65 spm, up to the maximum machine speed, 140 spm. The feedstock thickness was varied in three equal increments from 3 mm up to 5 mm. Table 1 lists a summary of the quantities used for the dimensional analysis derivation.

Buckingham's method

The Buckingham *Pi* (π) theorem was used here to determine the dimensionless groups for the XPM press. This theorem states that any homogeneous equation expressing a functional relationship ϕ between n variablesA$_1$, A$_2$, A$_3$,...A$_n$, of the form

$$\phi[A_1, A_2, A_3, \ldots A_n] = 0 \tag{5}$$

has a solution of the form

$$\phi[\pi_1, \pi_2, \pi_3, \ldots \pi_n] = 0 \tag{6}$$

where each π is a dimensionless group and v is the number of fundamental dimensions required to represent the n variables. Since the fundamental dimensions are mass (M), length (L) and time (T), v=3. As the number (n) of the variables as listed in Table 1 is 6, we expect n - v = 3 dimensionless groups for the XPM press. From Table 1 and Equation 6 we have:

$$\phi[\pi_1, \pi_2, \pi_3,] = 0 \tag{7}$$

Ideally it would be desirable to obtain a solution containing p, h and sps in each π group. Therefore we will solve for p, h and sps.

Arranging the dimensional information in tabular form, where the indices of the fundamental units appear in rows. Each column is numbered from the left-hand side (K_1, K_2, K_3, K_4, K_5 and K_6).

	K_1 p	K_2 h	K_3 sps	K_4 a	K_5 D	K_6 ρ
M	1	0	0	0	1	1
L	-1	1	0	1	2	-3
T	-2	0	-1	0	-2	0

We can now write the dimensional equation in terms of the K_s for each fundamental unit as follows for M, L and T respectively.

That is for M

$$K_1 + K_5 + K_6 = 0 \tag{8}$$

and for L

$$-K_1 + K_2 + K_4 + 2K_5 - 3K_6 = 0 \tag{9}$$

and for T

$$-2K_1 - K_3 - 2K_5 = 0 \tag{10}$$

As three π groups are expected and it would be preferable for sound pressure (p), feedstock thickness (h) and the press operating speed (sps) to appear independently, we will solve for K_1, $K2$ and $K3$ in terms of K_4, $K5$ and $K6$; that is the K's associated with the variables a, D and ρ.

$$K_5 + K_6 = -K_1 \tag{11}$$

$$K_4 + 2K_5 - 3K_6 = K_1 - K_2 \tag{12}$$

$$2K_5 = -2K_1 - K_3 \tag{13}$$

Letting $K_4 = 1$, $K_5 = 0$ and $K_6 = 0$, we obtain

$$K_1 = 0 \quad K_2 = -1 \quad K_3 = 0 \tag{14}$$

Now letting $K_4 = 0$, $K_5 = 0$ and $K6 = -1$, we obtain

$$K_1 = 1 \quad K_2 = -2 \quad K_3 = -2 \tag{15}$$

Also letting $K_4 = 0$, $K_5 = -1$ and $K_6 = 0$, we obtain

$$K_1 = 1 \quad K_2 = 3 \quad K_3 = 0 \tag{16}$$

We can now write the dimensionless groups in matrix form using the values of the K_s derived in Equations 14, 15 and 16 above.

	K_1 p	K_2 h	K_3 sps	K_4 a	K_5 D	K_6 ρ
π_1	0	-1	0	1	0	0
π_2	1	-2	-2	0	0	-1
π_3	1	3	0	0	-1	0

Collecting the terms from the matrix we obtain

$$\pi_1 = \frac{a}{h} \tag{17}$$

$$\pi_2 = \frac{p}{h^2.sps^2.\rho} \tag{18}$$

$$\pi_3 = \frac{ph^3}{D} \tag{19}$$

$$\phi[\pi_1, \pi_2, \pi_3] = 0 \tag{20}$$

$$\phi\left[\frac{a}{h}, \frac{p}{h^2.sps^2.\rho}, \frac{ph^3}{D}\right] = 0 \tag{21}$$

These dimensionless groups were similar to those derived using Rayleigh's method.

Results and discussion

Experimental data had previously been acquired at five different press operating speeds and three different feedstock thicknesses, giving a total of 15 data sets. The

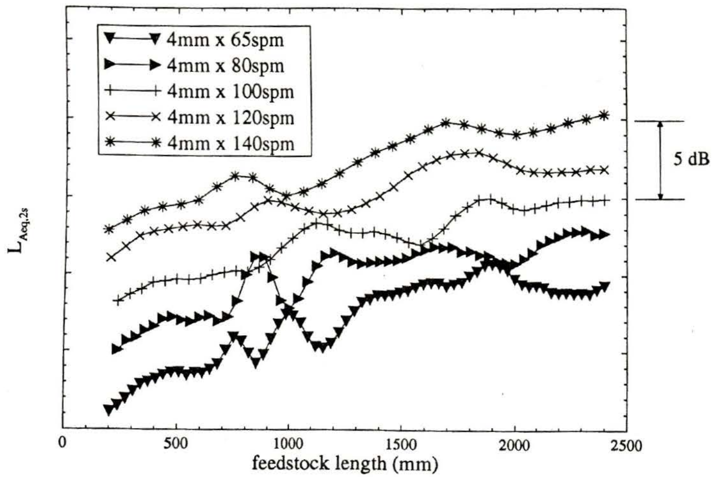

Figure 3 $L_{Aeq,2s}$ verses feedstock length for 4mm thick feedstock for the Emil Bender expanded-metal press.

information within each individual data set was obtained by taking a continuous 2 second duration $L_{Aeq,2s}$ for the entire 2400 mm sheet of feedstock. Figures 3 and 4 are plots of this raw data. In Figure 3 the feedstock thickness is held constant at 4 mm and $A_{eq,2s}$ is plotted against the feedstock length for various press operating speeds. In Figure 4 the press operating speed is held constant at 120 spm while the $L_{Aeq,2s}$ is plotted against feedstock length for various feedstock thicknesses. It is worth noting that Figure 3 is one in a *family* of three possible graphs while Figure 4 is one graph in a family of five possible graphs. When the 15 data sets are plotted in terms of the non-dimensional π_1 and π_2 groups as shown in Figure 5, they can be seen to collapse into three very distinct feedstock thickness families. It is evident that these curves obey a power law of the form:

$$\pi_2 = \pi_1^n \tag{22}$$

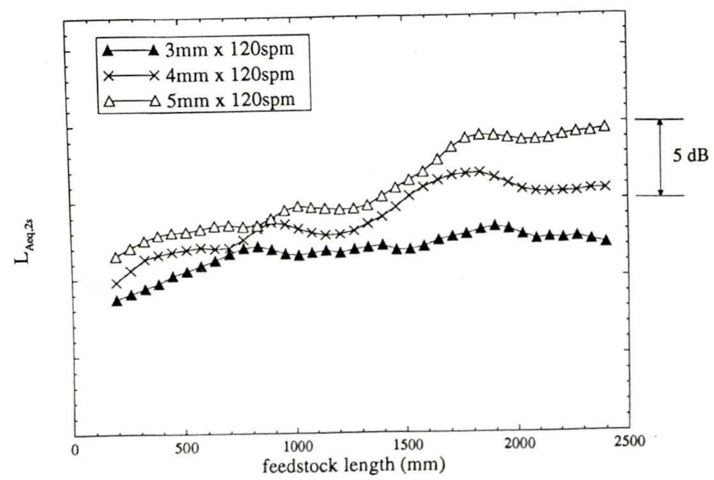

Figure 4: $L_{Aeq,2s}$ verses feedstock length for 120spm operating speed for the Emil Bender expanded-metal press.

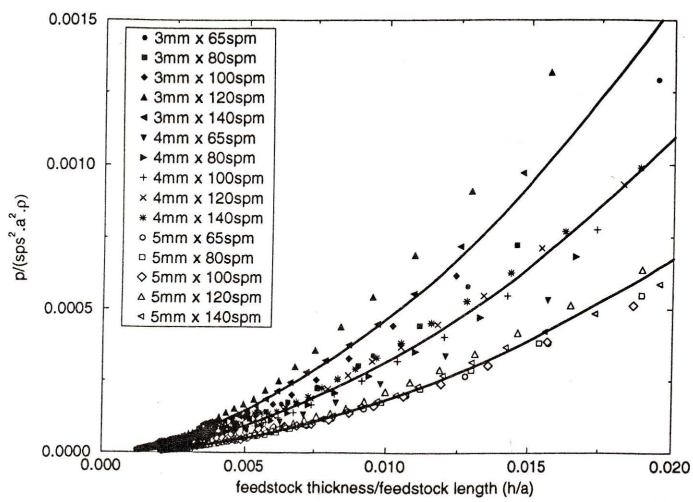

Figure 5: Non-dimensional π groups for the Emil Bender expanded-metal press.

If we assume that n is a function of the feedstock thickness h, then:

$$\pi_2 = \pi_1^{n.f(\pi_1)} \tag{23}$$

or

$$\log_{10} \pi_2 = n.f(\pi_1)\log_{10} \pi_1 \tag{24}$$

The data sets for the five press operating speeds for each feedstock thickness were then grouped and a non-linear curve fit applied to determine the functional relationship for n.f(π_1). Three values of n were thus obtained. These values of n were in turn plotted against h and a linear regression performed to obtain the slope and n-intercept values.

The results were then normalised to a reference feedstock thickness h_{ref} to convert them into a non-dimensional form. The following relationship was thus obtained:

$$\pi_1 = \left(\frac{a}{h}\right)^{(1.36+0.012(h/h_{ref}))} \tag{25}$$

where; a = feedstock length, h = feedstock thickness, h_{ref}= 5 mm feedstock.

The 15 data sets are then plotted using $\pi_1 = \left(\frac{a}{h}\right)^{(1.36+0.012(h/h_{ref}))}$ and $\pi_2 = p / h^2.sps^2.\rho$ As shown in Figure 6, the measured data are now reduced to a single and elegant relationship for the $L_{Aeq,2s}$ at varying operating speed, feedstock thickness and feedstock length. By reversing the process it is a simple matter to use the non-dimensional π groups to predict the sound pressure level for this press at any operating speed, feedstock length and/or feedstock thickness. Thus it is now possible through the use of these simple relationships to predict the relative noise levels based on limited measurements. The third dimensionless group $\pi_3 = ph^3/D$ was not used in the data reduction as we were unable to conduct experimental tests using different feedstock materials to test our predictions.

Conclusions

Dimensional analysis was used to establish three non-dimensional π groups for the acoustic radiation from an expanded-metal press, namely:

$$\frac{a}{h}, \frac{p}{h^2.sps^2.\rho}, \frac{ph^3}{D}$$

The experimental results were presented graphically in their dimensionless form. Within the experimental accuracy the complex experimental results have been successfully reduced into a single curve in terms of the first two dimensionless parameters. Further testing and analysis will be directed towards confirming the third π group if other feedstock materials are used by BHPBH under production conditions.

Acknowledgments

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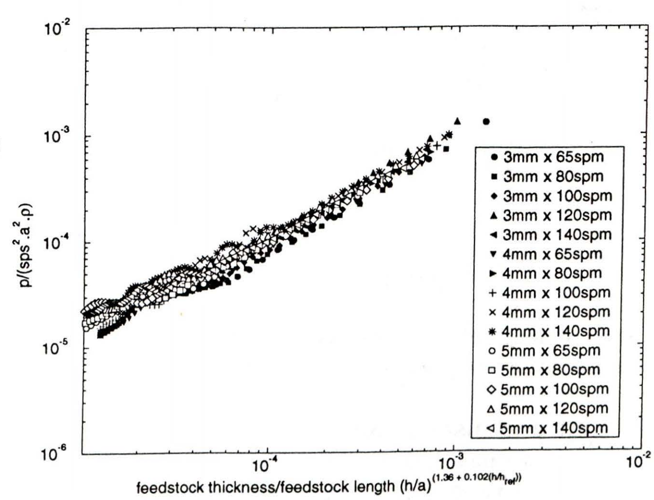

Figure 6: Amended non-dimensional π groups for the Emil Bender expanded-metal press (log-log).

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Feedback control ear defender

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Abstract

Unwanted acoustic noise can have adverse psychological effects on people exposed to noise environments. It can create an inefficient working place due to increased fatigue and loss of concentration and many workers suffer from temporary or permanent noise-induced hearing loss. In order to reduce the level of noise reaching the ear, passive protection devices are commonly used. However passive means is only limited to the higher frequencies. In the low frequency range such devices have poor sound attenuation characteristics. Since practical sound absorption materials have poor absorption characteristics at low frequencies, the use passive means becomes bulky and impractical. In order to improve the low frequency attenuation of passive earmuffs active control can be employed. In this paper the design and implementation of two analog feedback ear defenders will be presented to show their effectiveness compared to passive systems in low frequency attenuation. Moreover the feasibility of a tailor-made ear defender to suit a specific noise environment will be discussed . Finally the general performance of the prototypes will be presented and important considerations towards manufacturing outlined.

Introduction

Active Noise Control(ANC) was first proposed by Paul Leug in 1936 in his patent " Process of Silencing Sound Oscillations"[1]. It was the first time that the idea of active control was described and the principle of measuring the sound field with a microphone, manipulating the resulting signal electronically and then feeding it to an electroacoustic secondary source used. The next major work in the field occurred in the 1950's when Olson et al. explored the feasibility of active noise attenuation in a partial space around a man's head. In 1953 Olson and May[2] described an active noise control system which they called an "electronic sound absorber". While the work of Leug is characterised as feedforward control, that of Olson used the feedback principle to achieve control at the microphone location. Because of the limitations of electronics at the time no further work of practical importance was carried out until the 1970's[3,4]. Since then the application of active attenuation techniques to hearing protection has attracted a lot of attention and researchers have investigated both feedforward and feedback control strategies. Carme[5] proposed a filter to compensate the open loop of the electroacoustic system in a feedback strategy. Sha et al.[6] used feedforward control in an active ear defender. Both methods make use of the principle of superposition of waves to achieve cancellation at the ear.

From a practical point of view the process of attenuating sound consists of detection of the unwanted signal, inverting it and superposition of the primary and secondary control signals. The detector used, such as a microphone is required to have a flat response and linear phase characteristics. The controller carries out the inversion and adjusts the amplitude of each signal. After the signal is detected and processed by the controller the output is used to drive a secondary source, such as a loudspeaker. The various arrangements of the detector, controller and secondary source determines whether the control is feedforward or feedback. In the former the sensor outside the ear defender detects the unwanted noise and the signal fed through a controller and finally output to the ear by the secondary loudspeaker. Whereas in the latter the sensor is fitted inside the electroacoustic cavity itself and the detected signal is fed back to the secondary source via the controller. The main limitation of feedback is the compromise between good attenuation and good stability of the closed loop system. In feedforward control the main drawback is the variation of the primary transfer function which is the complex ratio of the uncontrolled signal at the ear to the primary signal outide the ear defender. The feedforward controller has to continually change to adapt to the changing conditions. In this paper the design and implementation of an analog feedback compensator will be described.

The compensated feedback principle

Figure 1 shows a dynamic circumaural headset fitted with a microphone M. Suppose that such a device is worn in a random noise environment of pressure P_o. Let the uncontrolled pressure sensed by the ear with the control circuit off be P_u and the pressure after control be P_c. In feedback control the signal picked up by the microphone is sent to the compensator C which in turn drives the loudspeaker L. The resulting electroacoustic sytem is assumed to be linear and can be represented by the block diagram of Figure 2(a) where H_m, H_l and H_c are the transfer functions of the microphone, loudspeaker and compensator respectively. H_a is the acous-

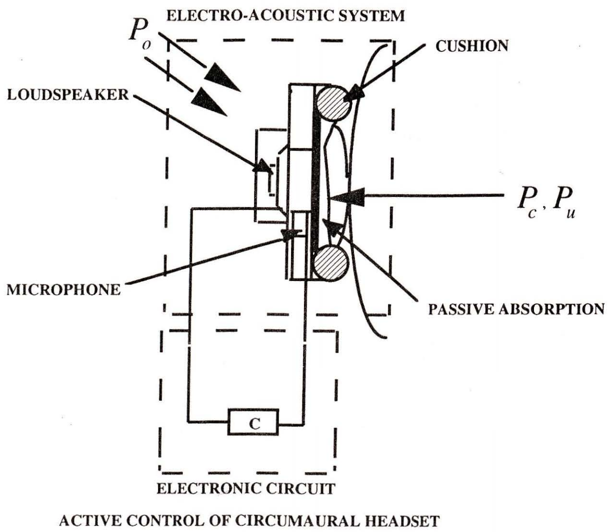

Fig. 1. Feedback ear defender arrangement.

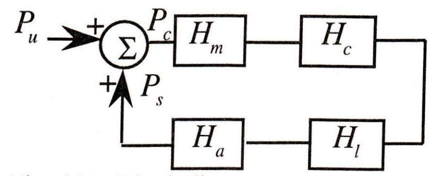

Fig. 2(a). Block diagram representation.

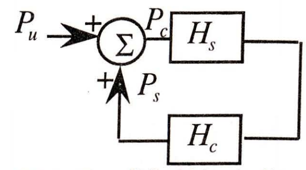

Fig. 2(b). Simplified block diagram.

tic cavity and absorption transfer function. For analytical purposes the block diagram of Figure 2(a) can be simplified into that of Figure 2(b) in which H_s is the secondary transfer function of the electroacoustic system which comprises of the cavity, loudspeaker and microphone. That is $H_s = H_m H_l H_a$. The open loop transfer function can be defined as $H_o = H_s H_c$.

The closed loop transfer function can therefore be written as

$$\frac{P_c}{P_u} = \frac{1}{(1 - H_o)} \tag{1}$$

Eq. (1) also represents the attenuation at the microphone location. An absolute value much less than unity is desirable and represents a reduction of sound level whereas a value greater than unity indicates an increase in level. The overall performance of the feedback system depends on the open loop response of the electroacoustic system and the purpose of H_c is to electronically alter the uncompensated open loop characteristics of the system. The main limitation to the feedback structure is the inherent stability problem which occurs when the denominator term of Eq. (1) vanishes, that is when

$$1 - H_o = 0 \tag{2}$$

If for some critical frequency the open loop magnitude is unity and the phase shift is 0^o, $\pm 360^o$, then the system becomes unstable in closed loop. In physical terms the pressure at the ear at the critical frequency becomes very large. For the system to stay stable in closed loop the magnitude of the total open loop response must be less than unity at the phase crossovers. It should be noted that just outside the band of attenuation there is always some increase of level instead of reduction because of the phase shift in the system. This can be explained if the magnitude of Eq. (1), say r, is set to unity. For a phase shift $\varnothing$ in the total open loop response it is necessary that

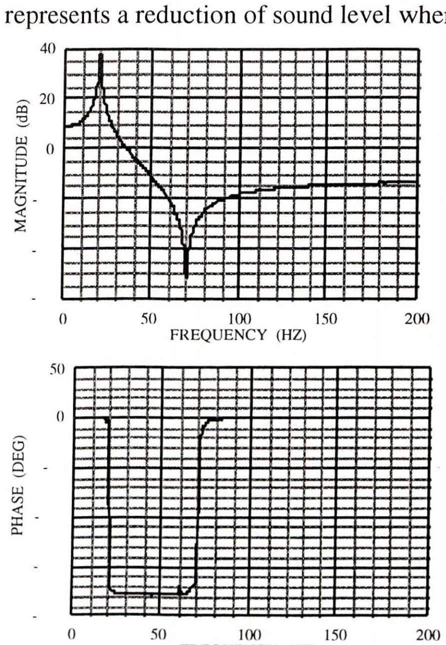

Fig. 3(a). Lightly damped compensator response.

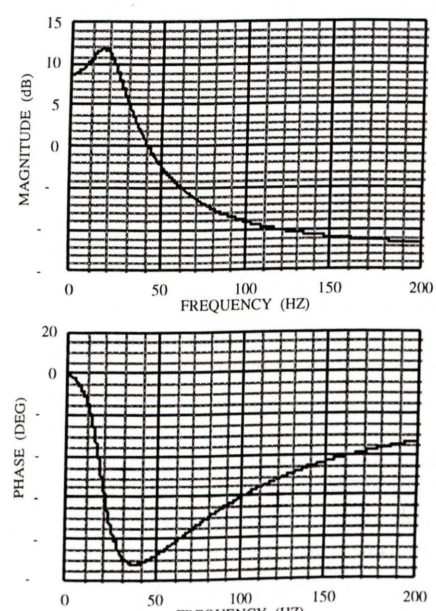

Fig. 3(b). Effect of increasing damping.

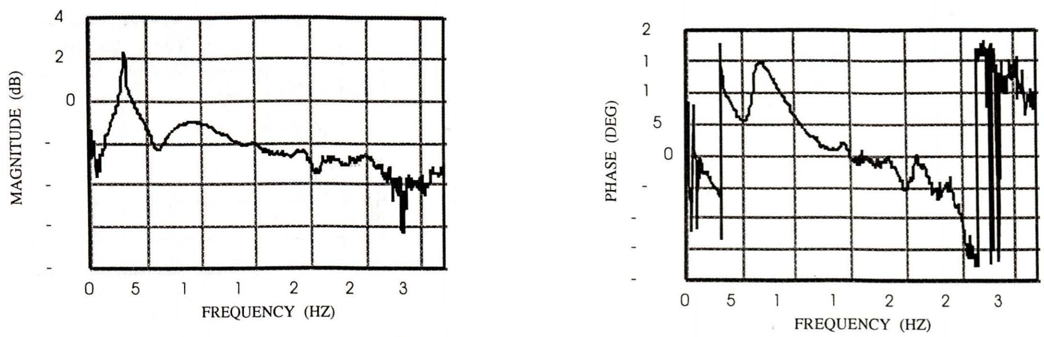

Fig.3(c). Total open loop response

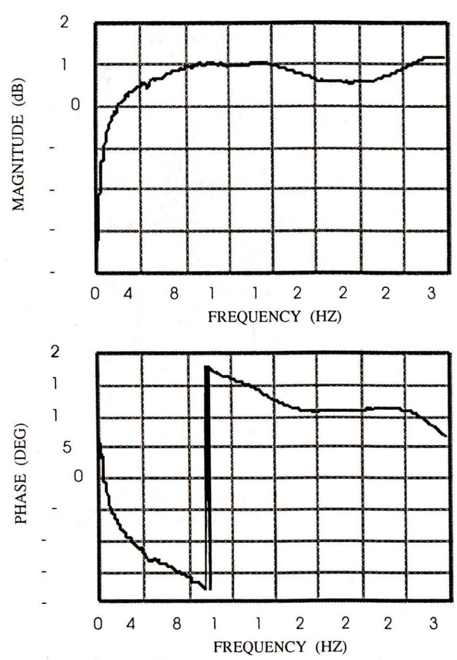

Fig. 4(a). Uncompensated open loop response of circumaural headset.

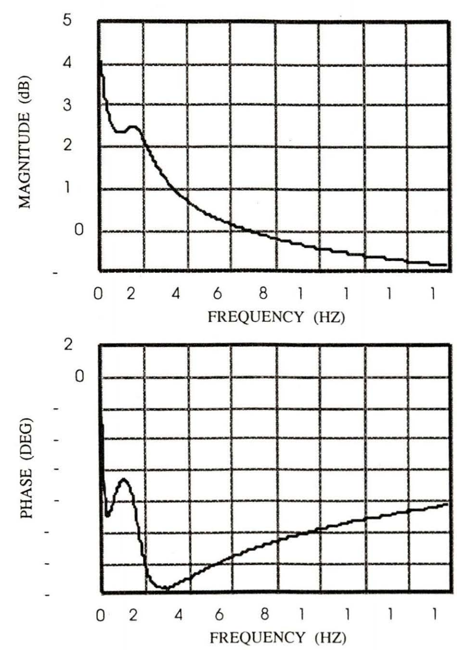

Fig. 4(b). Compensator frequency response

$$r = 2\cos\varnothing$$

$$\text{for } -90^{o} \leq \varnothing \leq 90^{o} \qquad (3a)$$

and

$$r = 0$$

$$\text{for } \varnothing \text{ otherwise} \qquad (3b)$$

If for a given phase shift $\varnothing$ in the compensated open loop system the magnitude r is less than that given by Eqs. (3) there will be an enhancement of sound level at the ear under stable operation.

Compensator design

The feedback compensator used in order to modify the open loop response is based on the placement of poles and zeroes. The transfer function of a biquadratic filter can be written as

$$H_c = \frac{(s^2 + 2\xi_z \omega_z s + \omega_z^2)}{(s^2 + 2\xi_p \omega_p s + \omega_p^2)} \qquad (4)$$

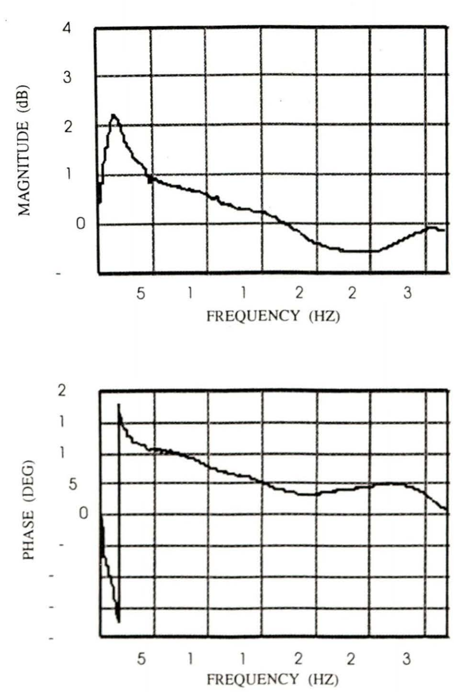

FIG. 4(c). Compensated open loop of system.

Where ω_z, ω_p are the zero and pole locations respectively and ξ_z, ξ_p are the associated damping coefficients. The experimental frequency response of such a filter with a lightly damped pole at 20 hz and zero at 70 hz is shown in figure 3(a). If the damping coefficients are increased the frequency response of figure 3(b) is obtained. The resulting filter is of minimum phase characteristics and suitable for feedback compensation[5].

Experimental results and discussion

Two headsets have been implemented with feedback control based on the biquadratic compensator of Eq. (4). In each case only two cascaded second order compensators have been used in order to keep the electronics simple. The system performance in terms of maximum attenuation, wide control bandwidth and good stability has been optimised numerically to achieve the desired end results. In the first instance a circumaural high impedance headset is employed to actively attenuate a band of low frequency noise. In the second instance a passive ear defender fitted with transducers is used to attenuate a narrow band. The peak attenuation obtained for such a system depends on the desired bandwidth of attenuation and vice versa. The wider the band in which control is desired the lesser will be the peak attenuation, and the narrower the control band the higher the peak reduction at the ear. For the circumaural headset of Figure 1, the uncompensated open loop frequency response is as shown in Figure 4(a) below. When two cascaded biquadratic filters of Figure 4(b) are added into the loop the total response becomes that of Figure 4(c). The gain of the open loop of this active ear defender is limited by the phase cross over in the low frequencies. For good system stability a gain margin of 6 dB and phase margin of 45^{0} has been allowed for.

The attenuation of the headset was obtained by measuring the power spectrum at the microphone without and with control from an octave band generator of center frequency 125 Hz. The results are as shown in Figure 5. The band of attenuation is from 30 Hz to above 1500 Hz with a peak of above 20 dB at 180 Hz. A slight increase in level below 30 Hz is a direct result of Eq. (3) and can be minimised by ensuring that the magnitude of the open loop response is much less than unity in this region.

Passive ear defenders are widely used in industries but are deficient in the attenuation of low frequency noise. The problem can be reduced by using a combination of active and passive control as described previously. If however the need arises for the attenuation of a specific narrow band noise then broad band attenuation is not necessary and efforts can be concentrated in a narrow band compensator design. Consider the physical arrangement shown in Figure 6. Suppose that it is desired to obtain a narrow-band control centered around 315 Hz. For this system the measured uncompensated response is shown is Figure 7(a). If the proposed compensator of Figure 7(b) is added into the loop the modified open loop response is as in Figure 7(c). The desirable high

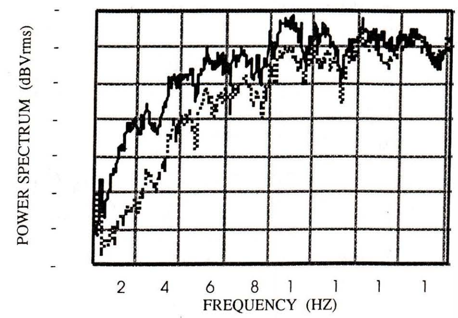

FIG. 5. Power spectrum before(solid curve) and after control(dotted curve).

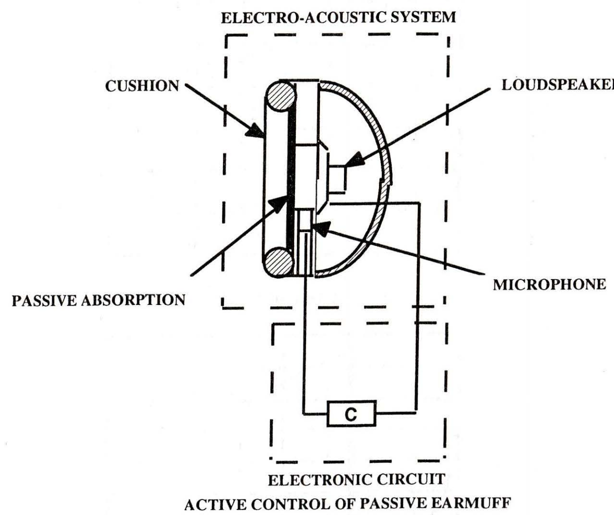

FIG. 6. Physical arrangement of passive earmuff fitted with active control.

gain at 315 Hz is clear while the magnitude drops off very rapidly on either side. The power spectrum of a random noise measured without and with control are given in Figure 8. The narrow-band attenuation of the system is as expected with a peak of about 27 dB at 315 Hz. The pure tone attenuation achieved at 300,315 and 330 Hz are 10, 27 and 14 dB respectively.

Development of a prototype printed circuit board

Once the prototype compensator has been designed and tested the miniaturisation of the control circuit can be envisaged. There are several key factors that have to be taken into account. The main consideration is cost. This in turn determines the selection of amplifiers and other components which are critical to the overall system performance. The second most important criterion is the design of the power supply. If the system is battery operated then a choice of powering the unit directly from batteries or via a dc to dc converter exists. This again is linked with cost as well as portability and performance and a compromise has to be made by the designer. The prototype developed at the University of Western Australia makes use of two sandwiched 60 x 100 mm printed circuit boards using surface mount components and is powered by 4 x 1.5 V AA batteries via a dc to dc converter.

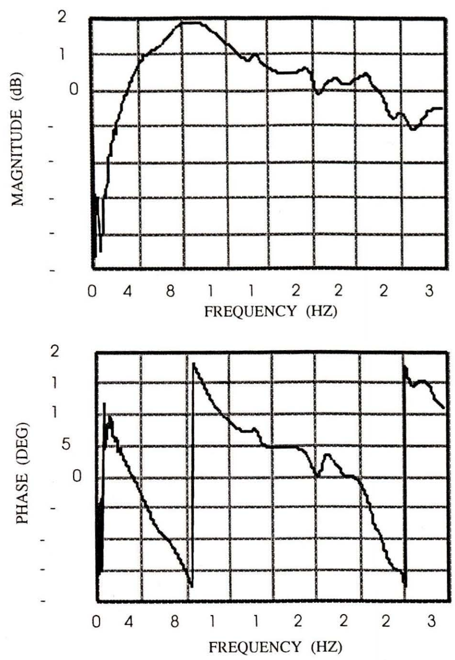

Fig. 7(a). Uncompensated response of passive muff.

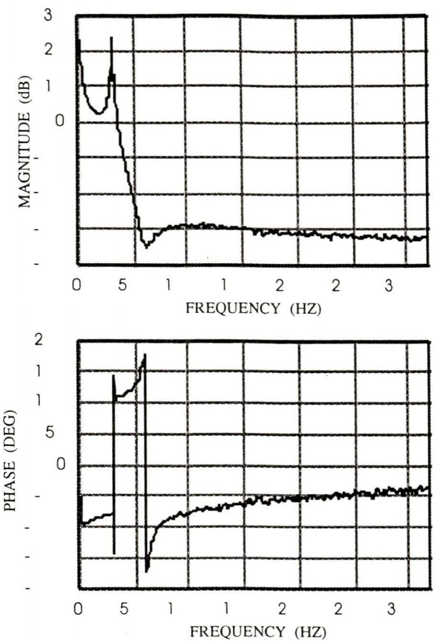

Fig. 7(b). Compensator frequency response.

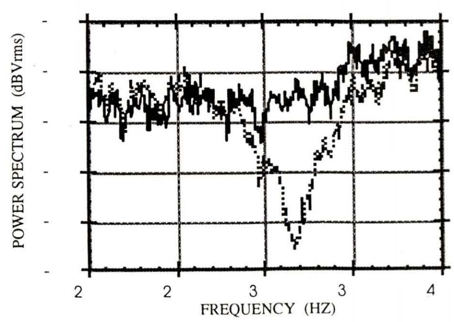

FIG. 8. Power spectrum without(solid curve) and with control(dotted curve) over a narrow band.

Conclusion

Feedback control has successfully been used to improve the low frequency attenuation of passive systems. The design method outlined also allows a particular narrow band controller design which leads to the possibility of tailor-made ear defenders for a specific noise spectrum. In this way a larger attenuation of the offending narrow band frequencies can be achieved compared to existing products. These selective ear defenders would find suitable application in the minerals and energy resource industries. Finally an end product for manufacturing will have to take into account cost, performance and portability.

Acknowledgment

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Measurement Of Four-Pole Parameters Using Direct Forces

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Abstract

Four-pole parameters are used to provide a full characterisation of isolator performance which is independent of the foundation system upon which the isolator is located. To date the standard test developed to measure the four-pole parameters of a pre-loaded isolator has involved the use of indirect force measurements on the isolator's input and output. The limitations of using this approach are analysed, and an improved technique using direct force measurements is proposed. Experimental results for isolator properties are presented.

Four-pole parameters

Knowledge of the dynamic properties of vibration isolators, machine mounting locations and foundations are needed to be able to estimate the vibration power transmitted through the isolators and to predict the effect of mounting a vibrating source on flexible vibration isolators.

In the design and selection of vibration isolation mounts it is important to have a description of the mount behaviour which is independent of the test arrangement and the foundation system upon which they are to be installed. The dynamic behaviour of anti-vibration mounts may be characterised by their four-pole parameters, (Molloy 1957, Snowdon 1976, Snowdon 1979, Verheij 1982 and Dickens et al 1993), which provide one such independent description.

The four-pole parameters relate the force F_1 and velocity V_1 at the isolator input to the force F_2 and velocity V_2 on the isolator output:

$$\begin{bmatrix} F_1 \\ V_1 \end{bmatrix} = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \begin{bmatrix} F_2 \\ V_2 \end{bmatrix} \qquad (1)$$

where A, B, C, and D are the four-pole parameters, and are complex, time invariant functions of ω.

Applying Maxwell's law of reciprocal deflections to the isolator means that the transfer impedance or mobility between the input and the output is independent of which end is treated as the input or output. This leads to the relationship:

$$AD - BC = 1 \qquad (2)$$

For the case of symmetric isolators i.e. those that behave the same if the input and output ports are interchanged, then the additional relation is applicable:

$$A = D \qquad (3)$$

Equations (2) and (3) mean that only two independent four-pole parameters need to be measured for a symmetric isolator in order to completely characterise it. At lower frequencies an isolator may be assumed to be a massless spring of stiffness k, in which case $A = D = 1$, $B = 0$ and $C = 1/j\omega k$, where $j = \sqrt{-1}$ and ω is the circular frequency.

The common measure of isolator performance in an installed situation is the effectiveness, E, which is defined as the ratio between the foundation velocity (or force) obtained when the source and foundation are directly connected and the foundation velocity (or force) obtained when the source and foundation are connected via the isolator. In terms of the four-pole parameters and the source and foundation mobilities:

$$E = \frac{AH_s + BH_sH_f + C + DH_f}{H_s + H_f} \qquad (4)$$

where H_s and H_f are the mobilities of the source mounting point and the foundation mounting point respectively.

Alternatively the insertion loss, L, is defined as the magnitude of E expressed in dB:

$$L = 20 \log_{10} \left| \frac{AH_s + BH_sH_f + C + DH_f}{H_s + H_f} \right| \qquad (5)$$

Experimental measurements

In general the dynamic properties of a vibration isolator are dependent upon its pre-load, and so measurements should be made with a pre-load similar to that experienced in service.

One possibility of determining two independent four-pole parameters is given by considering the special case in which the output side is blocked, i.e. $V_2 = 0$, which in equation (1) yields:

$$A = \left.\frac{F_1}{F_2}\right|_{V_2=0} \quad \text{and} \quad C = \left.\frac{V_1}{F_2}\right|_{V_2=0} \qquad (6a, b)$$

This allows a pre-load to be applied to the input side of the isolator and the isolator's output reacted against the blocked support. Measurements would then need to be made of the input force and velocity and the output force required to hold the isolator blocked.

Verheij (1982) describes a method for making measurements of the blocked transfer function of a resilient isolator, defined as the force out divided by the input acceleration. Dickens et al. (1993) used a similar method at the Aeronautical and Maritime Research Laboratory to measure the pre-loaded, blocked four-pole parameters. The assumption of the blocked mass introduces a small, but in some cases still significant error, and limits measurements to symmetric isolators.

Dickens and Norwood (1994), proposed an alternate measurement scheme which uses a floating mass. This scheme corrects the four-pole parameters for the small but finite output velocity of the blocking mass. It also allows the measurement of the four-pole parameters of unsymmetrical isolators to be made by using two floating masses. Both measurement methods use inferred forces at both the input and output sides of the isolator. A schematic of the measurement set-ups is shown in Figure 1.

As defined above, F_1 is the force applied to the input of the isolator, and F_2 is the force exerted by the output of the isolator on the blocking mass m_2. Let F_a and F_b be the sinusoidal forces exerted on the excitation mass m_1 by the two vibrators at a circular frequency of ω. A_1 and V_1 are the acceleration and velocity at the excitation mass / test isolator interface, and A_2 and V_2 are the acceleration and velocity at the test isolator / blocking mass interface. Denoting the inferred forces at the input and output as F_{1I} and F_{2I} respectively, then

$$F_{1I} = F_{in} - m_1 A_I \quad \text{and} \quad F_{2I} = m_2 A_2 \qquad (7a, b)$$

where $F_{in} = F_a + F_b$

While the use of the inferred forces is experimentally attractive because of its relative simplicity and ease of application, there are errors introduced. This paper analyses these errors and proposes the use of direct force measurements as a means of eliminating them. For the purposes of these analyses the masses are considered to be rigid bodies and the lower mass is treated as a blocking mass.

Analysis of equivalent mass spring system

The test arrangement may be modelled by the equivalent spring mass system shown in Fig 2, where m_1 represents the excitation mass, m_2 the blocking mass, k_1 the airbag through which the pre-load is applied, k_2 the isolator, and k_3 the supporting airbags. The displacements of the

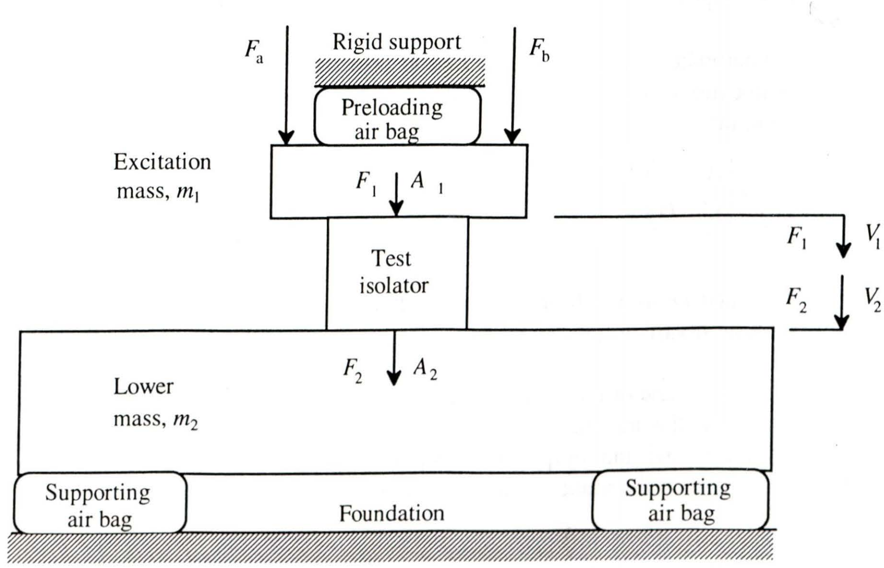

Figure 1: Schematic test arrangement for usual method

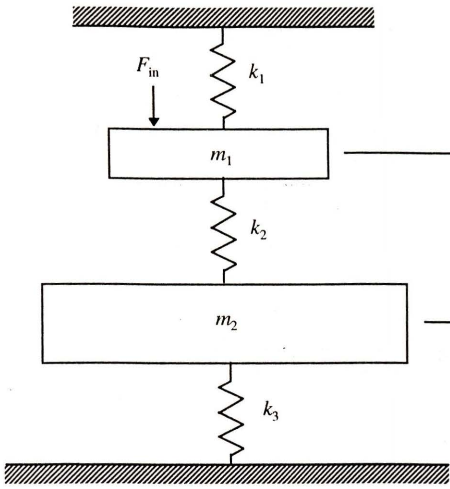

Figure 2: Model of test arrangement

excitation and blocking masses are X_1 and X_2 respectively.

The equations of motion for the two masses are:

$$m_1\ddot{X}_1 = F_{in} - k_1X_1 - k_2(X_1 - X_2) \quad (8a)$$

$$m_2\ddot{X}_2 = k_2(X_1 - X_2) - k_3X_2 \quad (8b)$$

Let the harmonic forcing function be represented in complex form as $F_{in}(t) = F_0e^{j\omega t}$.

Substituting equations (8a) and (8b) into (7a) and (7b) gives:

$$F_{1I} = k_1X_1 + k_2(X_1 - X_2) \quad \text{and} \quad F_{2I} = k_2(X_1 - X_2) - k_3X_2 \quad (9a, b)$$

By assuming the particular solution $X_1(t) = X_{10}e^{j\omega t}$ and $X_2(t) = X_{20}e^{j\omega t}$, we obtain from equations (8a) and (8b):

$$X_1(t) = X_{10}e^{j\omega t} = \frac{F_0\left(k_2 + k_3 - m_2\omega^2\right)e^{j\omega t}}{\left(k_1 + k_2 - m_1\omega^2\right)\left(k_2 + k_3 - m_2\omega^2\right) - k_2^2} \quad (10a)$$

and

$$X_2(t) = X_{20}e^{j\omega t} = \frac{F_0k_2e^{j\omega t}}{\left(k_1 + k_2 - m_1\omega^2\right)\left(k_2 + k_3 - m_2\omega^2\right) - k_2^2} \quad (10b)$$

Denoting the direct forces at the input and output of the isolator as F_{1D} and F_{2D} respectively, then from Figure 2 we obtain:

$$F_{1D} = F_{2D} = (X_1 - X_2)k_2 \quad (11)$$

The error E_1 is defined as:

$$E_1 = \frac{F_{1D} - F_{1I}}{F_{1D}} \quad (12)$$

Substituting for F_{1D} and F_{1I} gives:

$$E_1 = \frac{k_1}{k_2}\left[\frac{X_2}{X_1} - 1\right]^{-1} \quad (13a)$$

and substituting for X_1 and X_2 from equations (9a) and (9b):

$$E_1 = \frac{k_1}{k_2}\left[\frac{k_2 + k_3 - m_2\omega^2}{m_2\omega^2 - k_3}\right] \quad (13b)$$

Similarly, at the output side of the isolator:

$$E_2 = \frac{F_{2D} - F_{2I}}{F_{2D}} \quad (14)$$

giving:

$$E_2 = \frac{k_3}{k_2}\left[\frac{X_1}{X_2} - 1\right]^{-1} \quad \text{or} \quad E_2 = \frac{k_3}{k_3 - m_2\omega^2} \quad (15a, b)$$

Equations (13b) and (15b) may be written as:

$$E_1 = \frac{k_1}{k_2}\left[\frac{1 - (\omega_2/\omega)^2}{(\omega_1/\omega)^2 - 1}\right] \quad \text{and} \quad E_2 = \frac{-1}{1 - (\omega/\omega_1)^2} \quad (16a, b)$$

where $\omega_1 = \sqrt{\dfrac{k_3}{m_2}}$ and $\omega_2 = \sqrt{\dfrac{k_2 + k_3}{m_2}}$

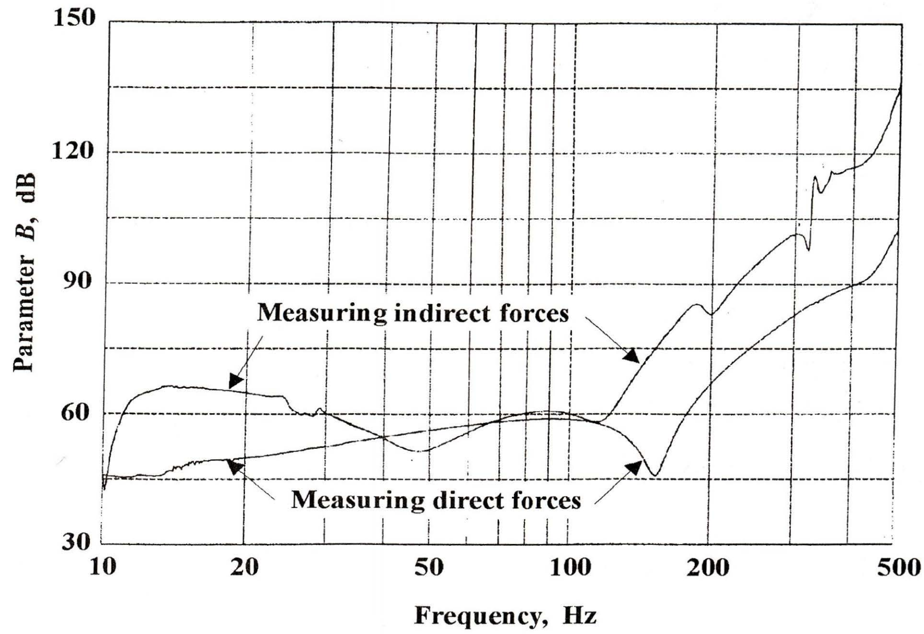

Figure 3: Four pole parameter B.

The various expressions for the error E_1 show that it is dependent upon the stiffness of the isolator under test, the stiffness of the spring element above the excitation mass, which is used to provide the pre-load, the mass of the blocking mass and the frequency. If $X_1 >> X_2$ i.e. $\omega >> \omega_1$ then $E_1 \cong -k_1/k_2$ and will be constant for the elements k_1 and k_2 having a constant ratio of stiffnesses. The magnitude of E_1 asymptotes towards the value of $-k_1/k_2$ as ω increases above ω_2.

The various expressions for the error E_2 show that it is dependent upon the stiffness of the supporting spring element below the blocking mass, the mass of the blocking mass and the frequency. If $X_1 >> X_2$ i.e. $\omega >> \omega_2$ then $E_2 \cong 0$. As ω increases above ω_2 the error E_2 asymptotes towards zero.

Experimental results and discussion

An experiment was conducted to measure both the direct and indirect force inputs and outputs for an isolator and to compare the four-pole parameters calculated from these forces and the measured error to that predicted by equations (16). The isolator used had a nominal dynamic stiffness of 5.3 x 10^5 N/m, and was mounted on a 576 kg blocking mass which was in turn mounted on four air-bags with a total nominal stiffness of 3.2 x 10^5 N/m. A

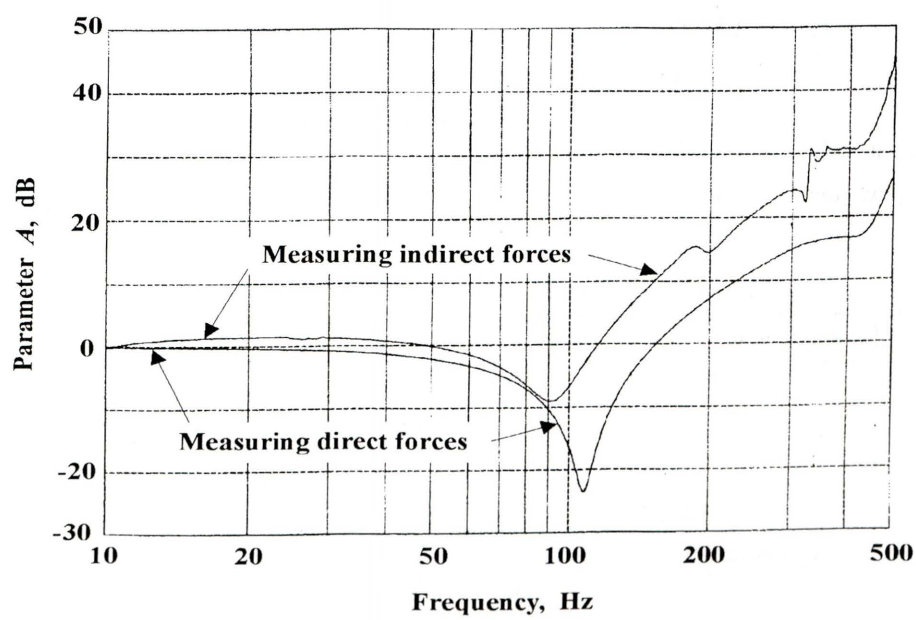

Figure 4: Four pole parameter A.

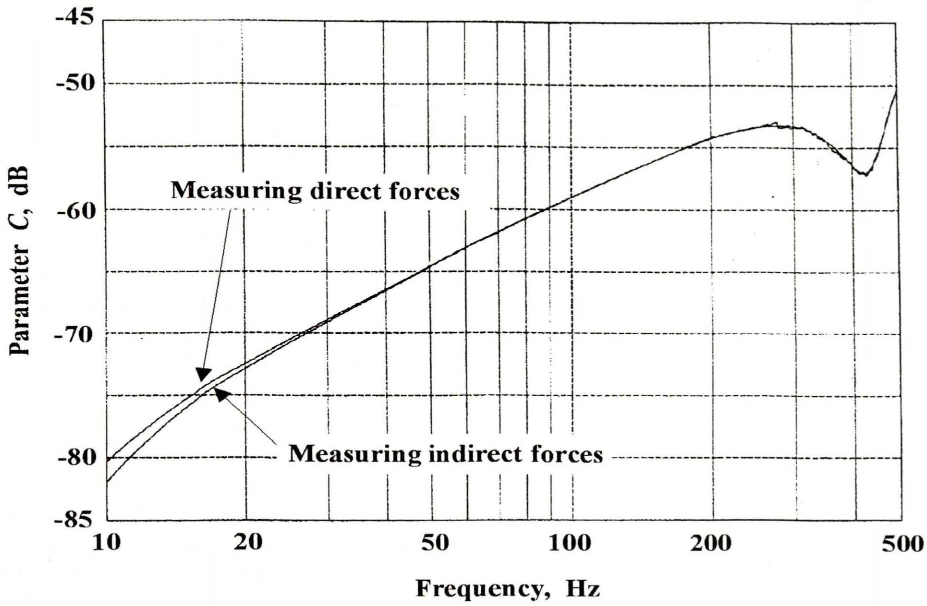

Figure 5: Four pole parameter C.

54.6 kg excitation mass was used and the pre-load was provided by an airbag of nominal stiffness 8.9 x 10^4 N/m. The isolator was pre-loaded to its rated service load and the dynamic excitation was provided by a pair of Gearing and Watson shakers for which each shaker had a rated output of 150 N.

The direct force inputs from the excitation mass to the isolator, and from the isolator to the blocking mass were measured using three matched Bruel and Kjaer 8200 force transducers mounted between two ground plates at each location. The charge outputs from the three transducers in each set were summed before signal conditioning. The total input force from the shakers was measured by summing the charge outputs from two matched Bruel and Kjaer 8200 force transducers. These force transducers were mounted between the driving stingers of the shakers and the excitation mass. The acceleration of the excitation and blocking masses were measured using a pair of Bruel and Kjaer 4379 accelerometers mounted on each. Bruel and Kjaer 2635 charge amplifiers were used to condition each of the signals and the data was collected using a sixteen channel Hewlett Packard 3566A spectrum/network analyser. The input force to the excitation mass from the shakers was approximately constant, which resulted in the input displacement to the isolator, X_1, being approximately proportional to $1/\omega^2$.

The four-pole parameters were calculated from both the direct and the indirect force measurements using the blocked output assumption, Figures 3 to 5. In calculating the indirect forces, the mass contributions of the masses, transducers, attachments, cables, ground plates, and air bags were included. The mass contribution from each air bag was considered to be one half of its total mass. In calculating the direct forces, the inertial force contributions of the ground plates, transducers and cables were subtracted to give the actual forces at the isolator's input and output. The error in the inferred output force is much less than for the inferred input force.

The parameter *C* shows little variation for the two sets of results, while parameters *A* and *B* show significant differences. This is because of the inputs used to determine the parameters. From equation (1) the four-pole parameter equation involving *C* does not contain the input force, and if the output force is blocked, then *C* is given by equation (6b). Parameter *A* is given by equation (6a) for the blocked assumption, and will be very dependent on the accuracy of the input force determination, as will *B*, which is derived from *C* and *A*.

Equation (16a) predicts that the error E_1 between the two input force measurements will be constant, provided that the stiffness ratio of the pre-loading spring above the excitation mass and the isolator is constant, and based upon the assumption that the airbag and isolator are massless. The dynamic stiffnesses of the pre-loading airbag and the isolator were determined, so that the predictions from Equation (16a) could be compared to the actual error calculated from the direct and indirect force measurements. The direct force measuring assembly used to measure the input force was placed between the air bag and the excitation mass and the experiment was repeated using the same pre-load. The displacement across the air bag was calculated from a pair of Bruel and Kjaer 4379 accelerometers mounted at each end of the air bag. The stiffness of the air bag was then calculated by dividing the applied force, after correcting for the added masses, by the displacement. The stiffness of the isolator was determined in a similar fashion.

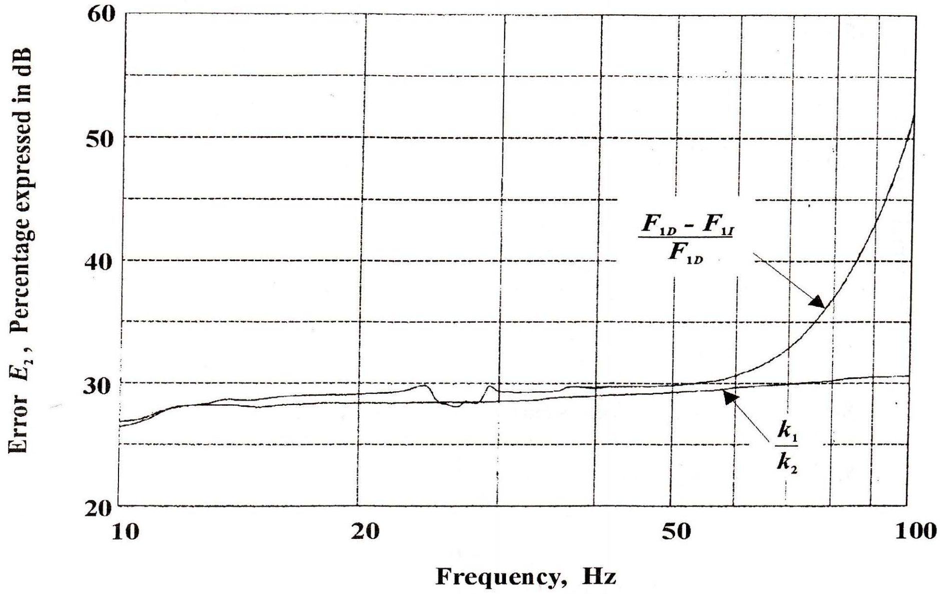

Figure 6
Error E_1 calculated from the indirect and direct force measurements, and the ratio of the stiffness.

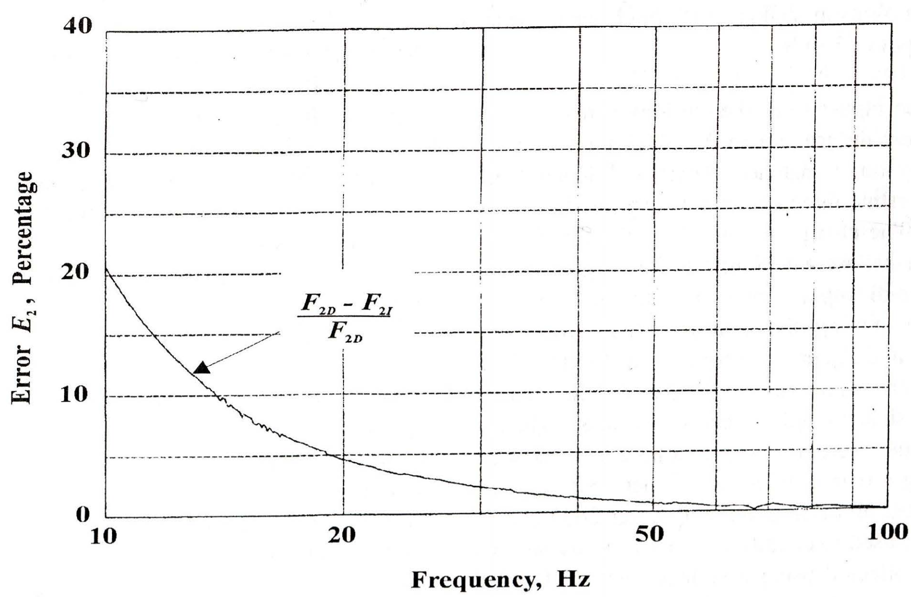

Figure 7
Error E_2 calculated from the indirect and direct force measurements.

Figure 6 shows a comparison of the error E_1 calculated from the indirect and direct force measurements, with the error calculated from equation (16a). These two curves show good agreement below 50 Hz, but above this frequency the two curves differ substantially. The curve for parameter A shows that it is approximately equal to unity up to about 50 Hz, indicating that the isolator behaves as a massless spring up to this frequency. However, above 50 Hz the mass effects within the isolator become significant, so that the simplified model used to give the error prediction in equation (16a) is only valid up to this frequency. Additionally below 50 Hz the error is not constant, indicating that the ratio of the two stiffnesses varies.

The error E_2 was calculated from the indirect and direct output force measurements using equation (14) and is shown in Figure 7. This curve is in good agreement with the predicted behaviour given by equation (16b), with E_2 approaching zero as the frequency increases. The agreement also indicates that any mass effects from the airbags are much less significant than those of the isolator, since the airbag mass is much smaller than the mass of the blocking mass.

Verheij (1982) was primarily concerned with measuring the blocked transfer function of the isolator which is defined as the output force/input acceleration, and is related to the C parameter. The results derived in this way are not significantly affected by the error in the input force, and are mainly effected by the errors in the blocked mass assumption, Dickens and Norwood (1994). At lower frequencies it is sufficient to use this to describe the isolator performance, but as the frequency increases the other four-pole parameters become important in the determination of the isolator effectiveness.

The simplified system shown in Figure 2 has two natural frequencies. In the first mode the excitation mass and the blocking mass move in phase, while in the second mode the two masses move out of phase. The lower frequency limit of previous measurements made with the inferred force method was controlled by the second, out of phase, mode. An important aspect of the direct force method is that the lower frequency limit of the measurements is given by the frequency ω_2, which is less than the out of phase modal frequency. Thus the lower frequency range of the measurements can be extended by using the direct force measurement.

Conclusions

The errors involved with the use of the inferred input and output forces to determine the four-pole parameters of an isolator have been investigated and the limitations of the method are discussed. An alternate measurement procedure, using direct force measurements is proposed, in order to overcome the problems associated with the inferred forces technique. The results show that there are significant errors in the inferred input force method, particularly at lower frequencies, but which may extend to all frequencies, and which result in large errors in the determination of the four-pole parameters. While the use of direct force measurement poses some experimental difficulties, it provides a far more accurate determination of the input force to the isolator and hence the four-pole parameters. The use of direct force measurements permits the determination of the four-pole parameters over a wider frequency range.

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Session 1B

Architectural Acoustics

A Little Acoustics History : Spaces Used For Speech

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Abstract

This paper takes a brief look at the history of acoustics, with emphasis on spaces used for speech. It briefly traces the science of sound from ancient Greece and Pythagoras through to current practice, with the major emphasis in the period 1895 to present day. The paper also looks at sound decay measurement and presents some indices used for the rating of spaces where speech is the predominate use.

History

The scientific study of acoustics, particularly theatre acoustics, can be traced back to the ancient Greek amphitheatres such as Epidauros, Hunt (1978 p9) places the origins of the study of sound to Pythagoras (ca. 570-497 BC). Later in history Roman architects were shown to have developed an understanding of theatre acoustics (open air amphitheatres) as illustrated by the Roman architect Vitruvius in his *De Architectura* books (*Liber V, chapter VII: De locis consonantibus ad theatra eligendis*):

"We must chose a site in which the voice may fall smoothly, and not be returned by reflection so as to convey an indistinct meaning to the ear. For there are some places which naturally hinder the passage of the voice, ... those (dissonant) places in which the voice, when first it rises upwards, strikes against solid bodies above, and is reflected, interfering as it settles down with the rise of the utterance ... those (circumsonant) in which the voice moves around, is then collected in the middle where it dissolves without the case-endings being heard, and ides away I sounds of indistinct meaning ... those (resonant) in which the words, striking against a solid body, give rise to echoes and make the case-endings sound double ... those (consonant) in which the voice reinforced from below rises with greater fullness, and reaches the ear with clear and eloquent accents. Thus if careful observation is exercised in the choice of sites, such skill will be rewarded by the improved effect of the actors voices. ... Whoever uses these rules, will be successful in building theatres." *(Hunt 1978, p34)*

Note: the modern equivalent for the terms of: "dissonant" is interference; "circumsonance" is reverberation; "resonance" is echo and "consonance" is the process where a sound is strengthened by reflection. (Sabine 1964, pp163,187)

From these early beginnings, the Greek and later Roman amphitheatres were concerned with projecting the voice of the actors to the audience (Jordan 1980, pp22-23). The actors used masks to both enhance facial expression and to amplify their voices (Jordan 1980, p28; Sabine 1964, pp190-192). Seating was raked and tiered; the audience kept as close as possible to the stage area. Not all of these early theatres exhibited good acoustics. Vitruvius proposed the use of resonant vases placed in strategic positions in the audience. This is reported by Sabine (1964, p192) as evidence of the lack of *consonance* in the contemporary Greek theatres.

As the overall noise levels in the markets grew, the need arose to provide some protection to the theatre (Parkin and Humphreys 1958, p10), though this was not the only reason to position walls and buildings behind the stage for the actors. Undoubtedly they did provide some improvement in the distribution of the sound, as the walls of the building added reflections from the back to the direct sound projected from the actors. These buildings positioned behind the stage grew to incorporate various rooms and features. They became part of the theatre providing alternative views, for example the Roman amphitheatre of Orange in France (Jordan 1980, pp24-25).

The transition from open air theatres to totally enclosed theatres is seen in the Renaissance Teatro Olympico theatre in Vicenza, Italy (Jordan 1980, pp25-26). The plan of the theatre is typical Roman amphitheatre with the major exception being, it is totally enclosed. Another theatre built not long after was the Teatro Farnese in Parma. Here the stage became an elongated shape with what was to develop in a proscenium linking a more traditional stage area and the audience area (Jordan 1980, pp26-28).

History shows that as the music form developed, so did the shape and styles of the auditoria. With the development of the operatic music form, came the need to reduce the reverberation in these auditoria for better understanding of the words. A noteworthy example is Wagners Festspielhaus in Bayreuth. This operatic theatre deviated from the accepted theatre design of the time. The audience is seated in a fan shaped auditorium, with very large diffusion walls consisting of pillars and niches. The seating is predominantly at floor level which has a steep rake (Jordan 1980, pp32-33).

The development of the music form has out stripped the development of the theatre. There are differing needs for classical music, opera and symphony. None of these requirements, such as reverberation, articulation, spatial effect, come close to the requirements of the late twentieth century amplified music forms.

To cater for the huge patronage of some pop groups, we now have almost come the complete circle with the outdoor concerts of recent times. Modern groups would be well suited to playing in the ancient Greek amphitheatres with their raked seating. The amphitheatre environment is well suited to this form of music, as it allows for large audiences, situated at considerable distances from the stage, with the visual images projected onto giant video screens, gross amplification can deliver the sound to all patrons for kilometres around. Background noise levels are rarely a problem. The major problem in this environment is how can the sound be restricted to the site!

Auditoriums

Considerable research has been carried out in the field of auditorium acoustics during the past decades. It would seem that the results of this effort have resulted in auditoriums many feel were only moderately successful. The *good acoustics is rather a matter of empirical experience and also of good luck than systematic and science-based design.* (Kuttruff 1994, p27)

A major turning point in the understanding of room acoustics came with Sabine's research in 1898. In 1895, Sabine was instructed, by the Corporation of Harvard University:

> "... to propose changes for remedying the acoustical difficulties in the lecture-room of the Fogg Art Museum, a building that had just been completed." *(Sabine 1964, p3)*

It is interesting to consider, that this research was prompted by the lack of intelligibility in the Fogg theatre - speech intelligibility problem, not a music appreciation problem. Three years of research led Sabine to the formulate the Reverberation Time Equation, which links room volume, frequency and absorption. (Beranek 1994, p4)

The reverberation time equation is:

$$T_{60} = 0.163\frac{V}{A_f}\text{sec} \qquad (1)$$

where T_{60} the time in seconds it takes a sound to decay by 1 millionth (60 dB) of its level

V the volume in cubic metres

A_f the total sound absorption at a given frequency in m² sabins

This formula has a number of limitations to its applicability and neglects room modes, the influence of the shape of the enclosure and placement of acoustic absorption within the enclosure. It is based on the theoretical assumption that the sound waves hit each of the boundary surfaces in succession (Beranek 1992, p3); that the sound in the enclosure is reasonable reverberant; the acoustic absorption is uniformly distributed; the sound field is diffuse[1]; and that the sound can be propagated in any direction with equal probably (Ginn, 1978 p38; Northwood, 1977 p116). This is not the case in speech auditoria. In fact the unaided voice is very directional and the major source is located in only one area of the auditorium. Generally the stage or source area of the auditorium is reflective (acoustically), the rear wall is acoustically absorbent, seating/audience is absorbent and the ceiling is reflective. Additionally when reflectors are utilised to increase the overall signal level present to the listener the sound has significant directionality thus reducing the diffuseness of the sound field.

With the aid of formulas such as Sabine (1964) and others[1], acousticians are able to predict, with increasing accuracy, the expected reverberation time for spaces. Reverberation is important in the acoustic performance of spaces as Beranek outlines:

> "During continuously flowing music, listeners hear the first 10 or so decibels of the sound decay. If this early reverberation time is long enough, each note is prolonged and the music takes on a singing tone. When the music stops abruptly, the listeners hear 35 or more decibels of the decay in the quiet interval. This longer reverberation adds both fullness of tone and loudness and gives the listener a sense of being enveloped by the sound." *(Beranek 1994, p4)*

Reverberation also has significant effect on the intelligibility of speech. As the reverberation increases the intelligibility decreases. This is readily experienced when two conversations are conducted in spaces at the opposite ends of the reverberation spectrum, eg. An anechoic and reverberation chamber. In the anechoic chamber, the lack of reverberation is evidenced by the reduced apparent power of the voice. (Loudness is reduced in the face to face mode). In the reverberation chamber, with the abundance of reflections, reverberation, has a dramatic and obvious impact on the intelligibility of the speech.

There was considerable research and development activity, to established optimum reverberation times in the period from 1900 to 1950, for various facilities of differing size and purpose. This research was centred on spaces used for music and concert halls. At mid frequencies (500-1000 Hz) the reverberation time, according to Beranek (1994, p5), for Baroque music is approximately

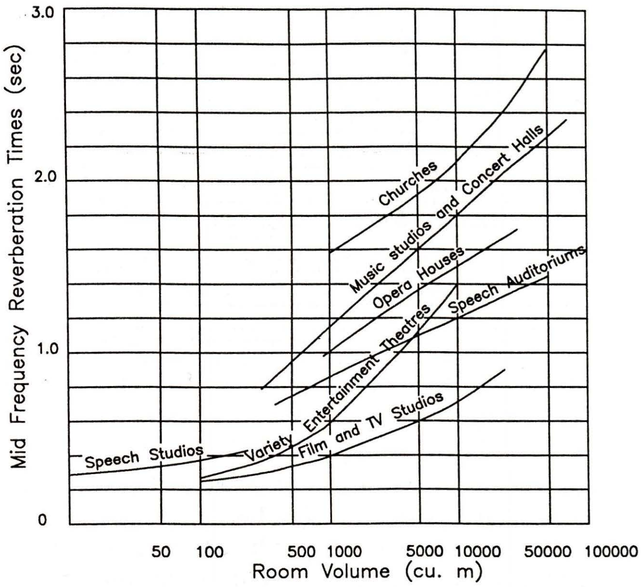

Figure 1: Typical chart of optimum reverberation times for spaces with differing uses. (AS2107 - 1987, p8)

1.6 seconds; for Classical music is approximately 1.8 seconds; for Romantic music is approximately 2 seconds.

As can be seen in Figure 0 (AS2107 - 1987, p8), the optimum reverberation times for speech auditoriums differs greatly from those of churches, opera houses and concert halls, etc. The lower reverberation times for the speech auditoriums demonstrate the influence of reverberation on speech intelligibility.

NB: Speech studios are a specialised space, in which the overall requirements for recording have more to do with the electronic requirements than simple speech intelligibility. In these situations, the requirement is for an acoustically dry or very short reverberation times. Additional reverberation can be added at a later stage electronically.

The importance of the Sabine reverberation time relationships can be seen in the almost predominance of reverberation in the acoustical design of spaces used for music and speech in period following the 1900. In 1958, Parkin and Humphreys (1958, p82) wrote:

> "The present state of knowledge about the acoustics of rooms for music is such that major faults (such as echoes) can be avoided in design ... nearly all the advice that can be given is qualitative only; at this stage of knowledge. The one important exception is the reverberation time which can be specified and ... measured objectively"

Research investigations, such as those reported in Beranek (1992), Jordan (1980), Schroeder, Gottlieb and Siebrasse (1974), with halls predominantly based on the reverberation time design revealed that other factors were important in the way people distinguish acoustic events (Beranek 1994, pp6-9; Jordan 1980, p58). Research into the transient effects of very short duration sound events led to a number of criteria being established, such as Early Decay Time, Articulation Index, Speech Transmission Index (and RASTI), Deutlichkeit and Centre Time.

Much of speech intelligibility development can be traced back to the telephone industry. For example, Bell Telephone Laboratories have been responsible for much of the research (French and Steinberg, 1947; Knudsen 1929; Cavanaugh, Farrell, Hirtle, Watters 1962, p480). It is largely this research that underpins our understanding of speech intelligibility and speech reinforcement systems.

"The most feasible scheme for such a rating [speech intelligibility testing] is probably the one used by telephone engineers for testing speech-transmission over telephone equipment, which goes by the name of articulation tests. ... The writer has used this same scheme for investigating the effects of reverberation and noise upon speech reception in auditoriums." *(Knudsen 1929, p56)*

In the period between 1922 to 1950, researchers, driven by the telephone industry, developed methodologies to test speech intelligibility. As these methodologies becoming available, development of optimum reverberation times for assembly halls were considerably easier (Jordan 1980, p58).

> "It is not a simple matter to give a quantitative rating to a room whish is used for music, since so much depends upon the musical tastes and disposition of the listeners. It is, however, a relatively simple matter to give a quantitative rating to a room which is to be used for speaking, since our primary concern is how well we hear spoken words of the speaker." *(Knudsen 1929, p56)*

In 1947, French and Steinberg published a procedure and methodology for computing the Articulation Index. This was based on work carried out some 25 years previously by H. Fletcher (French and Steinberg 1947, p91). Their paper examines factors which govern the intelligibility of speech by examining quantitatively the ability of the ear to distinguish sounds. The paper provides a platform to examine the influence of reverberation and masking noise on speech intelligibility.

The development of Articulation testing and the Articulation Index procedure in particular, as a quantitative measure of the intelligibility of speech in a closed system was important to the development of the optimum reverberation times as seen in Figure 0.

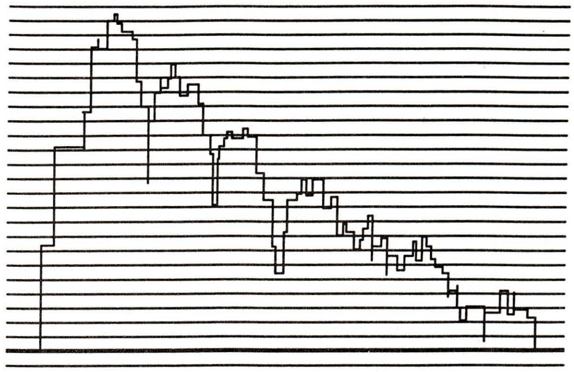

Figure 2 Typical reverberation decay trace.

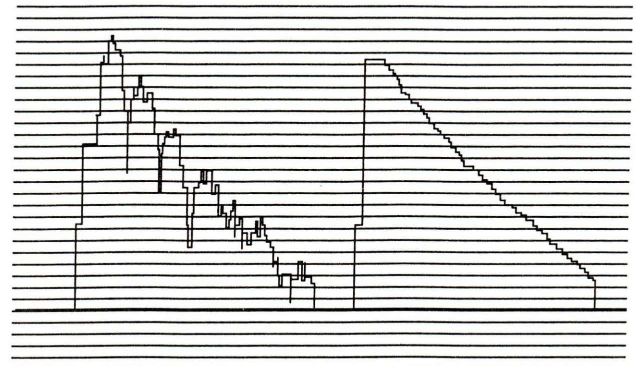

Figure 3: Tone Burst decay and Integrating Impulse Decay curve

Measurement Of Reverberation Time

The origins of the physical measurement of reverberation can be laid at the feet of Sabine, as it was he who coined the phrase (Sabine 1964, pp9-11) in his article titled "Reverberation" in the American Architect and the Engineering Record of 1900. The equipment of the time, used by Sabine in his historic investigations to measure the rate of decay of the sound, or more precisely the *duration of audibility* consisted of an organ pipe, for a constant sound source and a chronograph to record the duration of audibility after the sound had ceased.

It is of interest to note that various approaches to determine the rate of decay were tested by Sabine. Methods included using a sensitive manometric gas flame and measured used a micrometer telescope. Photography of the flame was trialed. Both methods were abandoned due to the fluctuating decay rate. Figure 0 illustrates a trace of a fluctuating decay. Ultimately, Sabine returned to using the ear aided with the chronograph to record the duration of the audible sound.

At the present time there are various methods used to determine enclosure decay rates. The traditional methods use a sound source to produce a steady state signal or an impulse source such as a pistol or gun shot. The decay is measured with a sensitive microphone and recorded via a level recorder (paper trace) or digitally using a computer. Normally the dynamic response of the sound signal above the background noise necessary to measure a 60 dB decay is rarely achieved. Therefore the reverberation time is measured in the -5 to -35 dB region of the decay (AS 1045-1988, pp10) and interpolated out to the 60 dB limit. Typically, many decays are averaged together. Schroeder (1965, p409) offers the following as an explanation of the rational behind the need to average many decays:

"The accuracy with which reverberation time can be determined from decay curves is limited by random fluctuations in the decay curves. These random fluctuations result from the mutual beating of normal modes of different natural frequencies. The exact form of the random fluctuations depends on, among other factors, the initial amplitudes and phase angles of the normal modes at the moment that the excitation signal is turned off. If the excitation signal is a bandpass-filtered noise, the initial amplitudes and phase angles are different from trail to trial. Thus, for the same enclosure, and identical transmitting and receiving positions within the enclosure, different decay curves are obtained - the differences being a result of the randomness of the excitation signal, not of any changes in the characteristics of the enclosure."

A new technique, the *Integrating Impulse Method* developed by Schroeder (1965) is currently being incorporated into many new devices. This technique is said to equal, in one single impulse measurement, the *ensemble average* of indefinitely many decays curves (1965, p409).

For this measurement procedure, a tone burst or filtered pistol shot is used. The decay is measured, squared and integrated producing a smooth curve. Figure 0 illustrates the comparison of the two decay curves. The second curve gives a far more accurate interpolation equation for the determination of the reverberation time over 60 dB.

A major problem in the measurement of reverberation times is having a sufficiently powerful sound source with a flat spectrum. This is particularly problematic in conjunction with the impulse measurement procedures. Common impulse sources such as electrical sparks, popping balloons, pistols, rifles and cannon shots do not always have a flat spectrum, while loud speakers often have insufficient power to achieve an adequate signal to noise ratio for the 20 or even 30dB decays, above the ambient noise levels.

There has been published an experimental procedure using a low power pseudorandom noise which is reported to be successful in noisy environments, such as during a lecture (Schroeder, 1979). To date, this procedure seems to be of academic interest only, as there has been no instrumentation commercially produced based on this procedure.

Early Decay Time

The Early Decay Time (EDT) is the reverberation time based on the first 10dB of the decay curve and extrapolated to 60dB (Beranek 1992, p3). Its importance can be seen in the words of Kuttruff (1994, p38):

> "... The full decay process in a room is audible only after an abrupt and complete stop of sound excitation, but not during running speech or music signals. Nevertheless, reverberation affects the temporal and also the spectral structure of any sound by smoothing it and by mixing its constituents - otherwise there would be no reason to give so much weight to it. However it is rather the initial part of a decay curve which determines to which extent the signal is smoothed or blurred, or generally: how we hear the signal."

Beranek (1994, p20) also confirms that it is the first 10 decibels of the decay curve that is the most important for understanding:

> "In continuous music, approximately 10 decibels of the sound decay can be heard after each note. If the early decay time is short, the sound is clear; if long, but not too long, the music is said to take on the desirable attribute of fullness or singing tone. A very long reverberation time will muddle all but liturgical music."

Jordan (1980, p52) also outlines a criteria by which deficiencies in the acoustical performance of an enclosure maybe detected:

> "Values of EDT which are considerable lower than values of RT are regarded, generally, as signifying acoustical deficiencies."

Early Decay Time as a measurement is an important indices in the design of speech performance enclosures.

Deutlichkeit

Deutlichkeit (D) (Meyer and Thiele 1956, p442), also known as Definition, was proposed by Thiele as a measure of clarity. Bradley (1983, p2051) defines Deutlichkeit as the:

> "ratio of the early sound energy in the first 50ms after the arrival of the direct sound to the total sound energy"

$$D = \frac{\int_0^{50ms} p^2(t)dt}{\int_0^{\infty} p^2(t)dt} \qquad (2)$$

The lower limit of time t_0 is the arrival of the direct sound (Kuttruff 1991, p190). This indices is based on a impulse or short duration pulse sound source (Jordan 1980, p58).

Research by Boré, has shown that there is a good correlation between Deutlichkeit and speech intelligibility (Kuttruff 1991, pp190-191).

Centre Time

Centre Time (seconds), also known as Centre of Gravity Time, Point of Gravity and Schwerpunktzeit (Bradley 1986, p199), is a:

> "... measure of the balance between clarity and reverberance that was proposed to avoid the abrupt division between early and late arriving reflections". *(Bradley and Halliwell 1989, p17)*

$$TS = \frac{\int_0^{\infty} tp^2(t)dt}{\int_{0.}^{\infty} p^2(t)dt} \qquad (3)$$

The Centre Time index is very similar to the *echo criterion* proposed by Dietsch and Kraak (Kuttruff 1994, p38):

$$TS = \frac{\int_0^{\tau} t|p(t)|^n dt}{\int_{0.}^{\tau} |p(t)|^n dt} \qquad (4)$$

where $n = 1$ for music and $n = 2/3$ for speech. When $n = 2$ and $\tau = \infty$ this equation agrees with the equation 3

Both Deutlichkeit and Center Time are closely related to speech intelligibility. The smaller the value of the Center Time (or higher the Deutlichkeit value) the higher is the expected speech intelligibility at the receiver position (Kuttruff 1994, p36).

Conclusion

This paper briefly looks at the history of acoustics in relation to the speech intelligibility and public spaces used for speech. Some speech performance indices are presented and sound decay measurement such as Reverberation Time and Early Decay Time are dicussed.

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The Responses Of Buildings And Occupants To Impulse Noise

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Introduction

The development of a criterion that predicts community reaction to noise is difficult, because of the variability in the reaction of individual persons in the community. However, when viewed as a community, and given a statistically large enough population, the reactions of a group can be quantified with respect to the number of persons that will react adversely to noise above a particular level.

The results presented in this paper are derived from a study into the relationship between community reaction and the Sound Pressure Levels (SPLs) that occur in residences adjacent to Defence facilities where noise of an impulsive nature exists.

The study specifically concerned the reaction of residents to noise generated by explosions and the firing of large calibre weapons. Reference 1 describes the development of an earlier criterion by these Laboratories in 1984 for predicting community annoyance, termed the Accumulated Peak Level (APL) Criterion.

The Community Annoyance Level Study

Due to some recognised shortcomings of the APL criterion, it was decided to conduct further research using a slightly different approach to that used during the original APL survey. For this study, the response from each resident is matched with sound level measurements taken at the time of the exposure. Thus, the resident's response to individual explosions can be gauged, as well as the response to the overall presentation of explosions at that particular residence. By this method the determination of threshold levels for audibility and annoyance can be determined. Three locations were selected to participate in the study: Puckapunyal, Victoria; Port Wakefield, South Australia; and Woomera, South Australia.

The study was conducted in five phases: source selection, three phases of measurements and a final phase of data analysis and reporting.

Selection of the explosive source

Phase one of the study was conducted at Holsworthy Army Range to evaluate four types of commercial explosives. The aim of the trial was to find a noise source which was a cheaper alternative to using actual field pieces and various ammunition types. The explosives evaluated during this phase were Dynagex, TNT, PE4 and ANFO.

The data obtained from this series of measurements were compared to the data collected at the Singleton Field Firing Range from M198 155 mm and Hamel 105 mm howitzers[2], with a view to finding explosives that would mimic as closely as possible the noise signatures of these weapons.

The results of some of the Singleton measurements show that at distances of 1.6 km to 6.4 km from the noise source, residences may be exposed to linear peak SPLs of »90 dB to »130 dB.

Close correlation was obtained between the M198 muzzle waveform (using charge 5 Green) measured at 3.2 km and 4 Kg of TNT measured at 3.2 km.

Based on the peak levels, frequency content, cost and ease of handling, it was decided to use Flaked TNT in three charge sizes; 4 kg, 1.25 kg and 400 grams. In addition to the above sizes, a fourth charge consisting of a 110 gram primer and detonator was added to approximate a 125 gram charge.

Acoustic measurements

A microphone, with windscreen installed, was mounted on a tripod 1.2m above ground on the outside of the residence towards the explosive source. Inside the residence, another microphone was mounted on a tripod 1.2m above the floor. The electrical outputs of both microphones were connected via cables to the inputs of two Precision Sound Level Meters (SLMs). Each system consisted of Brüel & Kjær equipment of type 4155 Microphone, 2639 Microphone Preamplifier and 2231 SLM.

The outside microphone was placed in a position where (as much as was possible) a clear path existed towards the explosive source location. In general, the location selected tended to be either on the front lawn of the residence or in the backyard. The inside microphone was placed either in the kitchen or the lounge, family or dining room of the residence. The selection of a location for the inside microphone did not follow any set pattern. Often the resident would be very close to the microphone, sometimes the resident would be in another part of the residence. In general, the kitchen was not a preferred location due to the noisy activities that occur there. The measurement location was also selected for ease of cabling access and minimising the intrusion of the NAL operator.

Four residences were surveyed during each series of 12 explosions and four series of explosions were conducted each day. Each series of explosions consisted of 12 shots in the following sequence; 1.25 kg, 400 g, 4 kg, 125 g, 4 kg, 1.25 kg, 125 g, 400 g, 1.25 kg, 400 g, 4 kg, and 125 g.

During the explosion sequence the resident was asked to go about his/her normal activities so as to try to obtain the best "representative" response from the resident to the noise levels presented.

After each shot, the NAL operator would ascertain if the resident heard the impulse and would ensure that the Resident's Questionnaire was completed. Thus, a questionnaire was completed for each shot regardless of whether it was heard by the resident.

Radio contact was maintained with the controller at the source to ensure that each NAL operator was ready for the next shot, had recorded the correct explosion size for each shot detonated and that the equipment full scale deflection was correctly set. As part of the detonation routine for each shot, a countdown from 10 was given over the radio to each NAL operator. This countdown (not audible to the resident due to the headphones being used for radio monitoring) enabled the NAL operator to reset each SLM just prior to the audible impulse arriving at the residence.

Resident's questionnaire

Questionnaires were developed to obtain the resident's response to the noise levels presented. The questionnaire developed for the Puckapunyal phase of the study was examined after the Puckapunyal data had been partly analysed, and modifications and additions were made for its use at Port Wakefield and Woomera.

When the NAL operator arrived at the residence to conduct the tests, the resident was guided through the questionnaires and asked to continue with his/her normal activities. The NAL operator set up and calibrated the equipment and then advised the controller at the source that he was ready for the first shot to be detonated. When all four locations were ready the first explosion would be detonated. The NAL operator would then ensure that the resident had completed a questionnaire for that shot. The NAL operator would record his acoustic data, answer his questionnaire, and then the process would repeat for each of the 12 shots presented.

For each explosion the resident and NAL operator recorded on their questionnaire:

- A rating of the loudness of the explosion
- An annoyance rating of the explosion
- An acceptance rating of the explosion
- Their observance of house vibrations, window rattles or objects on shelves moving
- Their reaction at the time of the explosion, i.e. did the explosion....... startle you? frighten you? cause you to feel irritable or edgy? make you feel tense or nervous? or disturb others in the house?
- Their location at the time of the explosion i.e. inside the house with the doors or windows open, inside the house with the door or windows closed, or outside the house.
- Their activity at the time of the explosion, i.e. listening to TV, radio, music, talking etc., concentrating e.g. reading, an activity e.g. housework.

Results of the analyses

Data were collected from the three communities, involving a total of 159 residences. Altogether 1883 individual explosions were detonated and nearly 70,000 data items were available for analysis.

During the analysis process, wherever possible, the data from Puckapunyal, Port Wakefield and Woomera have been combined to increase the sample size and improve the reliability of the results.

Statistical analysis of the data[3,4] was performed using the Statistical Package for Social Sciences. Using this software the data were examined using factor analysis, correlation techniques and regression analysis.

Analysis of the relationship between the actual noise levels presented to residents and their respective responses has been conducted using the software package Lotus 1-2-3 Version 3.1. All 70,000 data items were entered in a spreadsheet format and various graphs were produced using filtering techniques to sort the data.

Analysis of the data from all three locations was conducted to determine the noise measure that best correlated with the residents' reaction. A composite rating of reaction (termed General Reaction or GR) was formed by adding the results of the questions about loudness, annoyance and acceptability. Factor analysis of this composite rating of GR indicated that only one factor was significant in the results, i.e. the residents' reaction to the noise.

When GR was compared to each noise measure the resulting correlations indicated that at each of the three locations the outside peak level of the impulse was the best predictor of the residents' reaction.

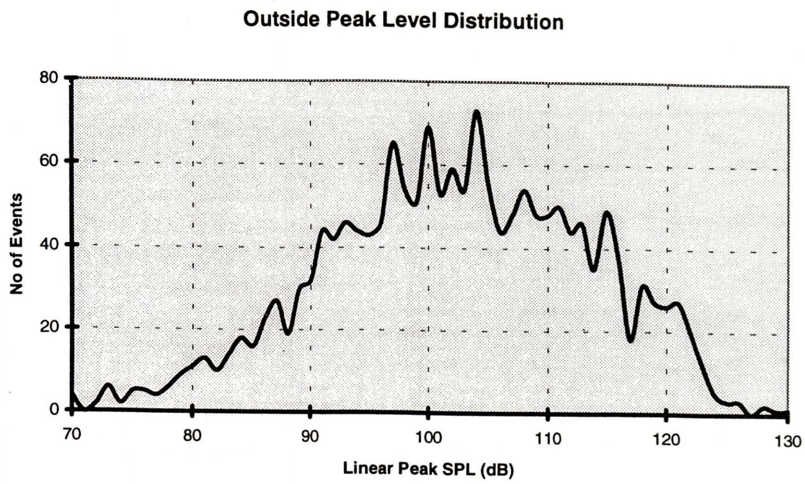

Figure 1. Distribution of outside peak levels for all sites

This is fortunate for a number of reasons:

- it is one of the easiest acoustic measurements possible,
- building vibrations and window rattles are caused by the outside noise environment,
- the prediction of peak levels at a distance from a source is possible, thus enabling predictions of community response to be made, and
- if an inside measure was used then allowances for the building structure would need to be made.

Having singled out the outside peak level as the best predictor of the residents' response, various analyses were conducted to derive the relationship between the residents' responses and the measured peak levels.

Analysis of the data showing the location of the resident at the time of the explosion shows that the resident was outside the residence on only 5% of all occasions. Consequently, the following analysis of the data is on the basis of the resident being inside the residence.

Figure 1 shows the distribution of measured outside peak levels for all sites.

With reference to Figure 1, several things are worthy of comment:

- for Puckapunyal 95% of the data was between approximately 75 and 116 dB,
- for Port Wakefield 95% of the data was between approximately 88 and 123 dB,
- for Woomera 95% of the data was between approximately 81 and 120 dB.

The lack of sufficient data outside the ranges mentioned above is a cause of sometimes spurious results in the analysis of relationships reported later.

Examination of the results recorded from the microphones outside and inside the residence reveal an average building peak attenuation of only 4 dB.

This surprising result can be better appreciated when it is realised that for all measurements conducted, the residence had at least one door or window open on 84% of occasions. This poor attenuation is also a result of the inability of building structures to effectively attenuate the low frequency dominated impulses of this type.

Figure 2 shows the loudness rating reported by the resident (when the resident was inside the residence) versus the measured outside peak level. Some explanation of the graph is required:

- Shown at each loudness rating level from 0 to 5 is a frequency distribution of the data at the rating level versus outside peak level. The amplitudes of the frequency distributions have been adjusted by a factor so that the amplitude shown at each 1 dB increment of peak level represents the percentage of the overall data at that level.
- The interval between each loudness rating represents a scale of 0 - 100%.
- For example, at 100 dB peak level, it can be seen that approximately 36% of residents reported a loudness rating of 0 (didn't hear), 48% reported a rating of 1 (just heard) and 16% reported a rating of 2 (easily heard). Further, at 80 dB peak level, 65% of the data were represented by a 0 rating (didn't hear) and 35% by a 1 rating (just heard). Similarly at 129 dB peak 100% of the data were represented by a 5 rating (very loud).

Loudness Response

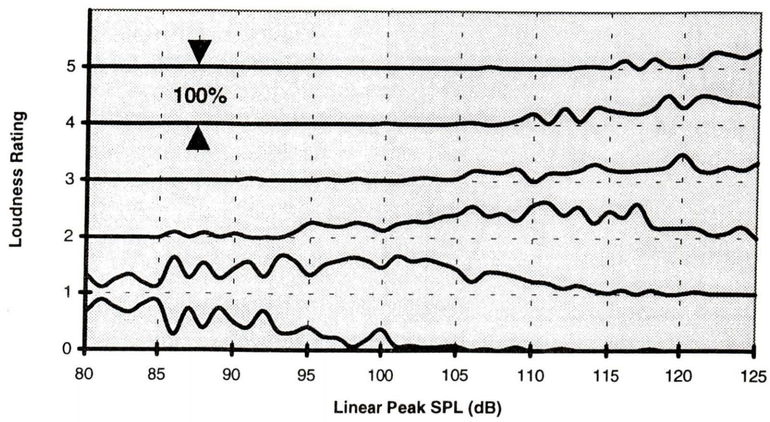

Figure 2. Loudness rating by the resident when inside residence versus outside peak level for Port Wakefield and Woomera data. Loudness rating: 0 = Didn't hear, 1 = Just heard, 2 = Easily heard, 3 = Slightly loud, 4 = Loud, 5 = Very loud.

All graphs presented in this paper that have been constructed using this method are characterised by a 100% scale marking on the graph as shown in Figure 2.

It should be remembered that at the extremes of the peak level data range there are few data points available and thus 100% of the data may represent a total of only a few measured values (see Figure 1).

The graph for the Puckapunyal data is similar to that shown in Figure 2 for the Port Wakefield and Woomera data, but with a loudness scale 0 to 6.

It can be seen in Figure 2 that, as would be expected, the loudness rating by the resident increases as the outside peak level increases. Also, for all intents and purposes, the rating of loudness 0 (didn't hear) ceases at approximately 100 to 105 dB peak level. In other words, outside peak levels above 100 dB are almost guaranteed to be heard inside the residence by the resident.

In addition, at 95 dB peak level 74% of residents reported a loudness rating of either 0 or 1 and 26% reported a rating of 2. There were no reported ratings of 3 or higher at this level. At about 94 dB outside peak level residents reported a loudness rating of either 0 or 1, that is, there were no ratings of 2 or higher. This points to a threshold value that should represent a level at which (and below) a substantial number of impulses per day should be permitted. This level of about 95 dB peak for a substantial number of impulses per day is supported by the graph of annoyance versus outside peak level shown in Figure 3.

Figure 3 shows the frequency distribution of annoyance ratings versus outside peak level. It can be seen that at the 95 dB peak level 82% of respondents reported the explosion as "Not at all" annoying with the remaining 18% of respondents reporting it as "Slightly annoying".

The level of 95 dB peak (or less) for a substantial num-

Annoyance Response

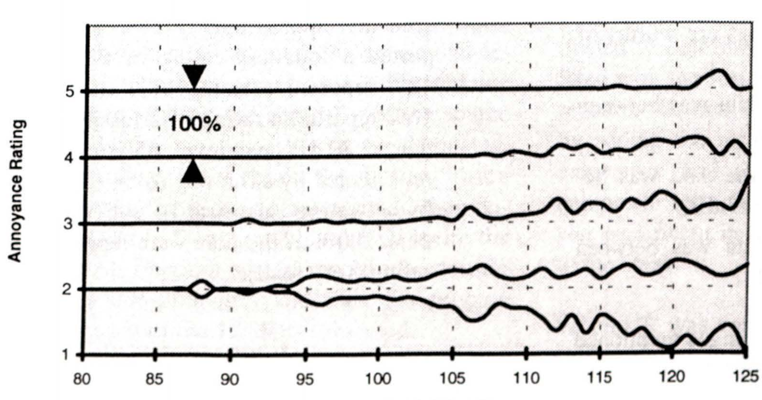

Figure 3. Annoyance rating by resident (when inside) versus outside peak level for Port Wakefield and Woomera data. Annoyance rating: 1 = Not at all, 2 = Slightly, 3 = Moderately, 4 = Considerably, 5 = Highly.

Acceptability Response

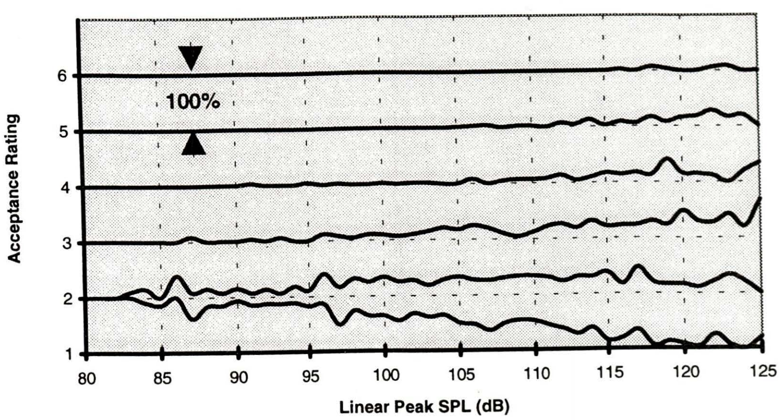

Figure 4. Acceptance rating by the residents versus outside peak levels for all data. Acceptance rating: 1 = Completely accept., 2 = Acceptable, 3 = Fairly accept., 4 = Fairly unacceptable., 5 = Unacceptable, 6 = Completely unacceptable.

ber of impulses per day is further supported by the residents' rating of acceptance as shown in Figure 4.

The data shown in Figure 4 indicates that at the 95 dB peak level, 86% of respondents reported it being "completely acceptable", 11% reported "acceptable", 0% reported "fairly acceptable" and 3% reported it being "fairly unacceptable".

In the questionnaires completed by residents after each explosion the resident was asked if the impulse startled or frightened him/her, caused him/her to feel irritable or edgy, tense or nervous or disturbed others in the house. Of these responses, only the startle or frighten response was reported often enough to be of significance.

Figure 5 shows the percentage of data items, for each 1 dB band of peak levels measured outside, where the resident reported being "startled". Data is from all locations. It is noticeable that there is a discernible rise in reported instances of being startled starting at approximately 100 dB peak level. At 115 dB peak it can be seen that approximately 20% of residents reported being "startled".

Figure 6 shows a graph of the reported instances of being "frightened" versus outside peak levels. Fewer residents reported being "frightened" by the explosion when compared to being "startled".

Analysis of the data concerned with window rattles and building vibrations revealed that the NAL operator reported these effects more frequently than the resident. This result is not unexpected as the NAL operator had forewarning of the event and was able to pay attention to determining if the effect was present, whilst the resident may not have been able to differentiate at the time between a loud noise with and without building effects.

The following figures show only the results obtained from the NAL operator.

Figure 7 reports the percentage of data items where the NAL operator recorded that the building vibrated during the explosion. It is noted that at the 115 dB peak level, building vibrations were reported on greater than 50% of all occasions.

Startle Response

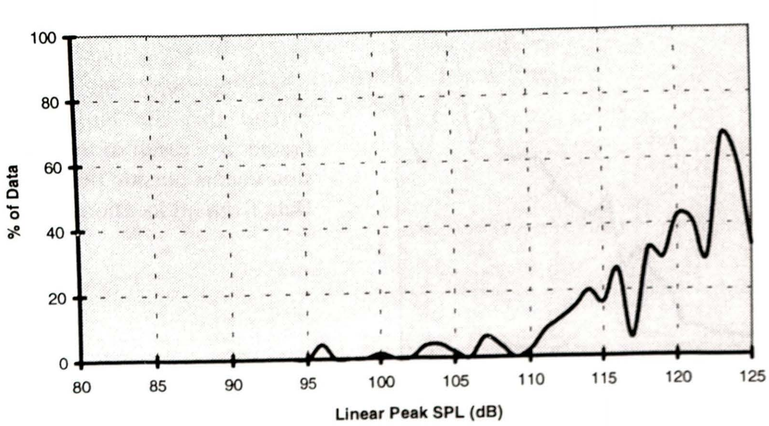

Figure 5. Percentage of data items where resident reported being "startled" by explosion versus outside peak level, for all locations.

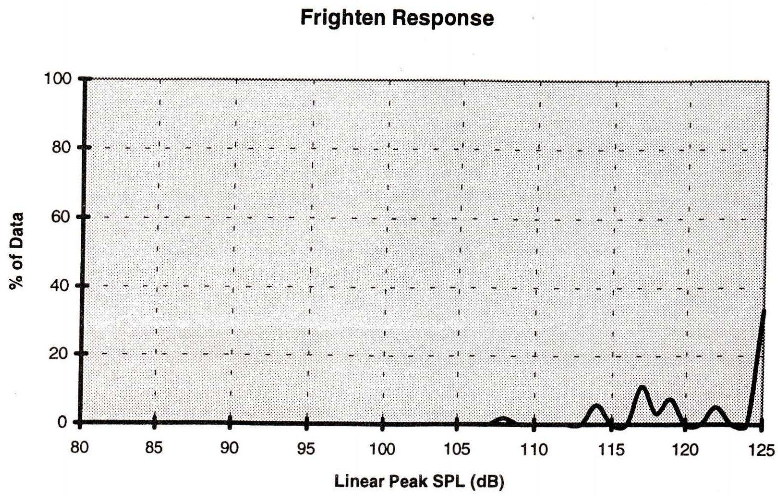

Figure 6. Percentage of data where residents reported being "frightened" by explosion versus outside peak level for Port Wakefield and Woomera data.

Similarly, from Figure 8 it can be seen that window rattles were reported in approximately 50% of all cases at 115 dB peak level.

Schomer[5] reports the results of others, confirming that additional annoyance is generated beyond that generated by audibility of the impulse, by house "rattles" and "startle" effects on residents. He also reports[6] that the subjective response of a resident changes by 6 to 13 dB, when the outside peak level is between 112 and 122 dB, due to the presence of "rattles", in the building structure. Examination of the Port Wakefield and Woomera data shows general agreement with these figures. In addition, in another study[7], he reports that house "rattles" and "startle" were the two most reported characteristics of artillery noise by respondents expressing high annoyance.

Conclusion

In this paper, some of the results of a study into the relationship between community reaction and impulse noise are presented. These results shall be used to develop a criterion for predicting community annoyance which will be the subject of a further paper.

The results from the study show that the linear peak SPL measured outside the residence is the best predictor of community reaction.

The results also show that for practical purposes a lower limit exists, below which, a substantial number of impulses per day should be permitted. This limit was determined from responses to questions about the loudness, annoyability and acceptance of the impulse. The limit was found to be at approximately 95 dB linear peak SPL.

Above 110 dB linear peak SPL, residents reported being "startled" by the impulse although very few reported being "frightened".

Building and window vibrations were noticed with increasing occurrence when linear peak SPLs of greater than 100 dB were measured.

Acknowledgment

This study has been supported by the Department of De-

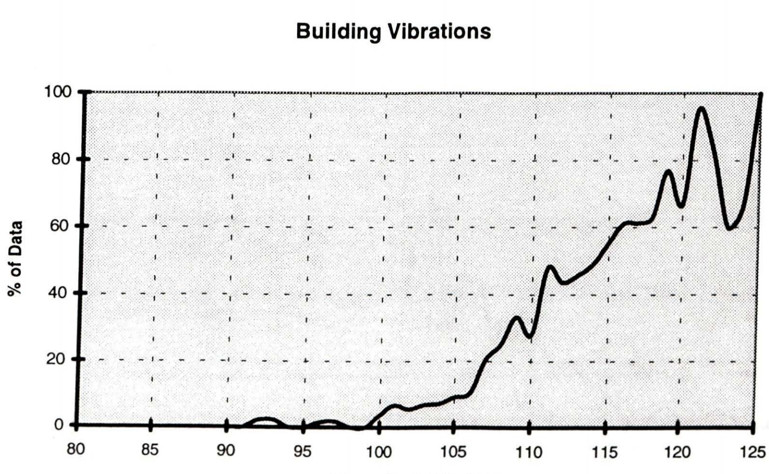

Figure 7. Percentage of data where the NAL operator recorded that the building vibrated as a result of the explosion versus outside peak level. Data from all locations.

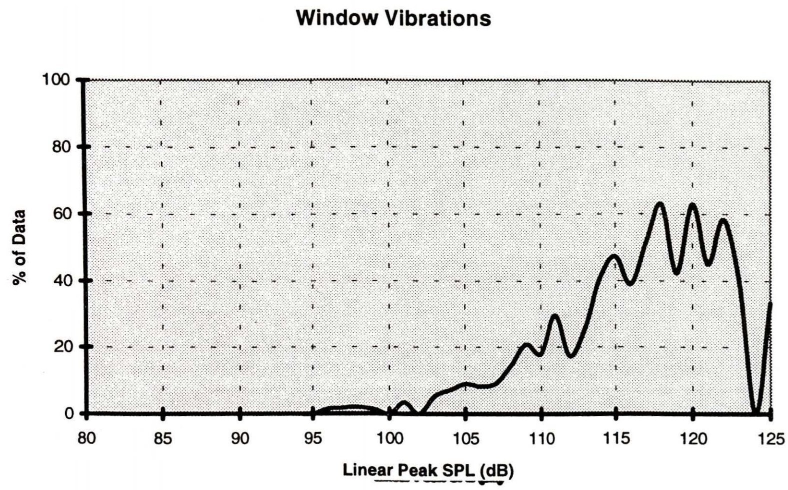

Figure 8. Percentage of data where the NAL operator reported that the windows rattled as a result of the explosion versus outside peak level. Data from all locations.

fence, Estate Management Branch.

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Session 2A

Road Traffic Noise

Feasibility Study for Determining the Acoustic Quality of Rooms for Music

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Abstract

The present work is aimed at producing a test, similar to a speech intelligibility test, which can be used to rate the quality of rooms for music. Listeners' perceptions of duration and frequency of sounds were investigated to find a useful indicator of the acoustics of a room. The duration and frequency research design was based on a discrimination task and a 2AFC experimental procedure. Clear trends in accordance with the varying reverberation times were found in results of duration discrimination test. It was found that both frequency and duration discrimination were influenced by the room conditions and that these discrimination procedures may form the basis for room acoustics assessments.

Introduction

The concert hall provides the acoustic link between the musical performer and the listener. Although there are many views on the acoustical characteristics which are important for good concert halls there have been very few attempts to make comparative subjective assessments of the acoustics of halls (Fricke & Haan, 1995). If auditorium acoustics is to develop as a science then a reliable procedure for rating halls needs to be developed which is independent of the music, performers and other confounding variables.

The skilled performer must produce subtle changes in tone quality as a component of musical expression. A successful room for music is one in which the listener is able to hear such subtleties of musical expression(Keefe & Goad, 1992). The assumption in this study is that listener's perception of duration and frequency effectively reflect the acoustical conditions of a room. Therefore, we have adopted a duration and frequency research design based on a discrimination task. Jeon and Fricke(1994a, b) have examined the possibility of music perception test for room acoustics. In their first trial, to find useful indicator of the acoustics of a room, human discrimination ability in duration perception was investigated as one measure. A simulated room sound field, provided by a digital signal processor, was used. They found that the variation of the percentage correct response(P(C)) of subjects with room type was affected by the room conditions.

Abel's work(1972), in which musicians served as subjects, shows that the difference limen of duration less than 50 msec is nearly proportional to the square root of the duration. Also for the tones whose durations are greater than 50 msec the DL is approximately 10% of the duration. Jeon and Fricke(1995) found there exist two break points at 100 msec and 2 seconds in duration discrimination. For short duration tones ranging from 25 to 100 msec, the percentage correct response was found to linearly improve with increasing duration of signals. However, the JND was constant for durations between 100 msec and 2 seconds while for extended stimulus durations(2 to 8 seconds) the JND was again linearly improved. In the present work the value of ΔT was determined on the basis of the JNDs obtained in the previous studies ($\Delta T/T=0.16$ for short duration, 0.0875 for long duration). Also Spielgel and Watson(1984) found that the musicians' thresholds are in the range $0.001<\Delta f/f<0.0045$. Fastl and Hesse(1984) reported that the difference limen existed in the range $0.001<\Delta f/f<0.004$, which was obtained by an adjustment procedure ($0.001<\Delta f/f<0.006$ for Yes/No procedure). In the light of previous work, three relative frequency differences($\Delta f/f$) of 0.008, 0.007 and 0.006 were chosen for different frequencies(500, 1K, 4 kHz), which are considered to give a reasonable coverage of the sounds in a concert hall.

An experiment was carried out to find the effects of the acoustical conditions of a room on duration and frequency perception. Four different reverberant fields were prepared within the reverberation chamber of the University of Sydney. In this experiment, duration and frequency discrimination tests were carried out. The method of constant stimuli and two alternative forced choice was used for this purpose. The main concern of this work was placed on the differences in correct responses between different sound fields, not on the difference limen. Simpson(1988) reported that for 100 trials or less, the method of constant stimuli is the better trial placement rule, after his Monte Carlo simulation of various psychoacoustic methods. The constant stimuli is known to have other virtues besides low bias and variability of thresholds resulting from its use:

Table 1. Measured reverberation times(1 octave) in each reverberation condition(unoccupied)

Frequency(Hz)	Condition 1	Condition 2	Condition 3	Condition 4
125	1.7	2.4	3.5	4.5
250	1.6	2.2	3.3	4.2
500	**1.5**	**2.0**	**3.2**	**4.0**
1000	**1.3**	**2.0**	**3.0**	**4.0**
2000	1.0	1.7	2.6	3.2
4000	**0.5**	**1.4**	**1.7**	**2.5**

1) The extreme simplicity in implementing the trial placement rule is the most obvious advantage.
2) The independence of the stimulus level presented on a given trial from that presented on previous trials.
3) It requires no prior assumptions about the shape of the psychometric function before proceeding. After the data have been collected, any of a number of functions can be fitted.

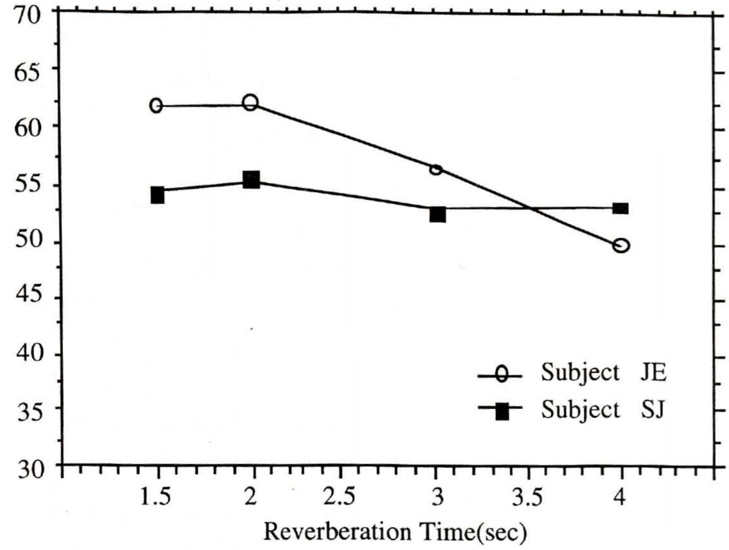
Experiment: The Effect Of Reverberation Time On Duration And Frequency Discrimination

Subjects

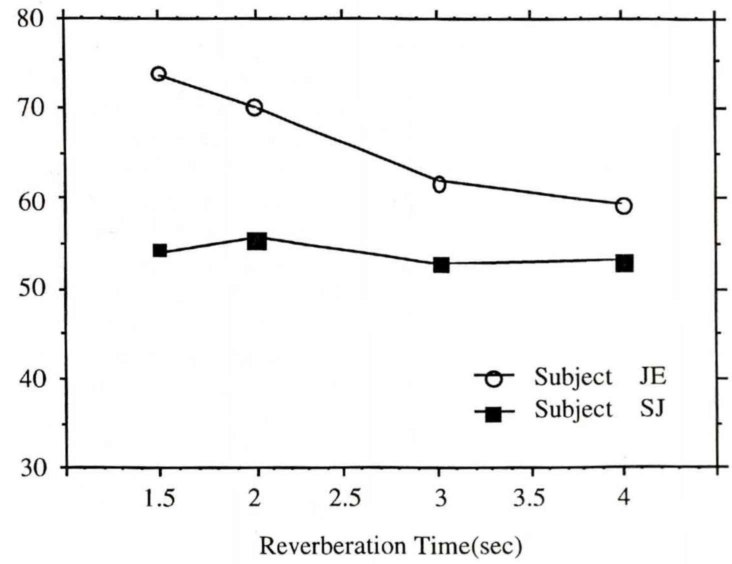
To avoid fluctuation in their performance and extract the effect of the acoustic condition of a room, subjects who have musical experience were considered. Fine & Moore(1993) reported that musicians have sharper auditory filters than non-musicians. Two female music students from the Australian Music Institute served as subjects. They didn't have any previous experience in psychoacoustics experiments. Each subject participated for two weeks and was paid an hourly rate.

Table 2. Pairs of stimuli used in both duration and frequency discrimination test.

	S/C Ratio	Used Pairs
Duration Discrimination	short duration pairs (50ms centred) $\Delta T/T = 0.16$	26:34 msec, 34:42 msec, 42:50 msec, 50:58 msec, 58:66 msec, 66:74 msec
	long duration pairs (400ms centred) $\Delta T/T = 0.0875$	295:330 msec, 330:365 msec, 365:400 msec, 400:435 msec, 435:470 msec, 470:505 msec
Frequency Discrimination	500 Hz-centred frequency pairs ($\Delta f/f = 0.008$)	492:496 Hz, 496:500 Hz, 500:504 Hz, 504:508 Hz
	1 kHz-centred frequency pairs ($\Delta f/f = 0.007$)	986:993 Hz, 993:1000 Hz, 1000:1007 Hz, 1007:1014 Hz
	4 kHz-centred frequency pairs ($\Delta f/f = 0.006$)	3952:3976 Hz, 3976:4000 Hz, 4000:4024 Hz, 4024:4048 Hz



(a) short duration(26-74 msec)



(b) long duration(295-505 msec)

Fig.1. Percentage of correct response for the duration discrimination test under four different reverberation conditions

Experimental arrangement

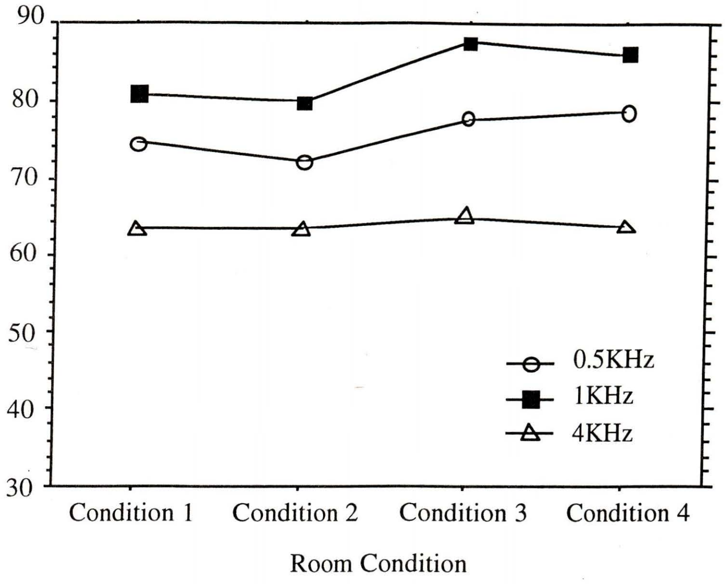
Macintosh Power PC controlled the experiments. The MacRecorder sound system with its application, SoundEdit^{TM}, was used for recording, editing, playing and storing of sounds. Stimuli were presented by MacroMind Director. The processed stimuli were filtered by a bandpass filter KRON-HITE 3700 and amplified by JVC Integrated Stereo Amplifier. In each acoustic condition, the stimuli were presented through a speaker(INTERDYN 3WAY Speaker) located in front of a subject. Subjects were asked to face directly forwards, but no physical restriction was placed on head movement. To control the reverberation time in the reverberation chamber, sound absorbing fibreglass panel units(910 ∞ 1200 mm) were installed. These absorbing panels were evenly distributed along the wall and the floor. The reverberation times for each room condition used in this experiment are given in Table 1. The reverberation times at 0.5 and 1 kHz were varied from 1.3 to 4.0 seconds (unoccupied). The reverberation times at 4 kHz ranged from 0.5 to 2.5 seconds. The subject was seated in the middle of reverberation chamber and asked to respond by pressing the keyboard buttons.

Stimuli

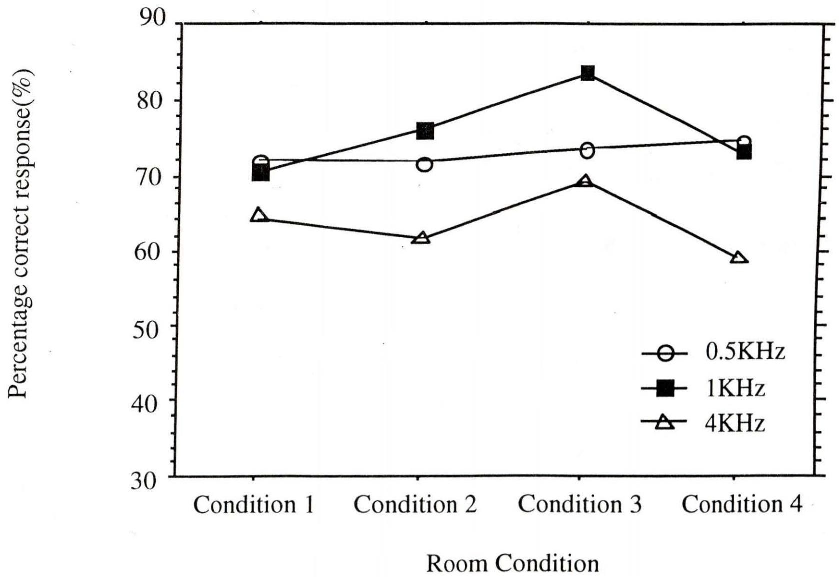
In the duration discrimination test, pure tones of 1 kHz were used. The duration of standard tone (S) was varied from 26 msec to 66 msec for short durations and from 295 msec to 470ms for long durations. The values of $\Delta T/T$ were 0.016 for the short duration signals and 0.0875 for the long duration signals (ΔT is the time difference between the standard tone, S, and the comparison tone, C). For the frequency discrimination test, stimulus pairs of pure tones were prepared for 3 frequency ranges (492-505 Hz, 986-1014 Hz, 3952-4048 Hz). The detailed information on stimuli are given in Table 2. The reproduction level of each auditory stimuli was controlled to have an equal intensity of 80 dB, when measured at the listener's position with a single 1/2 inch free-field microphone (B&K Type 4190). To minimise switching transients at the onset and offset of the signals, rise/decay times of 5 msec were used in all presentations. All pure tones used in this experiment were made as 16 bit sounds(at a sampling rate of 44.1 kHz).

Procedure

The test was performed in the reverberation chamber of the University of Sydney. The reverberation time was controlled by adding or removing sound absorbing panels. Daily sessions were composed of two different reverberant conditions. Four hours of training period was provided for each subject. During this period, it was observed that subjects adapted to the sequence of the experiment. For example, when the reverberant condition of 3.0 seconds was followed by 4.0 seconds, the results of second condition were sometimes slightly higher than



(a) Subject JE



(b) Subject SJ

Fig 2. Percentage of Correct responses for the frequency discrimination test under four different reverberation condition.

those of previous condition. To avoid this kind of adaptation, the reverberation condition was randomly organised Under each reverberant condition, the listener was presented with a pair of stimuli composed of different durations for 15 minutes, followed by a 20 minute frequency discrimination test. Subjects were asked to select the sound they believed to be longer or higher. In a pair each stimulus was separated by 500 msec inter-stimulus interval and 2000 msec was given as response interval. All the stimuli pairs were presented 100 times and the order of the longer/shorter duration stimuli and higher/lower frequency stimuli were equally distributed.

Results And Discussion

A. The influence of reverberant field on duration discrimination

The observed variations in the P(C) of duration discrimination, under different reverberant conditions, are related to the reverberation time of the room (Fig.1). The results obtained in both the short and long duration discrimination test show clear trend in accordance with the varying reverberation time. In particular the P(C) of JE steadily decreased as the reverberation time increased while that of SJ slightly diminished. It appears that there is a maxima in the P(C) vs reverberation time relationship at about 1.5 to 2 seconds but test at lower reverberation times are necessary to confirm this.

It was found that for the correlation between P(C) for short duration signals and the reverberation time is $r_S = -.976$ for subject JE with $p = .027$. Also for long duration, the result of JE showed high correlation with the reverberation time(r_S = -.718 with $p = .0342$). Both results are significant at the $p < 0.05$ level. Although the P(C) of SJ also showed high correlation with the reverberation time(r_S = -.718 with $p = .367$ for short duration, $r_S = -.681$ with $p = .406$ for long duration), it was found to be statistically insignificant.

Notable differences were found in the P(C) of both short duration and long duration discrimination between the two subjects. JE showed a higher percentage of correct responses in most reverberant conditions than did subject SJ. Also it was observed that subject JE discriminated more easily the differences in long duration pairs than short duration pairs while subject SJ didn't show any prominent differences. The results indicate that it would probably be better to develop an assessment procedure based on the shorter duration time as the difference between subjects is less, but this is not the only consideration.

B. The influence of reverberant field on frequency discrimination

The variations in the P(C) of frequency discrimination did not show any simple relationship with the reverberation time of the room(Fig.2). Both subjects performed poorly with the 4 kHz centred tones while a high percentage of correct responses was recorded for the 0.5 and 1 kHz centred tones. Also JE showed higher performance for the 0.5 and 1 kHz tones than SJ while any prominent differences were not observed between the subjects for the 4 kHz tones.

The results are encouraging as they indicate that there may well be an optimum room condition for frequency discrimination.

Conclusions

The influence of reverberant field on duration and frequency discrimination was observed and analysed. Clear trends in accordance with the varying reverberation times were found in the results of a duration discrimination test (See Fig. 1). The P(C) of frequency discrimination didn't have any simple dependence on the reverberation time of a room but it appeared that there was an optimum reverberation time for frequency discrimination. It is concluded that frequency and duration perception may be good indicators of the acoustics of rooms.

Each subject has a different degree of musical experience, background and interests. This may be a clue to the observed discrepancies between the results of the subjects. As Hantz et. al.(1992) reported this may be caused by the correlation of the degree of musical experience and/or the presence of absolute pitch ability with a differential neurophysiological responsiveness of the memory or information-processing systems. The fact that subject JE showed better performances in both duration and frequency discrimination test supports this possibility to some degree. Another possibility is that there are differences in individual preferences for the acoustical condition of a room. However this possibility can't be discussed at this stage as no information on preferences for different acoustical conditions has been obtained. This will be done in the near future.

In this work, listener's perception of duration and frequency was investigated to find a useful indicator of the acoustics of a room. The obtained results in duration and frequency discrimination tests, using pure tones, are obviously promising. Future work will involve the listening test using musical tones in different room conditions. The perception of harmonic structure, including timbre, will be explored to find a useful indicator of the acoustics of a room. Further testing also needs to be carried out , on the situations described in this paper, using extra subjects, to determine the variability between individuals and hence determine the number of subjects required for tests based on discrimination of tones. Similar tests need to be carried out in many different rooms and the opinion of subjects sought on the quality of the acoustics of those rooms for playing and listening to music.

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A Method For Assessment Of The Environmental Noise Impact Of New Traffic Bypasses

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Abstract

Noise impact assessments based solely on an analysis of expected noise levels do not reflect the impact on community annoyance. Community annoyance depends on other factors such as the noise history of the communities affected, the way in which the character of the noise will change, and the time period over which the expected changes in noise environment will take place. A noise impact assessment method has been developed which incorporates the results of recent research into sleep disturbance and community response to changes in traffic exposure. Significant impact is defined in terms of the likely change in community annoyance.

Introduction

In November 1992 Carr Marshall Day Associates was commissioned by VicRoads to provide a noise impact assessment of the proposed Western and Southern Bypasses. Now known as the City Link Project, this will be Australia's largest privately owned infrastructure project, and will link together three of Melbourne's radial freeways.

It was predicted that both bypasses would carry large volumes of traffic, and would be major road freight routes. The Southern Bypass was to be entirely underground, and its only noise impact would be due to changes in traffic volumes on existing roads. The Western Bypass was mostly elevated, and passed near two residential areas. Of the 11 areas assessed, 8 were affected only by changes in traffic volumes on existing roads.

Listening to the public

Early public meetings and meetings with community representatives made it clear that an impact analysis based on changes in noise levels was not going to be a popular one. The majority of the predicted changes in noise level were less than 3dBA. In one of their information leaflets, VicRoads went on to suggest that since 3dBA is only just perceptible, none of the noise impacts of the proposal were significant.

The next public meeting was like a re-enactment of the bombing of Sarajevo. One resident summed up the general feeling at the meeting by saying, "If you think we're not going to notice an increase in traffic volume of 50% [on opening] then you're either crazy or lying."

Through strongly worded, the gathering of such feedback was the genuine purpose of the public meetings, and it was important that the noise impact assessment approach was, to some degree at least, acceptable to the community representatives. I tried to imagine myself in a situation where traffic on Bell Street (a four-lane ring road near where I live) increased by 50% over a period of a few months and I could see that they had a point.

However, we couldn't just take their word for it. We would have to find studies which supported such claims, and which gave an indication of how to factor such considerations into an assessment of impact. What's more, the effects on the community of short-term changes to the noise environment would have to be measurable and long-lived. It might be quite true that people would notice the 50% increase when it happens, but a year later things might be back to normal.

In surveys of community attitudes toward traffic noise, noise from trucks at night is frequently cited as the main source of annoyance. This was also the main source of annoyance for most of the people who came to the public meetings. In fact, the choice of the $L_{A10,18hr}$ as a noise descriptor was seen by some people as a way of avoiding actually measuring truck noise at night. It was explained that any noise barrier put in place to reduce the $L_{A10,18hr}$ would also reduce truck noise, but it was clear that our assessment had to address truck noise and sleep disturbance directly.

What is significant?

In Victoria, environmental effects assessments must consist of a comparison of the consequences of doing nothing (the no-build scenario) with the consequences of the proposed development (the build scenario). It was clear early on in the process that without noise control measures such as higher noise barriers on feeder roads and noise insulation for the noisiest locations, the build scenario lead to noise levels that were higher, but only by 3dBA or less (except at one small row of houses where it was about 5-10dBA). The build scenario would result in higher noise levels, but the noise levels could not be said to be *significantly* higher.

The community representatives wanted a definition of what constituted a significant noise impact. The convenor of the community consultative meetings suggested that the definition be either in terms of change in noise level, or change in traffic volume. In the end, we gave him neither.

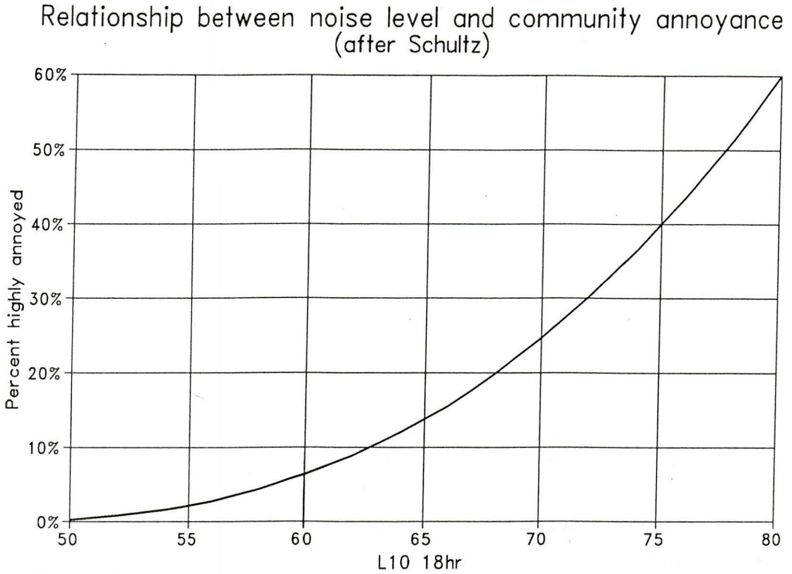

Figure 1

The story so far

We had to find an assessment approach that:

- acknowledged that short-term changes in traffic volumes may have a greater impact than gradual changes, provided such an impact was long-lived
- contained a definition of what constituted a significant impact
- directly addressed the problem of truck noise and sleep disturbance
- was firmly based on current (or well established) research.

Impact of sudden changes in traffic volumes

In a benchmark review paper published in 1978, Ted schultz[1] proposed an approximate relationship (shown in Figure 1, with L_{DN} converted to $L_{A10,18hr}$, that has now become widely used in prediction of community response to long-term steady noise exposure.

Schultz gave us a background against which to examine community response to sudden changes in traffic noise exposure. Several studies have shown that changes in noise exposure which occur over a short time period can lead to community annoyance levels that do not correspond with the predictions based on long-term steady noise exposure. In a study we undertook in 1989 of residents' attitudes to noise in areas adjacent to the South Eastern Arterial (47 residences), we found that of those surveyed, 51% were highly annoyed. The $L_{A10,18hr}$ at the residences where the survey was conducted varied from 55-68dB(A), and annoyance scores did not correlate with noise levels. At these noise levels, Schultz would predict 5%-20% highly annoyed. This was two years after the South Eastern Arterial was opened.

C J Baughan and L Huddart[2] of the Transport Research Laboratory in England, in a study of the effect of changes in traffic noise exposure on levels of community annoyance, found that community annoyance levels were more closely related to the change in traffic volumes than to the change in noise levels. This suggests that the community perception of noise intrusion is not just de-

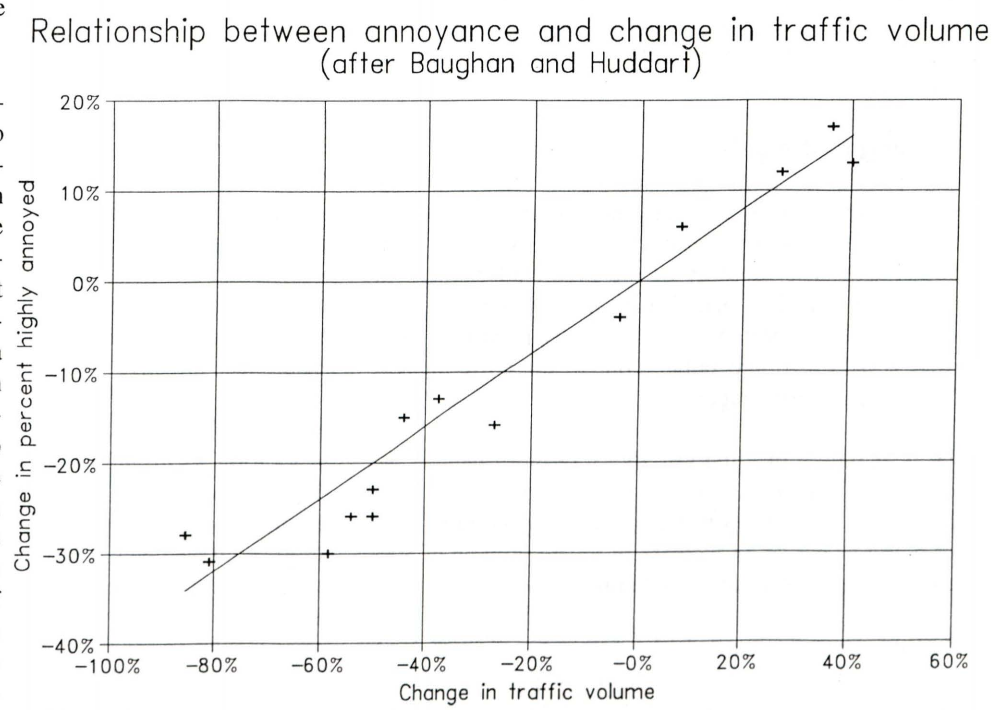

Figure 2

pendent on the noise level, but somehow includes an awareness of the traffic volume. Of the 14 sites studied, 9 experienced a decrease in traffic noise as a result of bypass construction and 5 sites experienced an increase in traffic noise exposure.

Figure 2 shows the relationship they found between change in community annoyance and change in traffic volume, with "mean dissatisfaction" converted to percent highly annoyed.

As to whether the effect persists, Baughan and Huddart say

> *The excess change in nuisance ratings appears to be more than simply a short term effect... other studies have found little or no evidence of adaptation to traffic changes.*

One study by I.D.Griffiths and G.J.Raw[3] did show some regression toward steady-state annoyance levels. They found that the levels of community annoyance remained steady for up to two years, and that after 7-9 years only about 60% of the effect had disappeared, implying that the time taken to return to steady-state annoyance levels would be about 20 years.

Truck noise and community annoyance

A direct relationship between truck volumes and community annoyance has been found by Dr Ragnar Rylander of the Institute of Social and Preventative Medicine at the University of Geneva, Switzerland[4]. This relationship is shown in Figure 3 for two groups of data, one with a typical L_{max} of 86dB(A) and the other with a typical L_{max} of 93dB(A).

Rylander has found that above the threshold of approximately 1,000 trucks per day, there is very little change in the level of community annoyance. So to achieve a reduction in annoyance, the number of trucks per day must be reduced to somewhat below 1,000.

Relationship between truck volume and community annoyance (after Rylander)

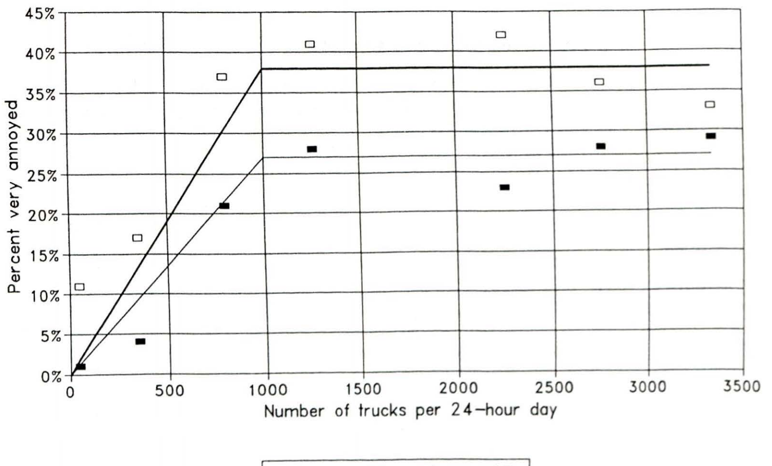

Figure 3

Sleep disturbance

Research by Rylander[4] supports a medical approach to the analysis of the relationship between noise exposure and sleep disturbance. He relates sleep disturbance to the number of noisy events in a night, and the maximum noise levels of those events. In the *Western Bypass Health Impact Study*[5], Rylander provides us with the following useful rule of thumb:

> *Sleep is likely to be disturbed with a rather low number of passes of 10-15 heavy vehicles at about 55dB(A) [measured indoors, per night].*

This approach is supported by work by Dr Barbara Griefahn[6] of the Institute for Occupational Health of the University of Dortmund, Germany, which shows that at this noise level, actual awakenings begin to occur, not just sleep disturbance. She has found that if there are more than 10-20 noisy events in a night, awakenings will occur in up to 10% of the population if the maximum noise level exceeds 54dB(A). Sleep disturbance occurs if maximum noise levels exceed 47dB(A).

Michael Vallet of the National Institute for Transport Research in France[7] recommends the following internal noise level criteria. For traffic noise, the recommended internal noise levels in a bedroom are:

L_{eq}	35dB(A)
L_{max}	50dB(A)

Defining significant impact

Our current understanding of the way in which the environmental, social and health effects of traffic noise arise is through interference with behaviour (rather than any direct physiological effect). The interference with behaviour is annoying and so we can determine the magnitude of the effect of the noise by assessing how annoying the noise is to the community.

Using the research results described above, we were able to predict the likely change in community annoyance due to the construction of the Western & Southern Bypasses. This meant we could assess the impact more directly than if we were to merely assess changes in noise levels.

It was important that the difference between the noise impacts of the build and no-build scenarios be assessed as significant or not. We chose to define significant impact in the following way:

> *A significant noise impact is a change in the noise environment that leads to a change of 5% or more in the pro-*

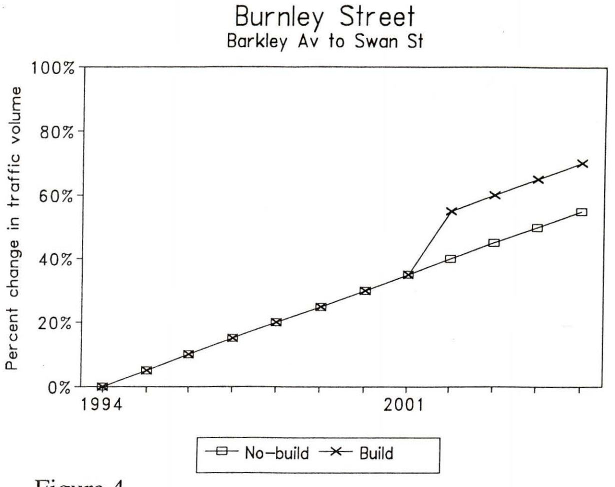

Figure 4

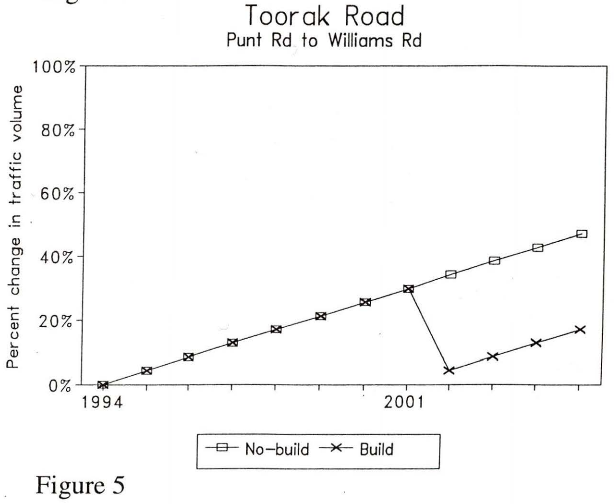

Figure 5

portion of the community who are highly annoyed.

Impact assessment methodology

Comparison of traffic volumes for the build and no-build scenarios was straight-forward. Doing nothing would lead to steady increases in traffic volumes on most roads in the study area. This steady increase would also occur for the build option, until the bypasses were opened to traffic. The difference between the build and no-build scenarios would occur on opening.

If the Western and Southern Bypasses were built, there would be two types of change occurring. The first of these is a change in noise environment due to exposure to noise from new roads. This would apply to residences near the proposed route of the Western Bypass.

The other type of change which would occur if the bypasses were built would be a change in noise exposure due to a change in traffic volumes on existing roads in the areas affected. This change would occur quite suddenly over a matter of 4-6 months after the Bypasses opened. This effect is shown in Figures 4 and 5, which illustrate the traffic volume increases predicted for Toorak Road and Burnley Street.

Five aspects of the change in noise environment if the bypasses were built were considered. These were:

Change in traffic volumes on existing roads
Based on the work of Baughan & Huddart2, it would appear that an increase or decrease in traffic volume of 13% would be significant.

Change in daily truck numbers
The work of Rylander[4] would indicate that if the number of trucks per day before and after the opening of the bypasses is greater than 1,000, then there would not be a change in the level of community annoyance. If the number of trucks per day before or after the opening is less than 1,000, then it would appear that a change in the number of trucks per day of 17% would be significant.

Change in the number of noisy events at night
Based on the work of Griefahn[6] and the recommendations of Vallet[7], it would appear that sleep disturbance can be related to the number of times 50dB(A) is exceeded internally during the night. If there is no change to the maximum noise levels, then there would be no improvement unless the number of noisy events drops below Griefahn's break-point of 10-20 events per night.

Changes in the $L_{A10,18hr}$ due to new roads
The work of Baughan & Huddart[2] would indicate that any increase in the $L_{A10,18hr}$ due to a new road would be significant. They found changes of 10% in the percent highly annoyed even when the increase in the $L_{A10,18hr}$ was less than a decibel.

Exceedances of 68dB(A) $L_{A10,18hr}$
Noise levels higher than this criterion are defined as "unacceptable" by the OECD[8]. This was included as a criterion for determining a significant negative impact in order to avoid the problem of accumulated small impacts. Several locations in the study area had been subject to increasing levels of traffic noise in the last decade or so, and it was felt that a line had to be drawn somewhere.

Exposure to New Roads

For areas exposed to new roads, three aspects of the change in noise environment if the bypasses were built were considered. These were:

- changes in the $L_{A10,18hr}$
- changes in the number of noisy events at night
- exceedances of the OECD criterion of 68dB(A) $L_{A10,18hr}$.

Noise levels were predicted using a computer traffic noise prediction model developed by Carr Marshall Day Associates which is based on the CoRTN method.

Changes to Existing Roads

For areas exposed to changes in traffic volumes on existing roads, four aspects of the change in noise environment if the bypasses were built were considered. These were:

- change in traffic volumes
- change in daily truck numbers
- change in the number of noisy events at night
- exceedances of 68dB(A) $L_{A10,18hr}$.

Noise impact assessment results

The study area was broken down into 11 areas which varied greatly in size. The largest was about 1-2 square kilometres and the smallest was one little street of about 10 residences. For each area, the impact was assessed according to the approach outlined above. Table 1 shows a summary of our conclusions. The areas that have the most affected residences are marked with an asterisk (*). Options A, B and C refer to options for locating the tunnel portals for the Southern Bypass.

When considered together with the construction of new noise barriers, and the provision of noise insulation at Debney Park Estate, the overall noise impact of the Western and Southern Bypasses was considered to be positive.

Having chosen an assessment approach - and a definition of significant impact - based in terms of community annoyance rather than noise levels, we arrived at a conclusion that made sense intuitively. Areas that were clearly going to be worse off were shown to be so, and areas that were going to experience 50% decreases in traffic (and even more in truck traffic) were shown to have significant positive impacts in a way that was much more realistic than "3-5dBA decrease in noise level, hardly noticeable".

Table 1 Summary of Conclusions

Area	Impact
Tullamarine Freeway*	Positive, when new noise barriers are considered.
Debney Park Estate*	Negative, unless noise insulation is provided.
Bent St	Negative.
Hardiman St	Negative.
Flemington & Ascot Vale	Minimal, but distant noise exposure may be a problem.
North Melbourne*	Positive, but distant noise exposure may be a problem.
Railway Place	Minimal.
South Melbourne	Positive, depending on residential development.
Richmond*	Positive, if new noise barriers are provided, especially if Option B or C are built.
South Yarra	Positive, if Option B or C are built.
South Eastern Arterial*	Positive, if new noise barriers are built.

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Sensitivity to Road Traffic Noise

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Abstract

A study is being carried out with the object of developing sensitivity criteria for traffic noise from roads in residential areas and evaluating which individuals, if any, adapt to a change on the noise environment. This proposes an individual model of the response (individual noise tolerance) and goes on to discuss the problems of undertaking a survey to establish an individual's sensitivity as well as determine the extent of adaptation to noise.

Introduction

There are three aspects of studies of the response of people to traffic noise. They are: (1) the study of subjective annoyance to traffic noise as a function of traffic noise measures as well as other non-noise parameters, (2) the study of interference with various activities and possible forms of health impairement, and (3) an evaluation of the merits of various noise measures as predictors of adverse responses (Bradley and Jonah, 1979). These aspects can be termed annoyance, disturbances, and noise measures. A large number of noise surveys have been reported in the areas. As Fields and Hall (1987) mentioned in the literature, low-investment research programmes may not provide any more useful information about the dose-response relationship than is available in existing research publications and so the initial part of the present study has been to evaluate previous studies.

A number of studies of traffic noise have been undertaken in which measurements of sound pressure level have been related to residents' annoyance with the object of establishing tolerable noise levels or community reactions (i.e. the percentage of respondents adversely affected by noise of a given level). The development of noise control criteria depends upon specifying what is tolerable to the majority of the population. However, noise criteria, derived from such procedures, have not been well established. A particular difficulty which has arisen in connection with the development of noise and annoyance criteria has been that of subjective scaling. The scales used have failed to predict individual responses with sufficient accuracy.

Noise level is known to correlate with annoyance. However, reactions to changes in noise environments have been different. Langdon and Griffiths (1982) found that reductions in noise results in changes in annoyance, whereas Fidell and Jones (1975) reported no change in residents' reactions after a severe reduction in night-time aircraft noise levels. In most studies of the effects of noise on communities, measures of noise exposure incorporate corrections for certain time operations.

Individual dissatisfaction scores correlate poorly with physical measures (Griffiths and Langdon, 1968), because of wide individual differences in susceptibility and experience of noise, as well as in patterns of living likely to be disturbed by noise. There are many factors in addition to the noise measures which affect subjective annoyance to traffic noise, and attitude surveys have shown that the correlation between individual annoyance and noise level is relatively insensitive to noise measures (Ollerhead, 1973). Edwards (1975) also indicated that it is not noise measurements and noise indices that cause the poor correlation, but measures of human response. It has been found that there are considerable differences between people in how they react to the same level of noise (Moreira and Bryan, 1972). Therefore, it is necessary to pay special attention to individual differences when selecting noise criteria for residential areas. The study reported here yields an assessment of an individual's tolerance of noise investigated by a self-administered questionnaire. The results were analyzed to evaluate a number of subjectivity scales, in terms of subjects' responses about their disturbances over the road traffic noise.

Background

The acoustical factors affecting annoyance are; numbers of individual noise events (Fields, 1984), day/night-time noise levels (Fidell and Jones, 1975; Yeowart et al., 1977; Langdon and Buller, 1977; Bradley and Jonah, 1979; Nemecek et al., 1981), residential noise environments (ambient noise) (Griffiths and Langdon, 1968; Robinson, 1971; Rice, 1977; Schultz, 1978), and changes in noise environments (Fidell and Jones, 1975; Langdon and Griffiths, 1982). The factors causing variations in individual reactions to noise are: time spent at home, the differences in noise levels at home, living style (outdoor/indoor, window opening, etc.), type of housing, timing and location of indoor activities, other indoor noises, etc. Other individual factors associated with variance in annoyance which can not be explained by the acoustical factors are; personal attitudes and demographic characteristics:

Annoyance is generally higher for people who are fearful that some danger to themselves or other people in the local area may be associated with the transportation activities which they can hear. The attitude of 'fearfulness' has proved to be important for road traffic noise as well as aircraft and railway noise. Things disliked about the area from neighbourhood evaluation and evaluations of the quality of the public services are related annoyance (Langdon, 1976). The people who believe that their health is affected by noise from the particular source are also likely to be annoyed by the source. Other aspects of the noise source's intrusion related to residents' evaluations of the noise in the area are: dirt, dust, lights, odours, visual intrusion, loss of privacy and severance by a right-of-way (Fields and Hall, 1987).

The variables in demographic characteristics have not been found to be consistently associated with noise annoyance. Sensitivity to annoyance by noise (or noise annoyance susceptibility) does not appear to depend upon personal factors such as age, sex, education, job responsibility (Moreira and Bryan, 1972), length of residence (Nemecek et al., 1981) and type of housing (Bradley and Jonah, 1979).

'Adaptation' is another matter to be reviewed in relation to any sort of annoyance surveys to road traffic noise. How the annoyance of an individual changes with time (adaptation) is also important and is one of the main objects of the present work.

Outline Of Survey

The main problem of choosing a sample for most surveys is normally one of eliminating as much bias as possible. As the survey variables are interrelated and it is difficult to provide strong evidence for the nature of the causal relationships between variables, cautions have been made in survey plans and findings about sensitivity and disturbances:

(1) If the sensitivity is correlated with noise level then part of the relationship of the sensitivity with noise annoyance may be caused by the noise level effect.

(2) The sensitivity variable must be investigated with a question which is clearly distinct from a noise rating or disturbance question.

(3) Some of the high correlation between annoyance and these disturbances should be discounted because both are measured at the same time in a single questionnaire under similar conditions and thus may be subjected to such correlated errors in measurement as 'response set' (the tendency for people to give answers that follow the form rather the content of the question).

(4) Also of concern is that the position of the 'disturbances' section may suggest that the medical symptoms are attributable to road traffic noise.

(5) In order to predict an individual's annoyance to a particular noise it is necessary to know not only the level of the noise but also his/her personality (Moreira and Bryan, 1972). Certain personality traits are responsible for differences in noise annoyance sensitivity.

(6) Models for biases in judging sensory magnitudes should be eliminated.

The questionnaire dealing with residents' attitudes and opinions consists of; (a) various sorts of disturbance which might be expected to result from noise, (b) a scale of dissatisfaction with the acoustic environment, © a study of nuisance caused by noise from motorways (room usage, sleep disturbance, preferred siting of the house in relation to the road, and seasonal or meteorological effects on the perceived noisiness of traffic), (d) sources of noise nuisance, (e) a scale of susceptibility to noise nuisance, (f) demographic characteristics.

A field survey

This investigation is to have a repeated measures design; the same respondents will be interviewed 6 and 12 months after the first interview.

Hypotheses

It seems that there are three reasons for variation in annoyance to noise: situations of the environmental stressor (noise source), differences in situation-specific attitudes (non-auditory effects of noise in naturalistic settings) and personality difference (sensitivity).

1) There are individual differences in adaptation to road traffic noise.

2) People's sensitivity to noise varies in incidence.

3) Noise sensitivity, noise rating and disturbances are three major attribution to noise and have a relationship.

Questionnaires

The survey will be presented to the residents as a survey aimed at discovering the attitude of the population to housing and housing areas. The questionnaires, which in relevant respects are identical, commence with questions concerning the following;

1) Housing conditions:

2) Respondents' backgrounds:

3) Questions concerning the attitude to noise (SENSITIVITY TEST): (Indirect questions concerning annoyance by traffic noise) An instrument of 21 questions using a Likert-style format developed by Weinstein (1978) is used to assess self-reported sensitivity to noise. Most items are presented on a 6-point scale ranging from 'agree strongly' to 'disagree strongly'.

4) Direct ratings of the amount of noise (Perception of noise):

5) Questions concerning disturbances caused by traffic noise and residents' various manifestations:

Discussion

Noise level is not the only variable that can be used to control annoyance levels over long periods of time. There is enormous individual variability in the measured reactions at any one noise level due to acoustical, situational, attitudinal and personal factors including measurement errors. Unless the noise annoyance research discovers any step that might be taken to consider the privately felt annoyance with persistent road traffic noise, policy makers cannot ever hope to reduce public complaints against noise except that of reducing noise levels dramatically.

Acknowledgement

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Sound Design - a Multi-Dimensional Task in the Automobile Industry

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Abstract

Although the car is primarily a means of transport, it also constitutes a significant sound source in which each vehicle creates a concert of various individual sound sources. These can be heard both inside and outside the car. The engines ignite at regular intervals, various gear wheels interact in the gear transmission, tyres vibrate in contact with the road surface, the airstream flows around the car body, and a lot of individual aggregates such as dynamo, servo and petrol pump round off the sound pattern. In an orchestra an optimum interplay and harmonic tuning of the individual instruments are also important for a good sound impression; the same goal whose importance the automobile industry has become increasingly aware of in recent years. It is not enough to reduce vehicle noise to meet benchmark values, but questions relating to sound comfort, sound design and pleasant sound are becoming more and more important.

Sound does not only mean acoustic vibration, but transmits information of a more or less pleasant or unpleasant character, depending on the listener's expectation. Sounds with high energies impair human hearing, but create even at low energy totally different impressions (annoying, unpleasant, pleasant and so forth), depending on the temporal structure and spectral distribution.

Therefore the task of sound design in the automobile industry is multi-dimensional:

- The sound level must not exceed legal values (physical aspect).
- The sound should not leave an unpleasant or annoying impression (psychoacoustic aspect).
- The sound has to harmonise with the car type (design aspect).
- The sound must meet the customer's requirements (cognitive aspect).

Introduction

Regulations for determining noise levels are based on A-weighted SPL measurements performed with only one microphone. This method of measurement is usually specified when determining whether the ear could be physically damaged. Such a simple measurement procedure is not able to determine annoyance of sound events or sound quality in general.

A loudness measurement which takes account of the spectral and temporal structure of a sound event, better than the A-weighted SPL, has advantages. Nevertheless, it is not universally accepted because the loudness measurement does not allow a complete judgment of complex sound events comparable to human hearing. In addition to loudness, there are other psychoacoustic parameters such as sharpness, roughness and fluctuation strength.

For some years investigations with binaural measurement and analysis techniques have shown new possibilities for the objective determination of sound quality. By using Artificial Head technology in conjunction with psychoacoustic evaluation algorithms - and taking into account binaural signal processing of human hearing - considerable progress has been made for determine the sound quality of sound events[1,2].

This technique has been supplemented by a new calculation method for signals which are complex in terms of signal theory:

The analysis of a sound like door slamming is relatively complicated because its time duration is very short and its frequency spectrum very broad. In strong contrast to a normal FFT analyser, human hearing has a high resolution in the time and frequency domains. This is why an objective evaluation of a slamming door, corresponding to subjective classification values such as loudness, fluctuation strength and roughness, is insufficient, being based only on simple, quasi-stationary test signals. An additional calculation, based on three new types of signal processing, in addition to the Wavelet transformation, can be used to improve objective analysis of the sound of door slamming.

Binaural measurement technology

Human hearing differs in many respects from conventional sound measuring systems. The outer ear is a directional filter which changes the sound pressure level at the ear drum by +15 to -30 dB, depending on frequency and direction of sound incidence. These filter properties of the outer ear are due to diffraction, reflections and resonances caused by the pinna and concha geometry. Human hearing has two paths - the left and right ears. This capability permits binaural signal processing and pattern recognition in conjunction with spatial hearing, selectivity and noise suppression.

The auditory impression is not only determined by the sound pressure level, but also by psychoacoustic properties. Depending on the time sequence of the signals or spectral distribution, there can be different subjective impressions due to pre-, post- and simultaneous-masking properties of the hearing mechanism.

Human hearing involves very complex signal processing but it has very short memory. By faithfully recording, digitally storing and reproducing a sound event, the comparative human ear equivalent judgement of different sounds becomes feasible and can be documented. The psychoacoustical properties of human hearing determine the subjective impression of sound events such as noise annoyance. Also, both sound pressure level and loudness are important. Standard methods of measuring loudness do not consider the binaural signal processing that occurs in human hearing. This explains the poor correlation between measured sound levels and the subjective impression of noise. Since the evaluation of sound quality by large numbers of test subjects usually produce similar results, the sounds must have physically measurable properties which can be correlated to subjective impressions. This suggests there are properties that have not been identified and accounted for by conventional measurement methods. Objective acoustic measurement methods used up to the present time have not provided sufficiently useful algorithms for evaluation of individual sources in a complex sound field. An artificial head measurement system that emulates human hearing as the "receiver" has been introduced for the judgement and analysis of sounds. The artificial head measurement system in binaural measurement method can be used in all fields where acoustic emissions (acoustic energy radiated by a source) and immissions (acoustic energy incident on a receiver) must be determined or where sound serves as an indicator of comfort, quality and safety.

Figure 1 illustrates how recorded information from an artificial head can be evaluated with the help of digital signal processing to identify and eliminate annoyance. Binaural digital signal processing within a computer controlled measuring system allows input, storage and processing of an event and listening to selected segments which can be continuously repeated without audible artefacts. The measuring system displays both right and left ear signals in the time and frequency domains. The sound segment sample can then be directly manipulated with the result being displayed and reproduced by headphones in "real time". Euphony is evaluated as part of the analysis; that is calculations of roughness, sharpness, pitch and timbre as well as loudness, that take into account the ear's pre-, post- and simultaneous-masking properties.

Combined analysis of spectrum and time structure

As is well-known, human hearing not only performs spectral analysis, but also the evaluation of the envelope from the signal. The ability to distinguish between simultaneously occurring sounds can be modelled using band pass division. The frequency resolution obtainable depends essentially on the bandwidth of the hearing-related filters. Modulations are primarily recognised from changes in specific loudness level over time, i.e. from the non-linear, distorted lowpass filtered envelope curve of individual bandpass signals.

The discrete Fourier transform is often applied for short-term spectral analysis of acoustic signals. Since in this technique, the same window is applied for the investigation of all relevant spectral components, the modulation remains constant over the whole frequency domain. Definition of the analysis window simultaneously defines time and frequency modulation. This kind of spectral analysis allows either high frequency modulation of low-frequency signals (long window), or high-resolution analysis of timestructure in the upper frequency range (short window), and this makes it less suitable for a perception-oriented signal description. Thus, selection of window length always represents a compromise between these various demands.

These considerations resulted in the development of the "variable" Fourier transform. The basis of this aurally-equivalent spectral analysis in efficient sub-band division (using filters with hardly any overlap). The spectral composition of the bandpass signals is subsequently investigated by applying analytic procedures of various length (adapted to human hearing). A combination of an appropriate filter bank makes it possible to approach the resolution of human hearing in a series of steps.

Other hearing-oriented methods aim at spectral representation as a quasi-continuous function of time (in contrast to block-by-block processing) with hearing-related frequency modulation. These generally more time-consuming techniques can be represented as bandpass division of the signal under investigation, where the impulse responses applied are of different length. The basic differences between these various approaches include which type of filter and which bandwidth as a function of the filter center frequency are selected. The Fourier T transform, as a short-time spectral analysis with an exponentially decreasing weighting factor of proceeding values, has been known for some time (e.g. 4, p141 ff, 5,6). The essential advantage of a "filter bank principle" is that time constants, analogous to frequency selectivity in human hearing can be individually selected for individual analysis filters.

In the case of continuous Wavelet transform a time signal is also analysed using bandpass filters of various width. The Wavelet transform of time-discrete signals can be interpreted equally well as a special form of subband division. It may also be understood as octave band filtering with perfectly reconstituting filters. The fact that various earlier approaches to signal definition with variable frequency and time resolution can also be represented in terms of WT opens the way to a uniform definition and thus adds to the development of new ideas.

Auditory model according to Sottek

Human hearing perceives slight differences in frequency and rapidly changing time structures simultaneously. Any mathematical analysis applied exclusively in the frequency and time domains is therefore unable to provide aurally-equivalent results. An essential feature of the auditory model developed by Sottek[7] is calculation of excitation distribution over time. This is obtained as a function of two variables (frequency and time) from the curve of the sound pressure function and provides a basis for calculating psychoacoustic values. The model for calculating excitation distribution according to Zwicker and Feldtkeller[8] is limited to steady signals, since only spectral data are evaluated. Furthermore, the phase of individual vibrational components is neglected, which results in an inadequate simulation of the excitation due to several superimposed sound components: The spectrum of a sinusoidal, amplitude-modulated tone is composed of three neighbouring spectral lines. Ignoring the phase angles, the excitation distributions due each of the three individual components would produce a temporal constant running of the total excitation, although an amplitude modulation has, of course, a characteristically pronounced time structure.

Modulated tones or noise signals result in an impression of roughness, depending on their frequency and the modulation frequency. In determining this psychoacoustic value the time structure of the excitation is of particular significance and must be taken into account. When various sound components are contained within a given critical band, their phase has a particularly visible effect, because in this case separate processing of the various components in different channels is not possible. If the phase of the sidebands of an amplitude modulation is rotated through 90°, this is only perceivable if the physical spectrum is contained within a single critical band. The time structure of a (longer) signal segment is determined by phase data. Any change to the phase relationship affects the amplitude response of the time signal. If the amplitude spectrum is determined at sufficiently short time intervals, "phase" becomes less significant. This is taken account of in short-time spectrum analysis, by dividing the signal into weighted blocks of defined length and then Fourier transforming the seg-

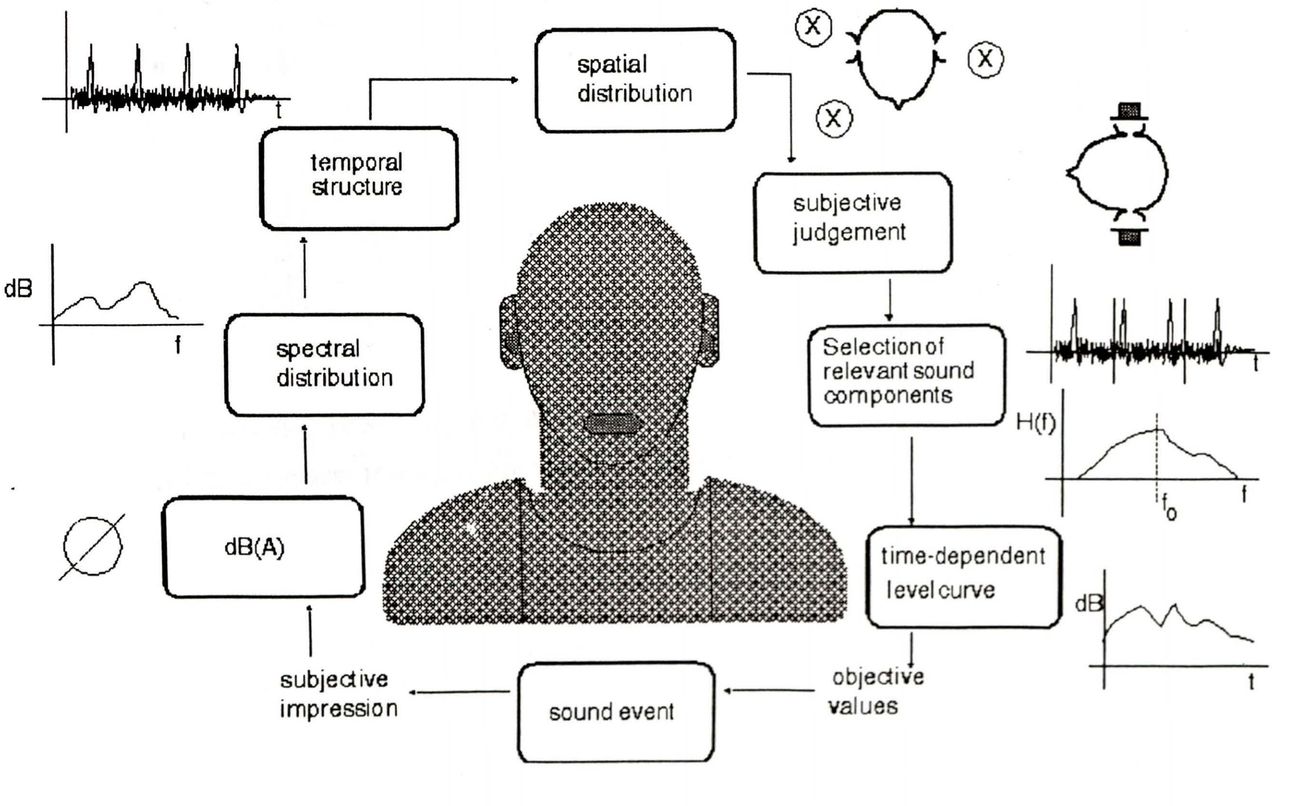

Figure 1: Subjective and objective diagnosis, analysis and clarification of sound.

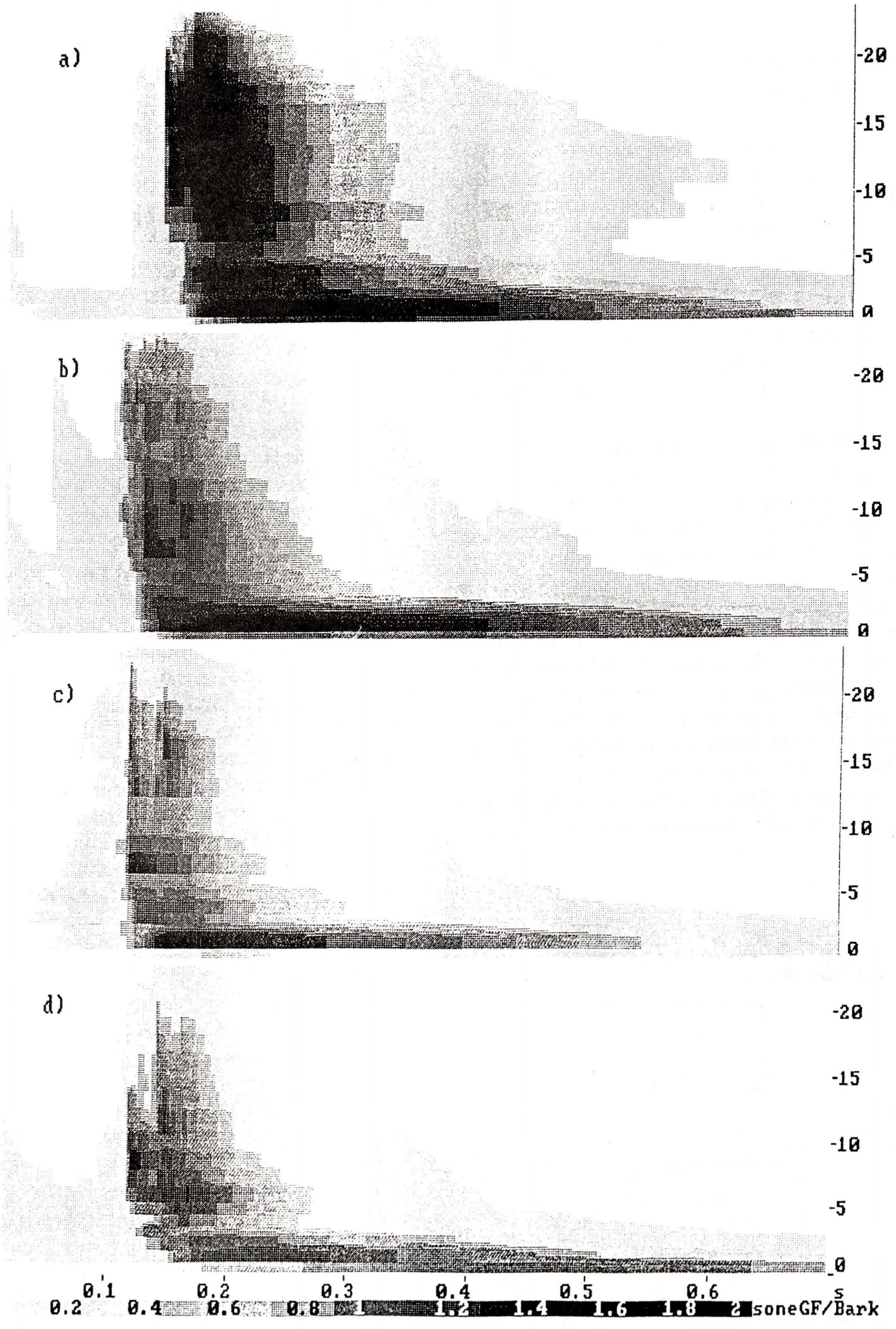

Figure 2: Specific loudness analysis (a) Door 1, (b) Door 2, (c) Door 3, (d) Door 4.

ments. This results in a meaningful representation of the amplitude distribution and of the time as a function of frequency. Admittedly, the equidistant frequency modulation resulting from this technique does not take into account the resolution of the human ear.

The procedure developed by Sottek does take into account the time and frequency resolution of human hearing, i.e. enables appropriate processing of non-steady signals.

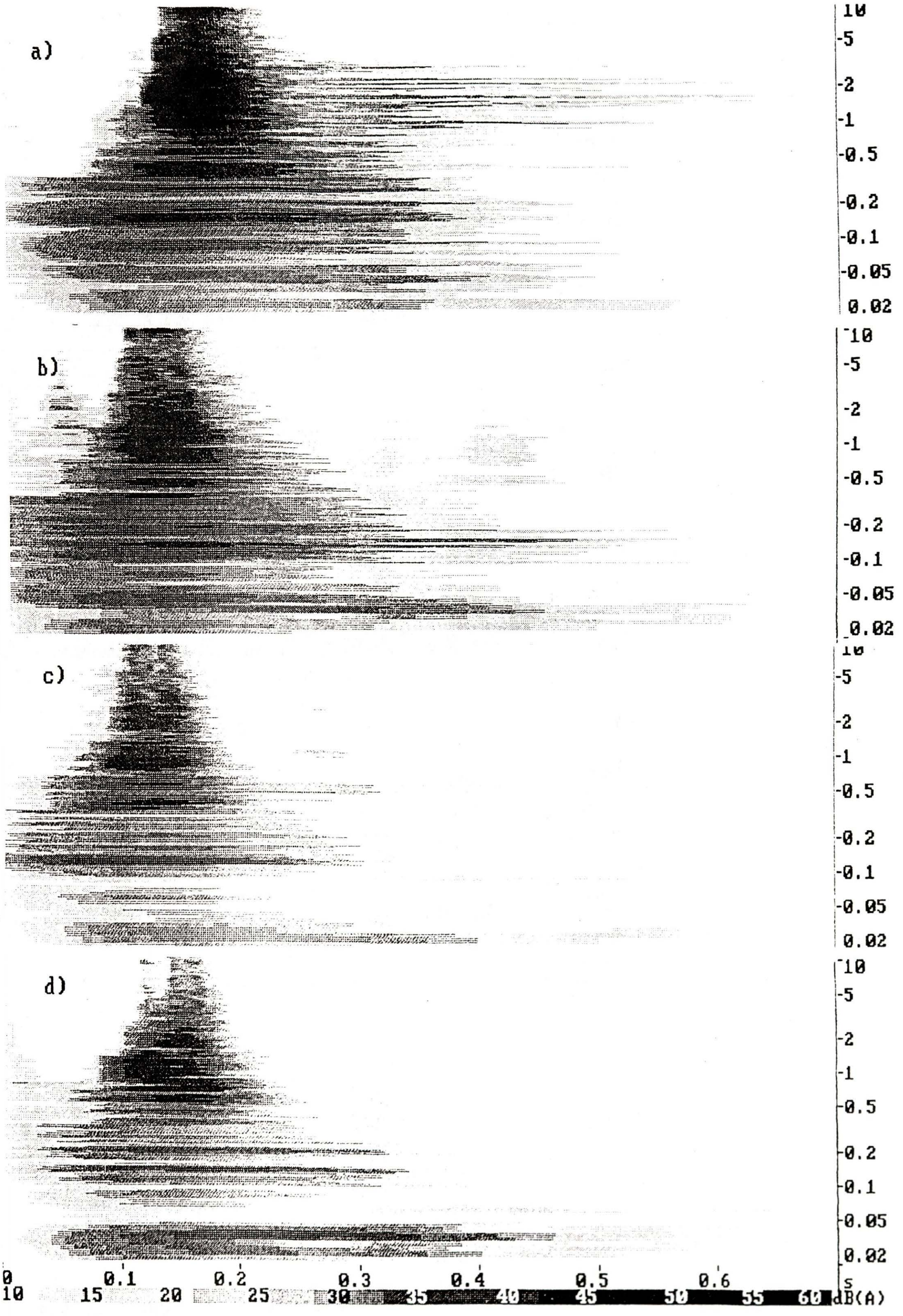

Figure 3: Variable Frequency Resolution (VFR). (a) Door 1, (b) Door 2, (c) Door 3, (d) Door 4.

Application in door slam noise

It is hard to describe the door slam noise in terms of signal theory, because this signal is very short and is based on broad-band excitation. However, if the signal analysis does not make possible a sound evaluation according to human signal processing, an objective determination of the subjectively perceived sound quality of the door slam noise becomes more difficult. Investigations of four different door slam noises will be represented in the following. The investigations were made in an anechoic chamber with an Artificial Head Measurement System. The system was positioned 1.5 m off the left front door with its face towards the door and recorded the sound event faithfully to allow both a subjective and objective

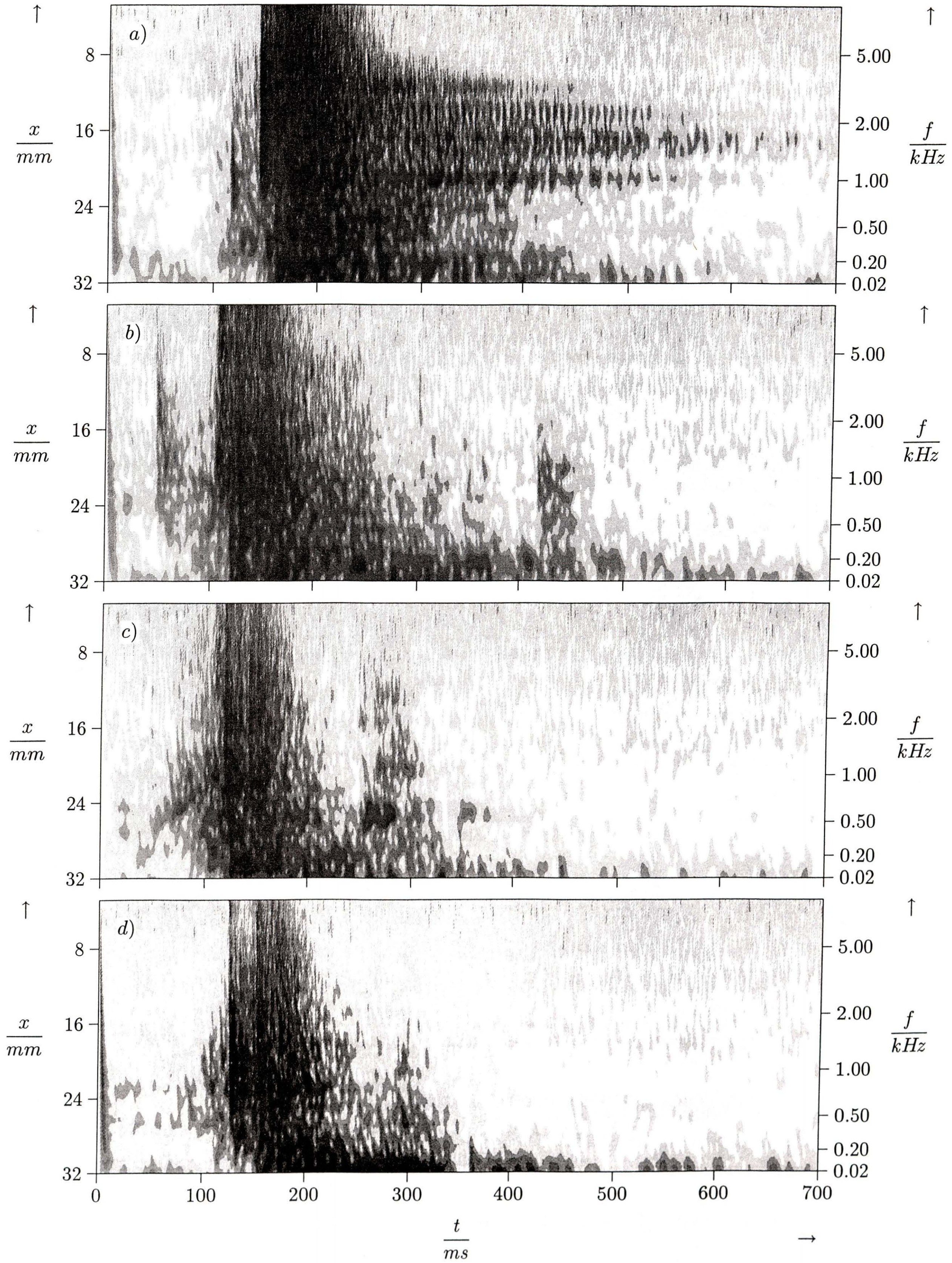

Figure 4: Analysis with Sottek inner ear model. (a) Door 1, (b) Door 2, (c) Door 3, (d) Door 4.

evaluation. Four different vehicles of different classes producing widely differing door slam noises were evaluated. Due to the great differences in sound quality the listeners determined a clear priority order in the subjective listening tests. The four doors (Door 1, Door 2, Door 3 and Door 4) are represented below in the order of improved sound quality.

Previous considerations showed that a simple FFT analysis is not suitable for this kind of analysis. Figure 2 shows the relationship of specific loudness in dependence on time for the four doors. This representation and analysis are not suited completely to show the subjective impression adequately. In case of Door 1 the increased loudness at higher frequencies becomes evident which is - among other things - responsible for the negative judgment of the sound. Further classifications of the individual door noises are, however, not possible. It does not become apparent, particularly in the case of Door 3, that a clear echo by slamming the interior ventilating flap could be heard. The subjective evaluations showed that Door 4 was preferred due to strong levels in the low-frequency spectral range. This does not become obvious in Figure 2. Here, it is disadvantageous that the curves of the same loudness according to Zwicker[8] are based on listening tests with stationary signals. A statement for short-time impulse like wide-band sounds cannot be made easily.

Figure 3 shows the result for the variable frequency resolution VFR (comparable to Wavelet). The temporal structures of the door slam noise are striking: the long decay in the frequency range of 800 Hz up to 3500 Hz in case of Door 1, the time structure due to the non-uniform contact of individual components in case of Door 2 as well as the discrete echo in case of Door 3. At the same time a high frequency resolution of up to 20 Hz is guaranteed, the strong boominess of Door 4 as compared to Door 3 becomes obvious. An adequate analysis and representation according to human signal processing is shown in figure 4 and is based on the hearing model for monaural signal processing developed by Sottek[7]. At the left hand-side the length is scaled according to the basilar membrane and at the right-hand side the corresponding frequency ranges.

Figure 4 shows the relationship of excitation distribution and time. For higher frequencies the representation and clarity regarding spectral distribution and the respective temporal structures is similar to the VFR analysis represented in Figure 3. In the frequency range below 500 Hz with the same good frequency resolution a significantly better reproduction of the temporal structures becomes apparent. Door 4 has the highest energy in the low-frequency spectral range when the door is slammed. This energy is produced at the same time as the entire door slam noise. In case of Door 3 the maximum in the low-frequency range exhibits a pronounced narrow-band characteristic and has a temporarily shifted structure.

Summary

In many sound situations it is necessary to make faithful recordings using artificial head technology in order to consider selectivity of human hearing on the one hand, and to take into consideration psychoacoustic parameters for judging sound quality in case of phase and level differences on the other hand. The improved method to mathematically describe psychoacoustic parameters according to Sottek allows a more accurate objective determination of sound quality. While questions as to the physical damage of hearing can be answered by the A-weighted SPL, the evaluation of sound quality is possible only in conjunction with psychoacoustic parameters. Using the door slam noise as an example it was possible to present an optimum analysis comparable to subjective evaluation.

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Session 2B

Audition

6000 Hz - Is It an Early Indicator of Noise-Induced Hearing Loss?

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Abstract

It is commonly reported that the effects of noise-induced-hearing loss are centred around the audiometric frequency of 4000 Hz.[1] The typical noise-induced hearing loss is reported as showing a dip at this frequency with improved thresholds at 8000 Hz. However a significant notch at 6000 Hz was reported for all mean audiograms in a study of the hearing characteristics of 89,500 otologically normal, noise-exposed Western Australian workers and it was suggested that this may be the frequency which is most susceptible to noise damage.[2] The present study examined this notch further to determine if it is due to artefact or noise damage. The audiometric test results of 68 otologically normal, not noise-exposed students aged 13-18 years were compared to 1,591 otologically normal, noise-exposed workers aged 16-19 years. The study showed a distinct "noise notch" at 6000 Hz for the noise-exposed subjects which was not present in the not noise-exposed subjects. There was a statistically significant difference ($p<0.001$) between the mean threshold levels at 6000 Hz for both groups, in both ears. The results of this study suggest that 6000 Hz may be the frequency most susceptible to noise damage and may therefore be a good early indicator of noise-induced hearing loss. The implications of this finding are discussed in comparison with other research which suggests that the notch at this frequency may be due to artefact.

Introduction

It is well known that noise in the workplace is a significant hazard to the hearing health of workers.[3,4,5,6,7]. Examination of the causes of hearing loss in Australia indicate that exposure to loud noise in the workplace may be a primary cause of hearing impairment among adult Australians[8]. Exposure to loud noise was reported to be the main cause of hearing loss for male subjects in a survey of the hearing health of South Australian adults[9]. The Australian Bureau of Statistics in 1988 reported that for Australians with a hearing impairment, work was reported to be the second main cause of hearing loss[10] and in 1993, for persons with a hearing impairment, work was reported to be the main cause of hearing loss[11]. Occupational noise-induced hearing loss is reported to be the most prevalent compensable disease in Australia[12]. In NSW in the 1992/93 financial year 64.5% of occupational disease claims were for noise-induced hearing loss, costing $54,470,000[13].

Prevention is the best weapon against occupational noise-induced hearing loss and employing routine audiometric testing, as part of a hearing conservation programme, is the best method of identifying if occupational health and safety practices implemented to control noise as a workplace hazard have been successful. Identifying changes on an audiogram is the only widespread method currently in place which acts as the first indication that the prevention techniques are not working. If hearing impairment can be detected as soon as it occurs action can be taken to prevent further hearing loss.

It has long been reported that noise-induced hearing loss is characterised as a having a distinctive dip in the audiogram at 4000 Hz[1,14,15,16,17,18]. This 4000 Hz dip has been classified as the hallmark which links noise as the aetiology of a sensorineural loss.[19] With further exposure, the increase in threshold level extends to the adjacent frequencies broadening and deepening the 4000 Hz noise notch[1,16,20].

Recent studies however suggest that 6000 Hz may be the frequency most susceptible to noise damage and the 4000 Hz notch an artefact of previous studies not testing at the 6000 Hz frequency[2, 21,22,23,24,25].

Monley et al[2] in a survey of the hearing characteristics of 89,500 Western Australian noise-exposed workers found a distinct notch at 6000 Hz for all mean audiograms for both percentage loss of hearing (PLH) and age ranges. This notch was even apparent in the mean audiograms of workers whose PLH was 0 and for workers in the 15-24 year age group. As all workers were noise-exposed it was suggested that this dip at 6000 Hz for the workers with a PLH of 0 may have been indicating that some workers classified as having 'normal' hearing may in fact have had the beginnings of noise damage. Similarly, the 15-24 year olds would apparently have been exposed to very few years of industrial noise. However, the corresponding mean audiogram also clearly exhibited a dip at 6000 Hz once again suggesting the beginnings of noise damage.

Dempsey[21] in his examination of the hearing of hospital workers and Rosler[22] in his review of 11 investigations regarding the progression of hearing deterioration among a wide range of industries suggest that 6000 Hz is an early indicator of noise-induced hearing loss. Similar findings were reported by Axelsson et al[23] in a study of 500 18 year old Swedish male conscripts, Bauer et al[24] in a study of 47,388 noise-exposed workers, and Bruhl et al[25] in a retrospective study of male workers who had suffered noise damage in an automobile sheet-metal processing plant.

However Lutman and Davis[26] suggest that there are shortcomings in the use of ISO 389, regarding its representation of the hearing threshold levels to be expected in otologically normal young adults, creating artificially poorer thresholds at 6000 Hz. They recommend a fresh re-derivation of normal hearing thresholds is required to overcome this problem.

The common usage of 4000 Hz in hearing conservation programmes as an indicator of noise damage may be an artefact of basing this on previous studies which in general did not test at 6000 Hz. Thus the dip at this frequency was not revealed. If 6000 Hz is an early indicator of noise-induced hearing loss then its inclusion in all industrial hearing screening programmes is vital if early detection and remediation of the problem is the goal. If it is in fact an artefact of our testing methods, then further research examining the basis of our assumptions of normal hearing thresholds need to be conducted to correct this discrepancy.

The purpose of the present study is to examine the hypothesis that 6000 Hz is an early indicator of noise-induced hearing loss by comparing the mean audiograms of otologically normal, young, noise-exposed workers with the mean audiograms of otologically normal, young, not noise-exposed students.

Method

Sample A

consists of 1,591 otologically normal, noise-exposed 16-19 year old workers who had a baseline hearing test for the purposes of the Western Australian Workers' Compensation and Rehabilitation Act 1981. All workers were exposed to a representative daily noise dose (8hr day) of 90dB(A) (or its equivalent) or greater noise level, or a peak noise exposure of 140dB(Lin) at any time[27].

Sample B

consists of 190, 13-18 year old students randomly selected from two metropolitan Perth high schools.

16 hours of quiet

To eliminate the possibility of temporary threshold shift all subjects were required to have 16 hours of quiet prior to the test, that is - not being exposed to noise levels above 80 dB(A).

Testers and equipment

All audiometers and booths/environments used were approved by WorkCover WA as meeting or exceeding the standards described in the Approved Procedures.[28] The methods described in the Approved Procedures are based on Australian Standard 2586-1983 for audiometers, and Australian Standard 1269-1989 for booths and testing methods. Where the Australian Standards were found to be lacking more stringent guide-lines were incorporated into the Approved Procedures. All audiometers were required to be calibrated every twelve months.

All testers were registered with WorkCover WA as being competent to perform air conduction audiograms. The students were tested by Post Graduate Audiology students under the supervision of a qualified audiologist.

Testing procedure

Testing was conducted according to the Approved Procedures[28] at frequencies 500, 1000, 1500, 2000, 3000, 4000, 6000 and 8000 Hz in each ear. All subjects were otoscopically examined prior to the test to determine whether any temporary obstruction of the ear canal was present which would prevent a valid result being obtained. Similarly, subjects suffering from colds or other temporary ailments that might affect the test results were not tested until the condition cleared.

Otological screening

All subjects had their hearing test results screened to determine if they met one or more of Waugh and Macrae's medical referral criteria[29]. Subjects who met any of these criteria were suspected of having a hearing loss due to injury or disease and were not included in this study. The students were also asked a comprehensive history and had impedance measures taken of each ear. Students who had any history of significant head injury, use of ototoxic drugs, ear disease or family history of hearing loss or who did not have normal impedance results were excluded from the study.

Noise exposure screening

The students were asked several questions relating to noise exposure. Students were only classified "not noise-exposed" if they had never been near discharging fire arms, never been in a significantly noisy factory or workplace, did not use walkman type radios or listen to loud recreational music, had not been to discos or rock concerts and had not worked on farms.

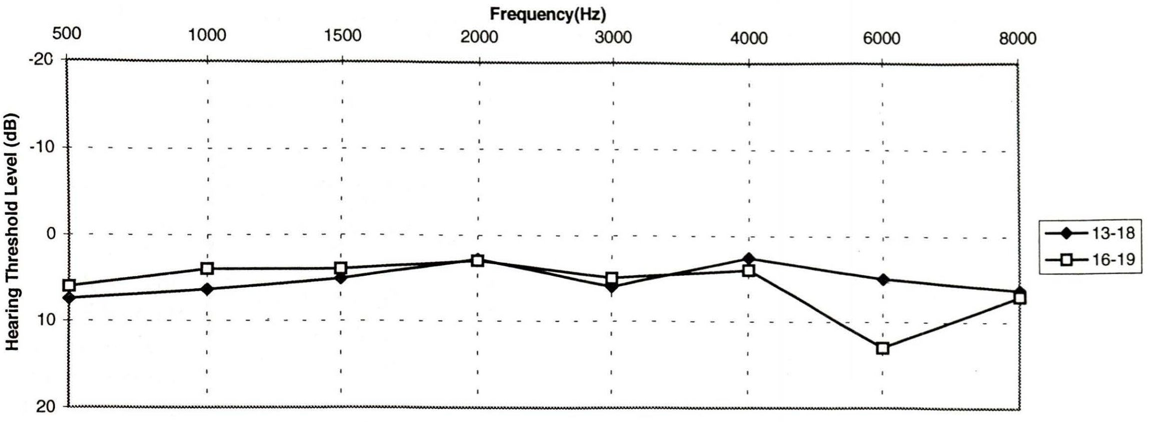

Fig 1a: Right ear audiograms of not-noise-exposed, otologically normal 13-16 year old students (n=68) and noise-exposed, otologically normal 16-19 year old workers (n=1,591).

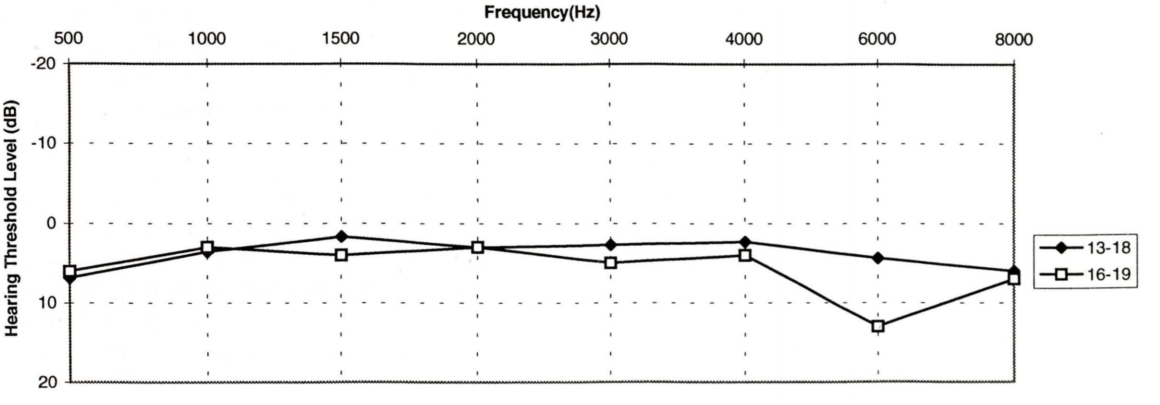

Fig 1b: Left ear mean audiograms of not-noise-exposed, otologically normal 13-16 year old students (n=68) and noise-exposed, otologically normal 16-19 year old workers.

Results

Of the 190 students tested 50.5% (N-96) were found to be otologically normal, a further 27 were excluded because of their history of noise exposure. The remaining 68 students were classified as otologically normal not noise-exposed and their mean audiograms were compared to those of the otologically normal, noise-exposed workers. As can be seen from figures 1a and 1b a distinct 6000 Hz notch was found in the workers audiograms which was not present in the students audiograms.

The notch was also absent in the 27 otologically normal noise-exposed students who were excluded from the study. The mean hearing threshold levels for the 27 noise-exposed students were similar to those of the not noise-exposed students at all frequencies. There was also no 6000 Hz notch observed in the mean audiograms for the students who were classified as otologically abnormal.

An independent group t-test was conducted to determine whether there were any significant differences between the mean hearing threshold levels at all the test frequencies in each ear for the not noise-exposed otologically normal students and the otologically normal noise-exposed workers (Table 1).

The mean hearing threshold levels for the not noise-exposed students were significantly higher at 500 and 1000 Hz in the right ear and significantly lower at 1500, 3000 and 4000 Hz in the left. The mean threshold level at 6000 Hz for the noise-exposed workers was significantly greater than for the not noise-exposed students at the .001 probability level.

No significant differences were obtained between the mean hearing thresholds for the not noise-exposed students and their noise-exposed counterparts at all test frequencies.

Discussion

The purpose of the present study was to examine the hypothesis that 6000 Hz could be used as an early indicator of noise-induced hearing loss. To examine this hypothesis the mean audiograms of two groups of otologically normal young subjects were examined. All subjects were tested in a standardised manner with the only independent variable being exposure to noise. The mean audio-

Table 1. Difference between the Mean Hearing Treshold Levels (dB) of Not-Noise Exposed, Otologically Normal 13-18 Year Old Students (N=68); and Noise Exposed, Otologically Normal 16-19 Year Old Workers (N=1,591)

Right Ear					Left Ear				
(Hz)	13-18	16-19	SD	t	(Hz)	13-18	16-19	SD	t
500	7.43	6.00	5.56	2.11 *	500	6.84	6.00	5.92	1.17
1000	6.40	4.00	7.72	2.56 *	1000	3.60	3.00	4.95	1.00
1500	5.15	4.00	5.53	1.71	1500	1.69	4.00	5.30	-3.59 **
2000	2.79	3.00	5.76	-0.29	2000	3.09	3.00	5.47	0.13
3000	6.03	5.00	5.96	1.42	3000	2.72	5.00	5.36	-3.51 **
4000	2.57	4.00	6.32	-1.86	4000	2.35	4.00	6.72	-2.02 *
6000	5.00	13.00	5.86	-11.26 **	6000	4.41	13.00	5.57	-12.72 **
8000	6.32	7.00	7.05	-0.79	8000	6.03	7.00	8.04	-1.00

* $p < 0.05$
** $p < 0.001$

grams for the not noise-exposed students indicated an absence of a 6000 Hz notch for both ears. Analysis of the difference between the mean threshold at 6000 Hz indicated a significant difference for both ears. Although the difference between the mean thresholds at some of the other test frequencies were also significant, the calculated difference at 6000 Hz for each ear was found to be large compared to the difference at those other frequencies (see Table 1). These results support the hypothesis that 6000 Hz may be an early indicator of noise-induced hearing loss.

The finding of the present study that a distinct notch at 6000 Hz is present only for the noise-exposed workers is consistent with the results reported by other researchers who have reported that 6000 Hz is the frequency which shows the greatest hearing loss for workers exposed to occupational noise. [2, 21,22,23,24,25]. The results of the present study also support the suggestion by Dempsey[21] and Monley et al[2] that the distinct notch present at 6000 Hz on the audiograms of the noise-exposed workers may in fact be a noise notch as opposed to an artefact. Both Dempsey[21] and Monley et al[2] suggest that 6000 Hz may be a good early indicator of noise damage as 6000 Hz may be the frequency which is most susceptible to excessive noise. However, Bauer et al[24] while finding that the hearing threshold of their study population was highest at 6000 Hz, still reported that 4000 Hz may be a more accurate predictor of noise-induced hearing loss if other related health variables were also considered.

The present study does not delineate between recreational and occupational noise exposure for the noise-exposed workers, however it is the position of the authors that recreational noise exposure is typically minimal for most workers compared to occupational noise exposure. This position is supported by Australian data[(30)], and is also consistent with typical noise exposures reported for recreational noise internationally [(31)]. While the present study was extremely strict in eliminating students who may have been exposed to only minimal amounts of noise, data was not available on the actual noise exposure histories of the workers. What is known is that all workers were exposed to a representative daily noise dose (8hr day) of 90dB(A) (or its equivalent) or greater noise level, or a peak noise exposure of 140dB(Lin) at any time, thus all workers were exposed to noise levels which may damage their hearing. While all workers had a 0 PLH and at most 3 years of exposure to occupational noise, the distinct notch at 6000 Hz when compared to the not noise-exposed students suggests that they were all showing the early signs of noise-induced hearing loss.

The otological screening criteria employed for the students were stricter than those employed for the workers. Additional information such as medical histories and tympanometry were obtained for the students, while only the use of Waugh and Macrae's criteria were used to categorise workers as otologically normal. To eliminate this procedural difference as a possible confounding variable the mean audiogram of those students who had failed the otological screening, but were not noise-exposed, was examined and compared to that of the workers. No noise notch was found for the students and the worker's threshold for both ears at 6000 Hz was statistically greater than for the students ($p<0.001$).

The results of the present study support the hypothesis that 6000 Hz may be an early indicator of noise-induced hearing loss. The results do not support Lutman and Davis'[(26)] hypothesis that the notch at 6000 Hz is due testing artefact. It is hoped that the results of this study will stimulate further research to examine the cause of the 6000 Hz notch and the risk factors for hearing loss at that frequency. At the present time there is an abundance of literature on the cause of the 4000 Hz notch but very little on the 6000 Hz notch. This suggests that further investigation into the anatomical/physiological basis for the 6000 Hz notch is required.

The presence of a notch at 6000 Hz will have implications for compensation based testing programmes as well as future hearing conservation programmes throughout industry. The present results strongly suggest the need to include the inter-octave frequency of 6000 Hz in routine hearing conservation audiograms for early detection of the effects of noise damage. Early detection of occupational noise-induced hearing loss should lead to effective future prevention of further hearing loss among noise-exposed workers.

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Extending The High Frequency Response Of Hearing Aids

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Introduction

Boothroyd and Medwetsky (1992) have pointed out that there is still a tendency, in the design of hearing aids, to accept criteria obtained from filtering experiments carried out during the early days of telephone research. These experiments showed that an upper frequency limit of around 3000 Hz was adequate for speech understanding by adults with normal hearing listening under conditions of low noise and reverberation.

Boothroyd and Medwetsky have made out a strong case for extending the upper frequency limit of hearing aids beyond 3000 Hz. They argue that it is inappropriate to extend the results of the telephone research to subjects with sensorineural hearing impairment listening at conversational distances to speech in noisy, reverberant surroundings. They point out that sensorineural damage, low signal levels, noise and reverberation all combine to reduce the amount of information available to the hearing aid user and that it is not reasonable to reduce this information still further by removing potentially useful high frequency cues. They note that the ability to understand speech that is reproduced with an upper limit of 3000 Hz depends, in part, on the listener's ability to take advantage of the linguistic and semantic redundancy of connected discourse. Children who have acquired hearing impairment before achieving language competence are not necessarily able to take advantage of this redundancy and are, therefore, more dependent on full access to acoustic cues.

What, then, should the upper limit of the bandwidth of hearing aids be? Boothroyd and Medwetsky suggest that the upper frequency limit should be high enough for users to hear the sound /s/. They base their suggestion on the observation that /s/ is one of the sounds of English with the highest frequency content and the fact that /s/ carries a very high informational load in English. As a result of their spectral analyses of /s/, they recommend that the upper frequency limit of hearing aids should be in the region of 10 kHz. This paper presents the results of the early stages of a project aimed at evaluating the desirability and practicability of extending the high frequency bandwidth of hearing aids for children. So far, answers have been found to several questions.

(1) What residual hearing do children who use hearing aids have at frequencies above 4 kHz?

The hearing thresholds of 193 ears of children fitted with hearing aids were obtained from children's records at a Hearing Centre. Results for all children with sensorineural hearing losses who had measurable thresholds at 8 kHz were included in the study. The data were sorted on the basis of the type of hearing aid the child was using and the means and standard deviations of the thresholds at the various frequencies were computed. The results are presented in Figure 1. Various types of hearing aid had been provided. They included ::bernafon NAL IT312 programmable in-the-ear (ITE) aids (13 ears), Calaid VLK series behind-the-ear (BTE) aids (48

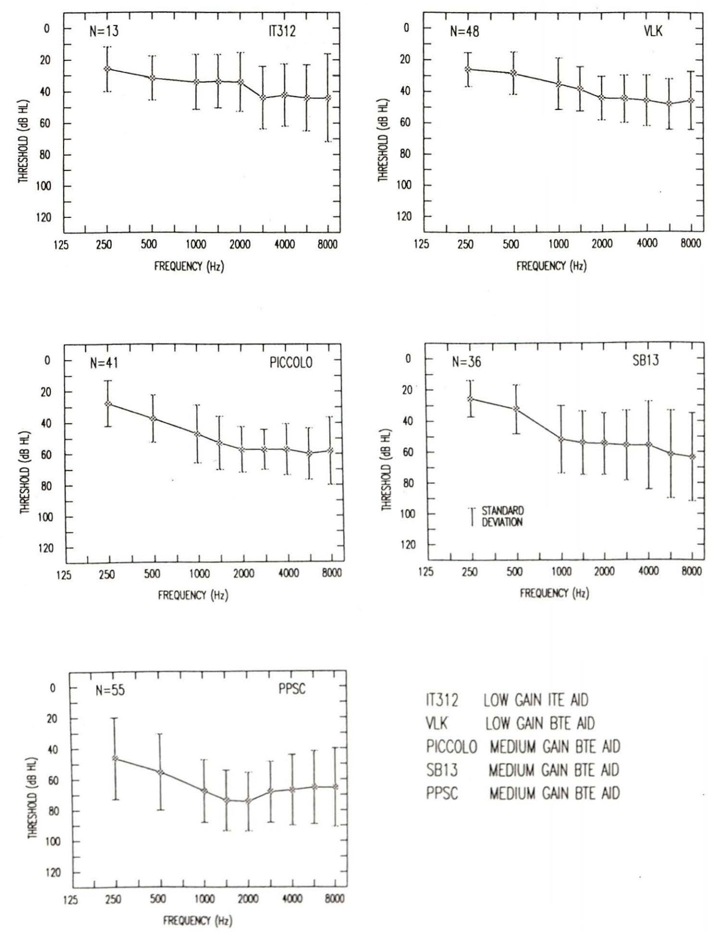

Figure 1: Group means and standard deviations of the thresholds at the various frequencies, determined on the basis of the type of hearing aid used.

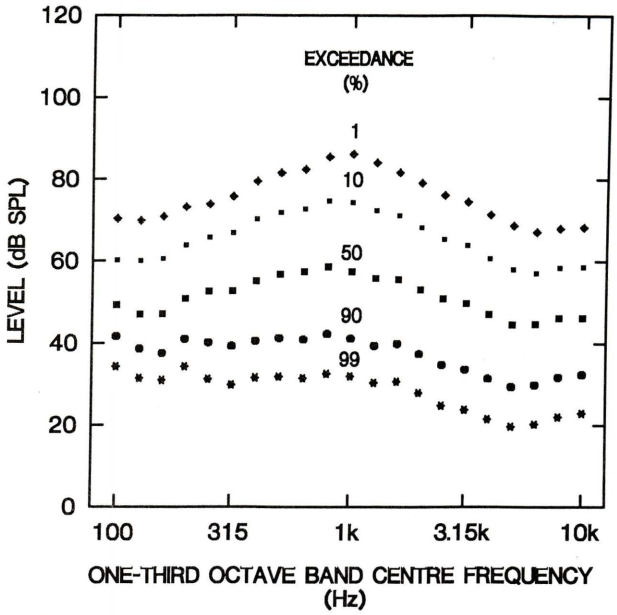

Figure 3: Group mean values of the one-third octave band spectrum of everyday sound. Exceedance levels for the 99th, 90th, 50th, 10th and 1st percentiles are given for each one- third octave band.

ears), Rexton Piccolo PP-O-GC BTE aids (41 ears), ::bernafon NAL SB13 programmable BTE aids (36 ears) and Phonak PPSC series BTE aids (55 ears). These results show that, for all the hearing aid groups, representing hearing losses ranging from mild to severe, the mean thresholds at 6 and 8 kHz are about the same as those at 3 and 4 kHz.

(2) What real ear gains are required at frequencies above 4 kHz by children who use hearing aids?

The National Acoustic Laboratories (NAL) recommended real ear insertion gains at 0.25 to 6 kHz were computed from the children's thresholds using the computer program HASP (Hearing Aid Selection Program). The data were sorted on the basis of the type of hearing aid the child was using and the group means of the recommended gains at the various frequencies were computed. Figure 2 presents the results of the computations. For each type of hearing aid, the group mean recommended real ear insertion gain is represented by the solid line. The recommended real ear insertion gain is about the same at 6 and 8 kHz as it is at 3 and 4 kHz for all types of hearing aid.

(3) What real ear gains are currently being achieved at frequencies above 4 kHz for children who use hearing aids?

The actual real ear insertion gains of the hearing aids were available from the records. They had been measured for all ears by means of real ear gain analysers. For each type of hearing aid, the group mean actual real ear gain is represented by the broken line in Figure 2. Even for the more recent types of hearing aid (the IT312 and the SB13), the upper limit of the bandwidth is about 4 kHz. It is possible that the actual gain tends to exceed the recommmended gain in the 1-2 kHz region for most of the hearing aid types because the audiologists are trying to obtain the best possible gain at 3-4 kHz and must exceed the recommended gain at 1-2 kHz in order to achieve that goal.

(4) What are the spectral levels of everyday sound at frequencies above 4 kHz?

In this investigation (Macrae, 1995), eight children with impaired hearing, ranging in age from 11 to 16 years with a group mean age of 13.0 years, used a Sony DAT Walkman tape recorder with an electret condenser microphone attached to the collar of their shirt or blouse

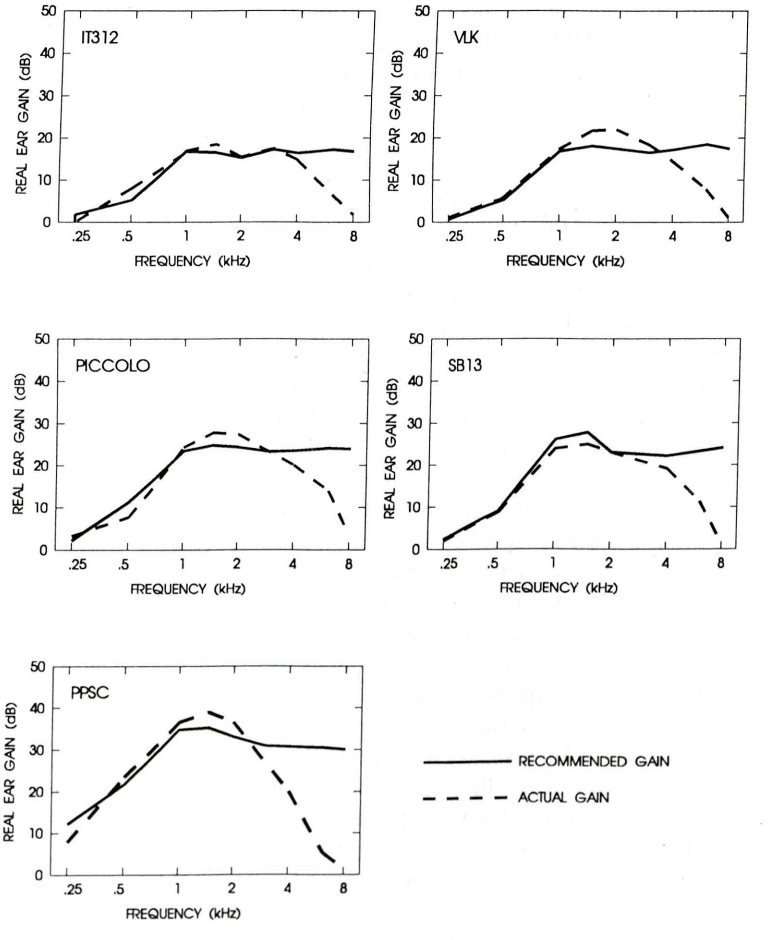

Figure 2: Group means of the recommended and actual real ear insertion gains at the various frequencies, determined on the basis of the type of hearing aid used.

for a period of about 2 hours during the course of an ordinary schoolday. Either morning recess or lunchtime was included in the measurement period for each child because they are relatively noisy times. The tape recordings were analysed by means of a CEL Sound Analyser Type 593. The analysis was carried out in one- third octave bands with centre frequencies ranging from 100 Hz to 10 kHz. For each one-third octave band, exceedance levels were calculated for the 99th, 90th, 50th, 10th and 1st percentiles. Figure 3 gives the group mean values. The exceedance levels at frequencies above 4 kHz are comparable to those at 4 kHz.

(5) What are the prospects of extending the high frequency response of various types of hearing aid using presently available hearing aid receivers?

Extended high frequency response can be achieved easily in in-the-ear (ITE), in-the-canal (ITC) and completely-in-the-canal (CIC) hearing aids and in low gain behind-the-ear (BTE) hearing aids with currently available hearing aid receivers, but is not possible with the receivers currently used in high gain BTE hearing aids. Figure 4 shows that, with an appropriate acoustic pathway from a Knowles ED 1912 receiver to an ear simulator, an upper frequency limit of 10 kHz or even 16 kHz can be obtained with an ITE hearing aid (Macrae & Frazer, 1979). Some current models of ITE, ITC and CIC hearing aids have extended high frequency response. Killion (1981) has shown that, with an appropriate acoustic pathway from a Knowles ED series receiver, a bandwidth of 16 kHz can also be obtained in an ear simulator with a BTE hearing aid.

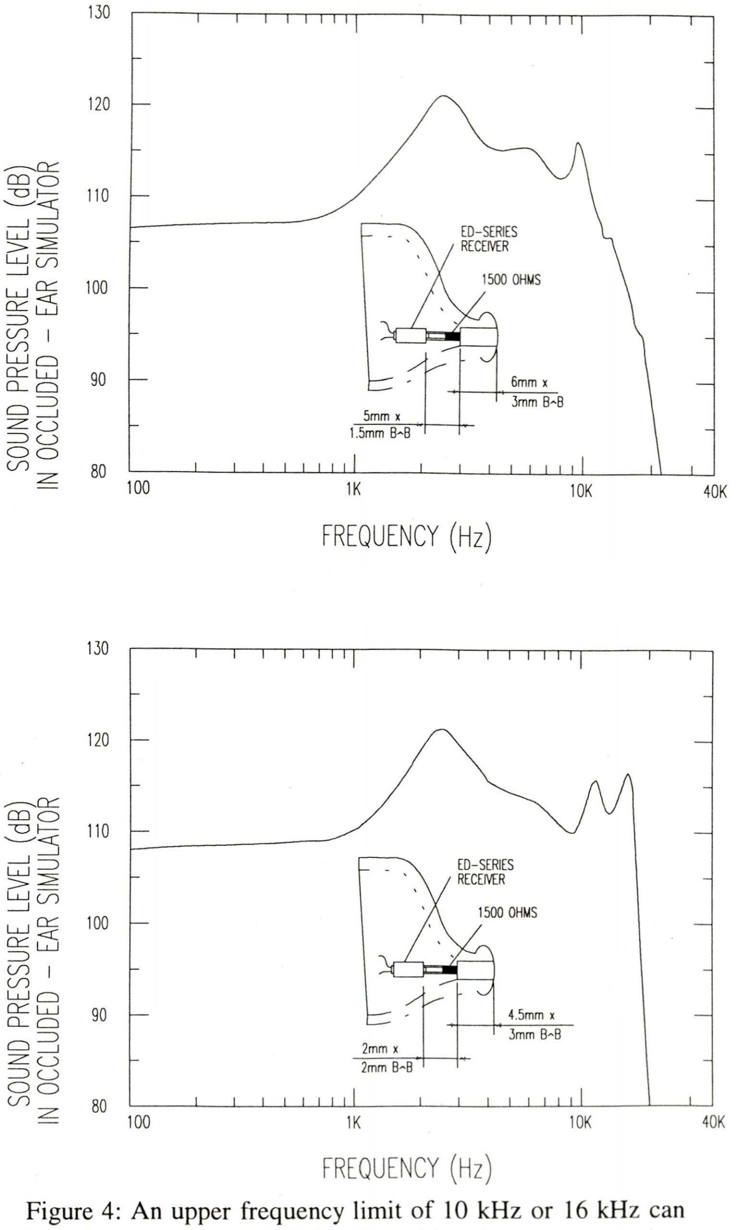

Figure 4: An upper frequency limit of 10 kHz or 16 kHz can be obtained with an ITE hearing aid, using an appropriate acoustic pathway from a Knowles ED 1912 receiver to an ear simulator.

(6) What difficulties are likely to arise with the various types of hearing aid if the high frequency response is extended?

Since extended high frequency response is possible in ITE hearing aids, why is it not available in most models of ITE hearing aids? The main difficulty for extended high frequency response is acoustic feedback. In order to demonstrate this problem, unpublished data on maximum real ear insertion gain possible before acoustic feedback that was collected by H. Dillon and G. Upfold (personal communication) will be used. Figure 5 compares the recommended real ear gain with the maximum gain possible before feedback for an ITE hearing with 1 mm i.d. venting of the earmould, a low gain BTE hearing aid with 2 mm i.d. venting and a high gain BTE with no venting of the mould. The error bars are standard deviations. The closer the two distributions at each frequency are to overlapping, the higher the percentage of users who will have trouble with acoustic feedback. If the two distributions overlap completely, then all users will experience feedback.

Standard deviations were not available for the ITE aid. However, the mean values of the two distributions are the same at 6000 and 8000 Hz, so it is apparent that if the high frequency bandwidth is extended, the problem of acoustic feedback will worsen. This holds true even if no venting is used. However, in the case of the low gain BTE hearing aid, the feedback problem will not worsen when the bandwidth is extended and similarly with the high gain BTE hearing aid. The prospects of obtaining

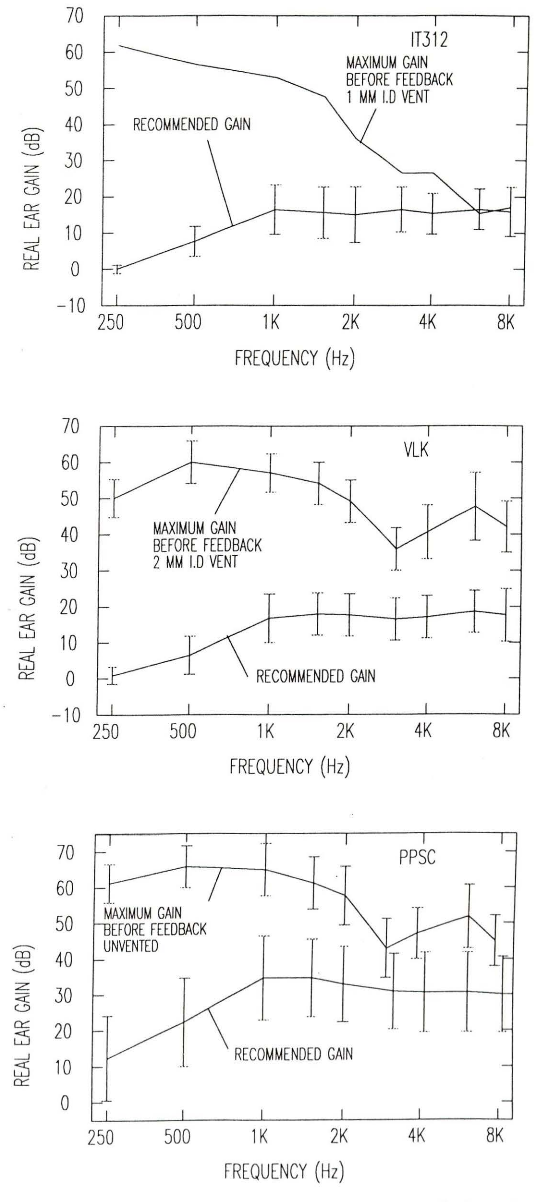

Figure 5: Comparison of the recommended real ear gain with the maximum gain possible before feedback for an ITE hearing with 1 mm i.d venting of the earmould, a low gain BTE hearing aid with 2 mm venting and a high gain BTE with no venting of the mould.

extended bandwidth in high gain hearing aids are slim at present but, as shown earlier, the bandwidth of low gain BTEs can be extended with the use of an appropriate receiver and acoustic pathway to the ear.

Discussion

The information that has been gathered so far indicates that it is practicable to provide extended high frequency bandwidth to children with mild or moderate hearing loss who use BTE hearing aids. Also, if the problem of acoustic feedback can be overcome, extended bandwidth will be provided to a group of children using low gain ITE hearing aids. Venting is, at present, necessary in order to reduce the occlusion effect but venting causes the problem of acoustic feedback to worsen. Among other things, the possibility of minimising the occlusion effect without venting will be explored. Digital feedback suppression reduces the problem of acoustic feedback but appears to be unsuitable, at present, for low gain hearing aids because the probe signal used is too loud for users who do not have much hearing loss.

It would also be interesting to provide a group of older children with mild or moderate hearing losses with CIC hearing aids. CIC hearing aids can reduce the occlusion effect by sealing the earcanal to the end of the cartilaginous section and thus reduce the requirement for venting. They do not have to provide as much gain as ITE aids at the high frequencies because they take advantage of the concha resonance. Extended high frequency bandwidth can be obtained in CIC aids and, because the microphone is just inside the entrance to the earcanal, cues to vertical plane localisation generated by the pinna may be preserved.

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Environmental Noise

Aspects Of Environmental Noise, Sleep And Health

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Abstract

It is widely believed that noise-induced sleep disturbance can affect people's health, not merely in the World Health Organisation sense, of wellbeing, but also in terms of disease. European research has indicated that noise events during sleep can affect heart rate, and it has been suggested that since this does not completely habituate with many years of exposure it may have long-term implications for health. The present paper considers this and other questions in relation to definitions of health and describes some research carried out by NAL in collaboration with the Royal North Shore Hospital, Sydney, on traffic noise and cardiac arrhythmia, and with the University of Newcastle on noise, sleep, and functioning of the immune system.

Introduction

Social survey data indicate that people moderately or seriously affected by aircraft noise frequently believe that the noise affects their health (Hede and Bullen, 1982). Submissions to the Senate Inquiry into Aircraft Noise in Sydney by the medical staff of the Royal Prince Alfred Hospital, Sydney, and the Faculty of Medicine at Sydney University, indicated that they believed that aircraft noise during sleep had deleterious effects on health and the healing process.

It is an unfortunate fact that each of the terms 'noise' 'sleep' and 'health' subsumes a number of meanings. This implies that there may be a number of answers to the question: "Does noise-induced sleep disturbance affect health?", depending on the interpretation or definition of "noise, "sleep" and "health".

Definitions

Noise

As acousticians we are used to a definition of noise as "unwanted sound". This is a relative term and as such presents difficulties by confounding physical measurements with human response. In practice we deal with noise as a by-product ('sound') of energy release produced mainly by machines. Then we go on to look at ways of characterising and quantifying aspects of these sounds, qualitatively in terms of 'intermittent-continuous', 'steady state-impulsive', and quantitatively with measurements of L_{Aeq}, L_{Amax}, SEL, Ldn, ANEF, and so on. A variation of signal to noise ratio may also be devised in which, for example, the L_{Amax} of an individual noise event is related to background noise level (say, L_{A90}) from a number of sources.

Having separated noise measures from human responses, we try to relate them by empirical investigations. We are aware however that the relations between noise and response are frequently modified by human characteristics. These include demographic (age, sex, soci-economic status etc.) and psychological factors (attitude to the noise source, noise sensitivity, perceptions of predictability, and possibilities for coping and control).

Sleep

'Sleep' is also complex. It is a state of diminished arousability within which several distinct states or 'Stages' can be distinguished. These are separated on the basis of physiological measures into Stages 1, 2, 3, 4, and REM sleep. Stages 3 and 4 are called slow wave sleep (SWS) because of the preponderance of very slow waves (<3.5 Hz) in the vertex electroencephalograph (EEG). REM sleep is so called because of the occurrence of rapid eye movements, associated with dreaming. REM sleep also includes diminished muscular tone, and an EEG pattern which resembles waking but may include characteristic 'saw tooth' waves.

Arousals and awakenings are also distinguished. Arousals may be limited to the appearance of alpha frequency waves (8-12 Hz) in the EEG, or may be combined with movement and/or sleep stage change. Awakenings entail the appearance of a full 'awake' EEG (mixed fast frequency, low amplitude waves) or overt behaviour such as pressing a button.

Noise-disturbed sleep may therefore be defined by reduced time overnight in the various sleep stages, increased time to fall asleep (sleep onset latency), or sleep fragmentation (e.g. time awake after first falling asleep, number of arousals/awakenings). It can also be measured by the number of sleep stage changes, arousals and/or awakenings, induced by individual noise events such as truck passbys and aircraft overflights.

Health

There are three commonly accepted definitions of health[1]:

a) The World Health Organisation (WHO) definition ('Wellbeing');

b) Health as 'Successful adaptation';

c) Health as absence of disease, impairment or damage.

The World Health Organisation has consistently maintained that health is not equated with absence of disease or infirmity but *"... is a positive state of physical, mental and social well-being"* (WHO, 1994). The notion of well-being is difficult to pin down, and it may be said that its inclusion in a definition of health is circular. Nevertheless the WHO definition has appeal, and would admit disturbances of sleep architecture, and certain apparently temporary effects of noise-disturbed sleep, such as mood changes, increased reaction time and sleepiness the next day, to the category of health effects.

Environments are not static, and considering health as 'successful adaptation' would also appear to have merit. Under this definition, impaired performance and sleepiness the next day would qualify as health effects. Increased heart rate and reduced finger pulse amplitude responses to noise events during sleep would also be health effects, since they appear not to habituate, even after years of nightly exposure. On the other hand the effects are small, they appear to be 'normal' physiological responses, and the possibility of that habituation itself may entail some physiological cost is not entertained. In fact, of course, researchers who regard these responses as evidence of deleterious effects on health have implicitly adopted the 'disease' definition of health because they assume that the responses are intrinsically unhealthy, and, since they do not habituate, may lead to permanent effects on the cardiovasular system in the long term.

The view of health as absence of disease, impairment or damage is perhaps the definition which has the most popular currency. It does not necessarily imply permanent damage, can refer to functional deficits, and may not be associated with organic dysfunction. Thus reversible states of reactive depression as consequences of chronic environmental noise during sleep could be instances of a health deficit under this, as well as the previous definition of health.

Such effects could be associated with interference with particular stages of sleep. For example Agnew et al. (1967) found that SWS deficit was associated with depression, while loss of REM sleep led to increased irritability the next day. At least 12 studies have suggested that intermittent traffic and aircraft noise during sleep are associated with SWS deficits. Significantly, Öhrström (1991) found that depression was greater in people living in flats adjacent to traffic than in people living in the same apartment block but in flats shielded from traffic noise.

Although the possibility is not often considered, physical damage or impairment due to noise-induced sleep disturbance may also be possible. Horne (1990) suggested that 'core' sleep, which consists predominantly of SWS in the early hours of the night, was related to brain 'restitution' (maintenance and repair of brain tissue). Could brain development or ageing also be affected by core sleep deficit? The author is not aware of any research on this, although recent work suggests that REM sleep may be related to brain development in infants (Mirmiran and Van Someren, 1993).

It has been reported that children up to school grade 6 who live near to airports are smaller and lighter than their counterparts who live away from airports (Schell and Ando, 1991; Chen and Chen,1993). If this is so, factors such as socioeconomic status, maternal smoking etc may be regarded as more likely causes than noise. It is not clear that these and other confounding factors were always taken into account in the studies cited. Nevertheless SWS deprivation in infants due to aircraft noise, reported by Ando and Hattori (1977) could provide a plausible link between aircraft noise and child development, because of the association between secretion of human growth hormone (hGH) and SWS (Sassin et al., 1969).

NAL Research on Noise, Sleep and Health

Three studies of possible health effects of noise-induced sleep disturbance have been carried out at NAL[2]. These were a field study of the relation between traffic noise and cardiac arrhythmia, and a laboratory study of effects of aircraft and truck noise events on arousal, cardiac arrhythmia and urinary catecholamines, both carried out in collaboration with the Cardiology Department of the Royal North Shore Hospital, Sydney; and a laboratory study of the relations between traffic noise, sleep and the immune system in shiftworkers, carried out with the Departments of Psychology and Pathology, University of Newcastle.

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[1] The author is indebted to S. Morell and Associate Professor R. Taylor, both of The Department of Public Health, Sydney University, for this clarification.

[2] Funding assistance for these studies was provided by the New South Wales Roads and Traffic Authority, and the Australian Roads Research Board.

Noise, Sleep and Cardiac Arrhythmia

Cardiac arrhythmia is due to abnormal initiation of depolarisation, or abnormal conduction of electrical impulses, in heart tissue. It qualifies as a health effect because it represents an inefficiency in heart function, but it also has prognostic significance in people with heart disease or who are high on cardiac risk factors. It is not known whether repeatedly inducing arrhythmia can ultimately increase the seriousness of the arrhythmia. The rationale for the studies of noise during sleep and cardiac arrhythmia was as follows.

a) Common cognitive stresses (mental arithmetic, driving in traffic, a stress interview) has been found to increase heart rate and cardiac arrhythmia in people already subject to this disorder. The arrhythmic response appeared to be more marked than the increase in heart rate.

b) Traffic noise during sleep increases heart rate. The response recurs during sleep, even though it had apparently habituated during wakefulness.

c) Several studies indicated that cardiac arrhythmia was associated with certain sleep stages, and sleep stage change.

d) Noise during sleep induces sleep stage change.

The field study, carried out with volunteers living adjacent to Pennant Hills Road, Sydney, indicated that in two subjects truck noises during Stage 4 sleep may have been associated with increased likelihood of arrhythmic events 20 to 40 sec later (Carter et al., 1994a). It was decided to repeat the study with volunteer outpatients of the Cardiology Department at Royal North Shore Hospital, who had presented with frequent cardiac arrhythmia. These subjects slept in a laboratory and were presented with recorded aircraft and truck noise events (65-72 $_{LAmax}$) during sleep (Carter et al., 1994b).

The findings of this study were statistically reliable, but unexpected. It was found that cardiac arrhythmia, in the form of ventricular premature contractions (VPCs), was associated with Stage 4 sleep, but not noise. Indeed there was a distinct, but not statistically significant, trend for fewer VPCs to occur after an arousal response to noise than when the noise failed to induce an arousal response. There was also a trend toward fewer VPCs to occur, the more sleep stage changes occurred immediately after a noise event. Stage 4 sleep is associated with very slow heart rate. VPCs during Stage 4 sleep are likely to be due to some cells depolarising before the normal pacemaker cells. An increase in heart rate due to noise or a noise-induced change to a 'lighter' sleep stage might have forestalled these 'escape' beats.

Do these results mean that noise during sleep has no effect, and may even benefit people with cardiac arrhythmia? Probably not, for several reasons. One is that the subjects in this study all had low grade (less serious) VPCs. Subjects with very frequent and rapid trains of VPCs were not tested. Another is that the noise was familiar, and of low frequency, moderate level and gradual onset. Wellens et al. (1972) found that the noise of an alarm clock (high frequency, high level, sudden onset) regularly induced ventricular fibrillation (very, fast, weak and uncoordinated ventricular beats) in a subject with heart disease. Again, Michalak et al., (1990) found that the noise of a low flying military aircraft caused very large increases in blood pressure in elderly people. These inceases in BP were probably associated with rapid increases in heart rate and therefore cardiac arrhythmia, but these were not studied. Second, Otsuka et al. (1982) showed that in addition to VPCs during SWS, tachyarrhythmias (rapid successive VPCs) were common in REM sleep, suggesting that noise during REM sleep may well increase the frequency, and possibly grade, of tachyarrhythmias, in susceptible people.

Final answers on whether environmental noise can increase the frequency or grade of cardiac arrhythmias will therefore depend on investigations into the effects of noise from e.g. low flying military aircraft and very fast trains, and studies of people with forms of arrhythmia other than isolated (Grade 2) VPCs.

Noise, Sleep and the Immune System

Sleep and the Immune System

Evidence is increasing that sleep, specifically SWS, and the immune system are linked. This evidence comes primarily from research into immune parameters during sleep and insomnia, experimental studies of sleep deprivation, and experimental infection in animals.

Thus the plasma cytokines Interleukin-1 and Interleukin-2, which facilitate immune response, increase during SWS (Moldofsky et al., 1986). Secretion of Human Growth Hormone (hGH), which is associated with generation of cytotoxic T lymphocytes, increases at onset of SWS, and is inhibited by sleep deprivation (Sassin et al., 1969). Non-REM sleep (NREMS), total sleep time, and sleep efficiency are correlated with natural killer (NK) cell activity in humans (Irwin et al., 1992).

Rabbits infected with Staphylococcus aureus infection recover quicker, the more time they spend in SWS during the period of infection (Toth and Krueger, 1988). Brown et al. (1992a) guessed that the effect of sleep deprivation depended on its occurring simultaneously with infection. They carried out three studies and showed that under these conditions sleep deprivation reduced antibody response to sheep red blood cells in rats (Brown et al., 1989a), while in mice it cancelled the effect of prior immunisation against influenza virus (Brown et al., 1989b). However in humans 24-hour sleep deprivation appeared to increase immune response to a typhoid vaccine.

(Brown et al., 1989b). However in humans 24-hour sleep deprivation appeared to increase immune response to a typhoid vaccine.

Recently, Everson (1993) found that prolonged (3-weeks) sleep deprivation led to decline and death in initially healthy rats, apparently without any physiological changes other than massive systemic infection in the last few days of the ordeal. She speculated that this was due to collapse of the immune system, a finding which she believed had immediate clinical implications, particularly for "... the critically and terminally ill who may suffer profound sleep disruption because of pain, grief, discomfort, and therapeutic interventions." (Everson, 1993, R1153).

The above studies totally deprived subjects of sleep. It may be drawing a long bow to suggest that environmental noise could have comparable effects. However, environmental noise affects millions of people around the world. In view of this and the above findings it must be asked whether such noise could interfere with sleep sufficiently to reduce the effectiveness of the immune system. It must also be asked whether, because of greater exposure to noise during sleep, illness, or other reasons, some people are particularly vulnerable to this effect.

Research by NAL and the University of Newcastle

There are reports of at least 12 studies, including studies of people sleeping in their own homes with, and without, sound insulation, which suggest that environmental noise can reduce total time in SWS in young adults, even after some years of nightly exposure. One of these studies, which showed 30% reduction in SWS due to traffic noise, was carried out with shiftworkers sleeping during the day (Ehrenstein and Müller-Limmroth, 1975). When taken together with data linking SWS with immune response, these results suggested that traffic noise during sleep could impair immune response in humans, and a study was carried out at NAL to test this possibility.

In this study 2 groups of nurses on permanent night shift slept in the NAL sleep laboratory for 4 days. Each subject took an oral typhoid vaccine (Typah Vax) before sleeping on Days 1, 3 and 4. Group 1 were exposed to traffic noise during the sleep period, consisting of noise from 8 truck passbys per hour at 64±2dB L_{Amax}, mixed with 45 dB L_{Aeq} continuous traffic noise. Group 2 slept in 30 L_{Aeq} continuous traffic noise. Saliva samples were taken before sleeping on Days 1, 4, 9, 14, and 21 and assayed for IgA and IgM antibodies. It was found that total SWS and total Stage 4 sleep was significantly less in the Noise than the Quiet group ($p<0.05$), and that there was a strong correlation ($r=0.65$; $p<0.05$) between total Stage 4 sleep and IgA antibody response at Day 21 over all subjects.

A somewhat puzzling finding was that although total SWS and total Stage 4 sleep differed between Noise and Quiet groups there was no significant difference between them in Day 21 IgA. A likely explanation is that, as well as noise, there were other determinants of SWS in these subjects which obscured the relation between noise, Stage 4 and the immune response. An alternative explanation, which is not, however in accord with sleep deprivation studies, is that the immune responses *caused* the increases in Stage 4 sleep. Again, the Noise group did show greater IgM response at Day 9. This result is in accord with Brown et al.'s previous findings in humans following 24-hour sleep deprivation, and represents an as yet unexplained perturbation of immune response which may be related to the Day 21 IgA assay.

Clarification of these results requires testing many more subjects under field conditions, such as for example, shift workers sleeping in their own homes under aircraft flight paths at a busy airport. Proposals to carry out such a study have not so far attracted the required funding.

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Prediction And Amelioration Of Environmental Noise From Large-Scale Industry

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Abstract

We present a description of the procedures and techniques used at *Vipac* for undertaking noise assessments for large industrial contracts. Project objectives are typically to characterise and quantify the various components of noise generated and to assess the impact of different operational scenarios on the existing and projected environment. We focus on two major industrial projects: an EIS for a proposed waste landfill at a disused quarry site, and reducing noise (in an adjacent community) from a large industrial plant with over 120 critical sources. The advanced SoundPLAN software package is used for predicting (and validating) the extent and levels of noise generated for different scenarios and conditions. In addition, the software can simulate the effect of various noise control measures and aid in the design and placement of noise barriers.

FUTURE LANDFILL NOISE LEVELS

A noise assessment study was performed for a putrescible waste landfill proposed for an exhausted sandpit in a large quarry. This involved conducting a survey of the current background noise levels at key boundary locations of the proposed site, and predicting the potential noise levels associated with landfill operations.

The site lies completely within an Extractive Industrial (EI) Zone and immediately adjacent to a Residential One (R1) Zone and the Hills Face (HF) Zone. The narrow strip of domestic premises lying along the boundary between the R1 and EI Zones will be classified as "Urban Residential with some Manufacturing Industry"; it needs to be determined whether, as a result of landfill operations, this region experiences noise levels greater than the maximum permissible levels cited in the statutory legislation, *"The Environmental Protection (Industrial Noise) Policy 1994"*.

The proposed landfill site is bordered to the south by current extractive and other quarry operations. To the north, west and east the site is bordered by steep, bare hillsides. The leading edge of two small residential areas starts about 20m back from the north and west boundary of the site. The western boundary is also bordered by a gravel road currently carrying minimal traffic.

The proposed landfill will be developed in a number of cells (50m wide) progressing from north to south; this will be performed in two major stages, where Stage 1 is projected to take 10-12 years. Digital Terrain Models (DTMs) show a total available volume of airspace of almost 2 million cubic metres.

Existing noise environment

The steady background noise level was measured in continuous remote monitoring mode over the period of one contiguous week, at two locations on the perimeter of the landfill site. Periods of manual measurement acquisition were performed at various locations within and around the site for effective event discrimination. This also allowed for the determination of individual noise source levels and spectra.

The measurements were performed using Type 2 Precision Sound Level Meters (Larson Davis Model 700; nominal accuracy to within ± 0.5 dB) with A-weighting and slow response. A sampling time of 15 minutes was used for the determination of the L_{eq} and L_n levels. The equipment was calibrated before and after each set of measurements using a Precision Acoustic Calibrator source.

The highest L_{eq} level measured was 55 dBA (below the criterion level of 58 dBA); typical day (7 a.m. to 10 p.m.) L_{eq} and L_{90} levels were 45 dBA and 38 dBA respectively. The noise level is dominated by wind and traffic noise generated outside the subject site and not by existing operations from within the quarry.

The measured data exhibited no significant tonality, intermittency or impulsive characteristics. Directivity was primarily governed by prevailing wind direction and immediate site topography.

The sound power level and spectral data (in octave bands from 31.5 Hz to 8 kHz) were generated for existing quarry sources (plants, trucks etc.) from LD2900 data obtained on site, providing vital input data for the predictive simulations.

Predicted noise environment

We simulate the potential new sources of noise generated by the proposed landfill operations (along with the major current quarry noise sources). The resulting noise levels are then calculated at six control/receiver points around the site for a range of different operational scenarios, future planning horizons and conditions.

Model description

The model was generated using the SoundPLAN (version 3.72) software package. The SoundPLAN environment allows the propagation of noise from mulitple source types (incorporating reflection, diffraction, and absorption/attenuation processes) to be calculated, interactively analysed, and graphically illustrated and dimensioned.

The topography of the site was entered as continuous contour lines. The residential houses bordering the site were entered as oblong obstacle (diffractive barrier) walls of height 4m. The bordering access road is treated as a dual carriageway supporting projected average hourly levels of traffic noise (from trucks and cars) and implemented as a linear source with truck spectral characteristics.

Current quarry sources incorporated into the model include a Crushing Plant and Batching Plant. Current truck noise is ignored as it is predominantly confined to other sections of the quarry.

New sources (due to proposed landfill operations) added to the model include: a Compactor Plant at a transfer station (about halfway along the western boundary), trucks and cars running and unloading at the transfer station, a bulldozer operating and moving earth/waste in landfill cells, trucks operating and dumping near end of cells, forecast traffic noise along the access road, decelerating/accelerating traffic noise in the vicinity of the transfer station.

Six monitoring receivers (2m height above ground) were established in the model at two background measurement locations, two source control points near the Crushing Plant, and two house monitor points located just in front of the leading edge of the two residential areas. The model was validated against measured existing background levels; accuracy of the model was within ± 1 dBA.

We implemented realistic scenarios with full operational conditions. One truck and one bulldozer will be operating continuously at full power at the centroid of the currently active fill cell. The compactor and trucks will be operating continuously at the transfer station. Throughout all scenarios the Crushing and Batching Plants in the main part of the quarry are operating continuously.

Results

For all scenarios the bulk of the residential areas will experience noise levels less than the 58 dBA criterion. However, for the worst case scenarios (filling at top of two nearest cells) the edge of the residential areas experience levels up to *6 dBA above* the criterion level. Traffic along the access road introduces insignificant noise levels (typically less than 50 dBA).

A model run with a 3m high barrier wall placed along the north and west boundary showed that the levels in the boundary region of the residential areas will remain below 58 dBA.

Suggested ameliorative measures for client

To reduce the noise levels in the boundary region below that which is permissible, it is recommended that a 3 to 4m high earth barrier wall be constructed along the north and northwest perimeter. Passive noise control on machinery sources (eg. retrofitted silencers for the filling bulldozer and truck exhausts) will be necessary to reduce levels. In addition, to a much lesser extent, dense plantings (at least 50m wide and 4m high) will help to further reduce the impact.

MAJOR INDUSTRIAL PLANT NOISE IMPACT

Following complaints regarding residential noise levels due to a major industrial plant, and associated assessments relative to EPA regulations, *Vipac* have developed a practical noise control package.

The initial task was to measure the sound power and spectra of 120 key sources and four extensive plant buildings at the factory. Predictions of residential sound levels were made using SoundPlan including derivation of individual source contributions. A clear noise reduction strategy was subsequently developed.

Existing noise environment

The total sound power of the principal plant noise sources was measured and calculated to be 114 dBA, with the Co-Generation Unit and filter fan exhausts being measured as the strongest sources.

Complete removal of these strongest sources will lead to only a 5 to 7 dBA reduction in plant sound power level. Secondary sources will most likely need treatment, in particular those located closer to the residential area.

Ranking of existing sources in terms of their contributions to residential sound levels showed that the Co-Generation Unit, the Cooling Tower, Boiler House and roof/wall-mounted filter exhausts are the most important sources.

Noise levels at residential locations exceeded regulation criteria by 5 to 14 dBA.

Predicted noise environment

A detailed existing noise model and noise control model were developed for various operating modes of the plant.

To construct the noise model, measured sound power data were imported to SoundPlan and combined with coordinate geometry data obtained from plant layout drawings and council area maps. Soundplan was then used to predict the residential noise levels adjacent to the plant for different scenarios/conditions.

It is clear that many sub-sources contribute to the community sound levels at several residential locations. Model predictions were validated and shown to agree with measured levels to within ± 1.5 dBA.

A *noise reduction of 15 dBA* is required; application of noise control treatments to many sources including the Co-Gen Unit, filters/fans, the Boiler House and the Cooling Tower will be necessary. Within the Co-Gen Unit there are many sources of similar strength so that treatment of one or two sources will achieve only marginal net sound power reduction. Barrier walls will need to be considered.

The desired noise reduction of 15 dBA is not achieved with any of the barriers modelled, primarily due to the reflection of sound from adjacent walls.

Amelioration measures

Even with a practical maximum barrier height of ten metres a noise reduction of only 8 to 10 dBA is calculated. The use of absorptive lining on the barrier was estimated to provide very little benefit (< 1dBA) as the main Co-Gen acoustic sources reflect most strongly off the adjacent Boiler House wall.

In order to achieve a 15 dBA reduction, additional noise control at source is required in addition to the barrier effects. Noise controls were applied in the model to the strongest sources; the resulting noise reduction was about 3 dBA.

A complete noise control strategy was evolved from the calculations to bring residential noise levels within the EPA criterion of 50 dBA. This involves:

A) Construction of a 8 metre high, 50 metre long barrier around the complete Co-Gen and gas turbine unit.

B) Treatment of Co-Gen Unit component sources where significant reductions can be achieved with relatively local treatments. For example, treatments are proposed for silencing the air outlets on walls; cladding of the stack casing; and silencing the Oil Cooler Discharge Stack.

C) Fitment of silencers to the filter exhaust stacks, air filter intake ducts.

D) Enclosure of compressors and several electric motors within Boiler House.

Sound level predictions were repeated with the noise controls listed above; resultant residential sound levels lie below the 50 dBA criterion limit.

These treatments should bring the residential noise contribution associated with operation of the Co-Gen Unit to less than 45 dBA, so that, when operated with the rest of the plant (including the above treated items), the overall noise levels in the residential areas will comply with EPA regulations

Aircraft Noise Impact At Grahamstown Public School

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Abstract

This paper documents the design approach, criteria and ultimate measurement of aircraft noise intrusion upon a school subjected to aircraft noise impact from FA18 fighter aircraft Achieving excellent acoustic conditions responsive to client needs is not simply a matter of installing the best components, it's a matter of design. For new facilities, potential problems should be designed out during the design review and evaluation process as was the case of Grahamstown Public School. One of the main challenges here was to achieve an acoustic environment where the school is habitable without intrusive aircraft noise.This paper also comments on: (a) the requirement to accurately determine the external aircraft maximum noise level, and (b) the importance of low frequency isolation performance, and © the importance of window unit selection in the overall isolation performance.

Introduction

Aircraft noise exposure was recognised during the early planning stages for Grahamstown Public School in 1989 when the site and surrounding area were still undeveloped rural land. The urban sprawl from Raymond Terrace was expanding in an easterly direction towards the planned subdivision of Grahamstown. The final design process for this school occurred during the period 1992-93.

Grahamstown Public School is located approximately 3 kilometres north-east of Raymond Terrace and approximately six (6) kilometres north-west and in the direct flight path of the main runway '12' at Williamtown RAAF Base. Williamtown RAAF is the home base of squadrons flying FA18 fighter aircraft and training aircraft, such as Macchi jet trainers. Other RAAF aircraft also visit Williamtown, including F-111, Hercules, Caribou, Orion, Boeing 707 and various types of helicopters. Civilian aircraft serving the Newcastle area also use the airfield.

ANEF contours

The 1990 Australian Noise Exposure Forecast (ANEF) map for Williamtown RAAF Base indicates the proposed school site to be located between the 20 - 25 ANEF contours.

Australian Standard 2021 in table 2.1 indicates that this location is conditional.

Noise impact

The potential noise impact on this site is considered greater than AS2021 indicates, due to "touch and go" exercises which occur on a regular basis. These events are usually two FA18's or a Boeing 707 touching the runway then taking off to circle the airport and repeating the exercise. The time between events ranges between 18 and 27 minutes. The 1990 ANEF map for Williamtown does not take into account these events.

Ventilation requirements

The normal situation for NSW Department of School Education school buildings is to naturally ventilate un-

Building Site Acceptability Based On ANEF Zones (Table 2.1 AS2021)

Building Type	ANEF Zone of Site		
	Acceptable	Conditional	Unacceptable
School, University	Less than 20 ANEF (Note 1)	20 to 25 ANEF (Note 2)	Greater than 25 ANEF

NOTES:

1. The actual location of the 20 ANEF contour is difficult to define accurately, mainly because of variation in aircraft flight paths. Because of this , the procedure of Clause 2.3.2 may be followed for building sites outside but near to the 20 ANEF contour.
2. Within 20 ANEF to 25 ANEF, some people may find that the land is not compatible with residential or educational uses. Land use authorities may consider that the incorporation of noise control features in the construction of residences or schools is appropriate (see also Figure C1 of Appendix C).

less the local climatic conditions or other site specific circumstances create unacceptable conditions within the teaching areas.

Australian Standard 2021, table 3.3

Building type and activity	Indoor design sound level, dB(A)
Schools, universities	
Libraries, study areas	50
teaching areas, assembly areas	55
Workshops, gymnasia	75

Site aircraft noise exposure

Site measurements and information received from Department of Defence indicated that the maximum aircraft noise would be generated by FA18 aircraft. The maximum noise level of 92dB(A) was also unofficially confirmed. This information matched our previous measurement indication.

Aircraft operations

The type of aircraft operations from Williamtown Air Force Base which impact upon this site range include normal activities and training exercises. Normal activities are both landings and take-off movements. The 1992 ANEF contour map indicates a average daily total of 195 aircraft movements, with 170 being military and the remainder light aviation aircraft. Training exercises consist of 'touch and go' routines.

The "touch and go" operations will generally number greater than 20 movements during the days these operations are flown. With the number of operations and the anticipated aircraft noise levels early acoustic advice was to relocate the school or fully isolate and mechanically ventilate, rather than naturally ventilate which is the Department of School Education preference.

Measured noise levels

The early aircraft noise level measurements recorded in 1989 at the site indicated a maximum level of 92dB(A).

This maximum level was used during the initial design process and included in the project specification.

Design process

There are many different project procurement methods available to provide new school buildings. In this case the Design Develop & Construct (DD&C) method was selected. The DD&C contract documents provided a developed plan and elevations plus specification indicating design requirements is provided. The specification was a combination of specific nominated products and overall performance criteria.

An acoustic design statement was provided in the specification which:

- nominated the maximum aircraft noise level,
- nominated isolation performance of major building elements, eg roof/ceiling , walls, windows,
- nominated reverberation times for teaching areas and multi purpose hall,
- nominated that post construction acoustic performance testing would be undertaken.

Design approach

The availability of aircraft noise data impacting upon the proposed school during the early design stages was limited. Using this information and the Australian Standard 2021, particularly the calculation approach outlined in the Appendix to determine the relevant facade ANA_c (Aircraft Noise Attenuation).

Australian Standard 2021 in table 3.3 indicates Indoor Design Sound Levels For Aircraft Noise Reduction Assessment.

In reviewing the available data it was determined that the following design criteria would be used for this project:

- maximum aircraft design noise level 92dB(A),
- reverberation time within teaching areas within the range 0.3 - 0.5 seconds @ 500Hz,
- determination of the individual facade element STC requirements would be provided to the tenderers. AS2021 calculation would be used to determine STC performance. These calculations indicate the following performance requirements with a 0.5 second reverberation time within the room. Roof/ceiling STC 50; wall STC 46; and window STC 43.
- require successful builder to provide acoustic report to ensure that acoustic design criteria has been incorporated into the construction.

Normal construction methods were used for the design details. In addition no specific acoustic site inspections were conducted during the construction period.

Construction

The early construction proposals indicated brick veneer external walls, multiple barrier layers. The roof ceiling proposal was for a metal roof sheeting with thermal insulation under and an acoustic barrier ceiling below on the horizontal. This ceiling consisted of perforated metal strip ceiling with fibreglass absorption behind and a lay in plasterboard barrier ceiling above.

Table 3.	Design	Measurement
Library	0.4 seconds	0.4 seconds
Teaching Area	0.3 seconds	0.35 seconds

During the final design process there was considerable discussion regarding the final window format. Ultimately a single glazed sliding window with 12mm laminated glass was selected. This was not the original window selected, but because of operational and maintenance requirements was selected over other window units.

The underside of the extended eaves were lined with perforated corrugated metal sheeting. Acoustic absorption was provided between the perforated lining and the metal roof sheeting.

Particular care was taken to ensure that normal construction methods were incorporated and ease of construction was achieved.

The sketch drawing attached indicates the final construction format.

Post construction measurements

The acoustic design approach was to keep the construction details within the normal construction methods. To this end considerable discussion and sketch details were provided to the builder explaining the logic of specific material selections. the success of the approach was the acoustic performance testing achieved the desired results.

Reverberation results

The design target reverberation times within teaching areas was 0.3 - 0.5 seconds @ 500Hz. Acoustic measurements have been conducted with the results shown in table 3 above.

Aircraft intrusion results

Simultaneous aircraft intrusion measurements have been conducted in one Teaching Area only. The results of these measurements indicate maximum external aircraft noise levels of 96dB(A) at the school. This is 4dB(A) higher than stated in the design criteria and also higher than our previous measurements.

The maximum internal noise levels with the windows closed were 59dB(A). During these flying operations the overall attenuation achieved was 37dB(A).

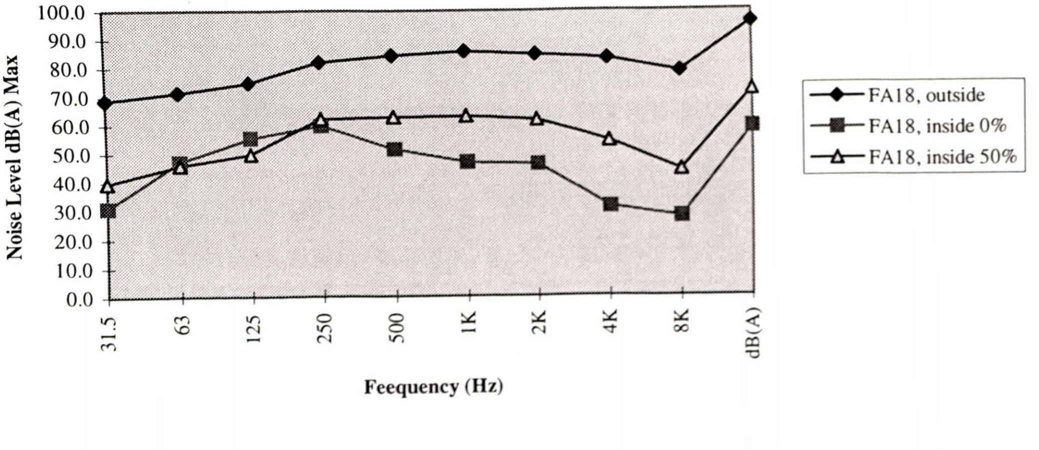
Additional measurements were also conducted with the windows 50% open. During these measurements the maximum external level was 95dB(A) and the internal level 72dB(A). The overall attenuation achieved was 23dB(A). The following graph highlights the performance achieved for both sets of measurements.

Subjective results

The principal and staff have indicated favourable comments about the aircraft noise isolation achieved at this school. Particularly when compared with other traditional educational design approaches in the local area influenced by aircraft noise.

Staff have also commented about the 'quietness' and 'ease of communication' within the teaching areas. This is attributed to the lower than normal reverberation time within the teaching area and the fact that effective com-

Aircraft Noise - Windows 0% & 50% Open



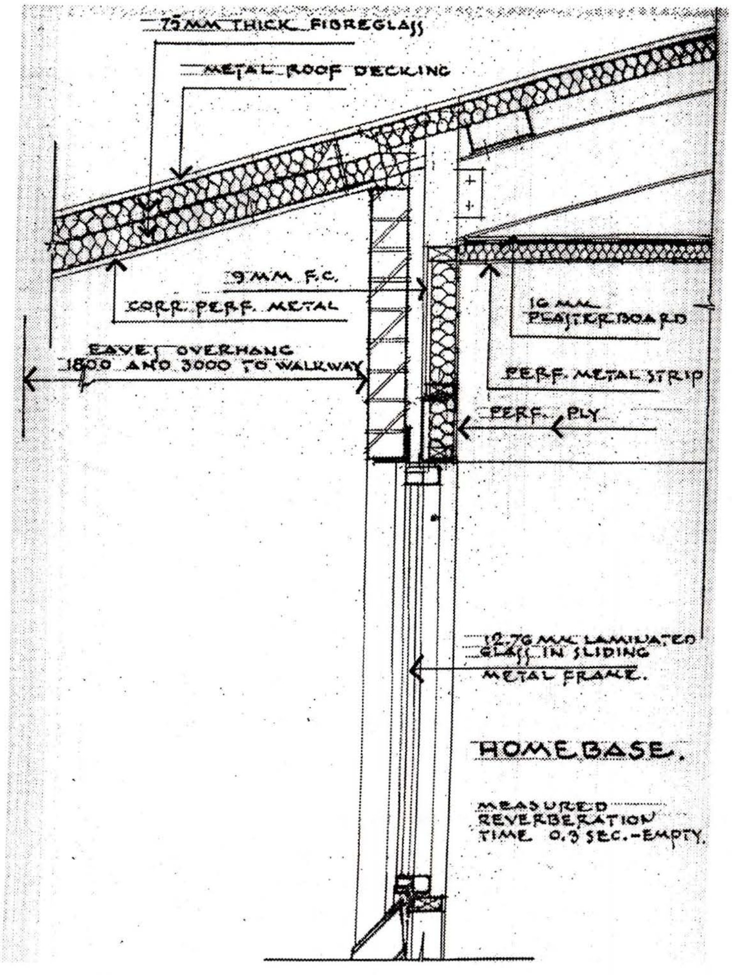
munication is achieved without the necessity to raise the voice level.

Discussion

Two important issues emerge in analysing this project and the performance results. The low frequency isolation performance and the determination of aircraft maximum noise level.

The performance results indicate a loss of performance below 250Hz between windows open and closed. This can be partially explained by the change of window design from double glazed units to single glazed sliding windows. Another contributing factor at this stage was the lack of accurate aircraft noise levels in octave band data.

The second and most important aspect concerns the determination of the maximum aircraft noise level. Measured results indicate internal noise levels of L_{max} 59dB(A) when subjected to an outside noise level of L_{max} 96dB(A). This outside noise level is L_{max} 4dB(A) above the nominated aircraft design level. The inside measured is also L_{max} 4dB(A) above AS2021 nomination for teaching areas.



The importance of correctly assessing the maximum external aircraft noise levels cannot be overstated.

The subjective responses from the teaching staff indicate that achieving results close to AS2021 requirements will provide acoustic conditions which can be used for teaching purposes.

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Acoustics Applied - putting the science and technology to work

1995 Conference of the Australian Acoustical Society
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Session 3B

Audition

Effects Of The Acoustically Degraded Or Competing Speech Signal On Auditory Processing

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ABSTRACT

The speech signal can be modified in a variety of ways. It can be enhanced by several means, but more often, it is degraded in the normal environmental context. We have been using a small number of standardised tests to describe the performance of school aged children when the primary speech signal is challenged by means of filtering, presented in an auditory figure ground of noise, and in competition with linguistic material. These types of tests generally fall within the realm of assessing deficits in auditory processing as an aspect of learning disabilities, and are a useful tool in planning educational strategies for the child experiencing these difficulties.

We have come to rely on the undistorted acoustic signal as a benchmark of normal communication. We spend vast amounts of money to ensure that the music through our home sound system is faithfully reproduced and we can quibble over whether the human ear is even capable of discerning the difference between listening to two systems which have marginally different specifications relating to measurements of distortion or frequency response.

The everyday world does not provide the luxury of high fidelity in normal communicative environments. Although this is most obviously evident in industrial noise environment, it is also obvious that the acoustic environment children's classroom plays a large part in their learning.

There are many forms of distortion or means by which signals (speech or otherwise) are degraded. By way of background these can be simplified into a number of categories. Of course any given category is not mutually elusive. There is no pure acoustic event. Those who are interested in Heisenberg's uncertainty principle know this. None-the-less, very general area of what we deem "hearing" or "speech perception and discrimination" are useful measures of an individuals ability to use an "acoustic" signal to understand, learn or make an appropriate response.

Among other things, a number of signal conditions have been identified as contribution to the processing of speech. The conditions are analogous to auditory functions which can be assessed with a variety of commercially available materials.

Function	Assessment
Binaural Separation	SSW (Katz)
	Competing Words (SCAN)
	Competing Sentences (Willeford)
	Competing Sentences (SCAN-A)
Binaural Fusion	Filtered Speech (Dichotic)
	RASP (Willeford)
Auditory Figure Ground	G-F-W (Test of auditory perception)
	SPIN
	AFG (SCAN)
Resistance to Distortion	Time Compressed Speech
	Filtered Speech (SCAN)

INPUT MODES

Monotic

Diotic

Dichotic

Distortion

EXTERNAL

Acoustic

Temporal

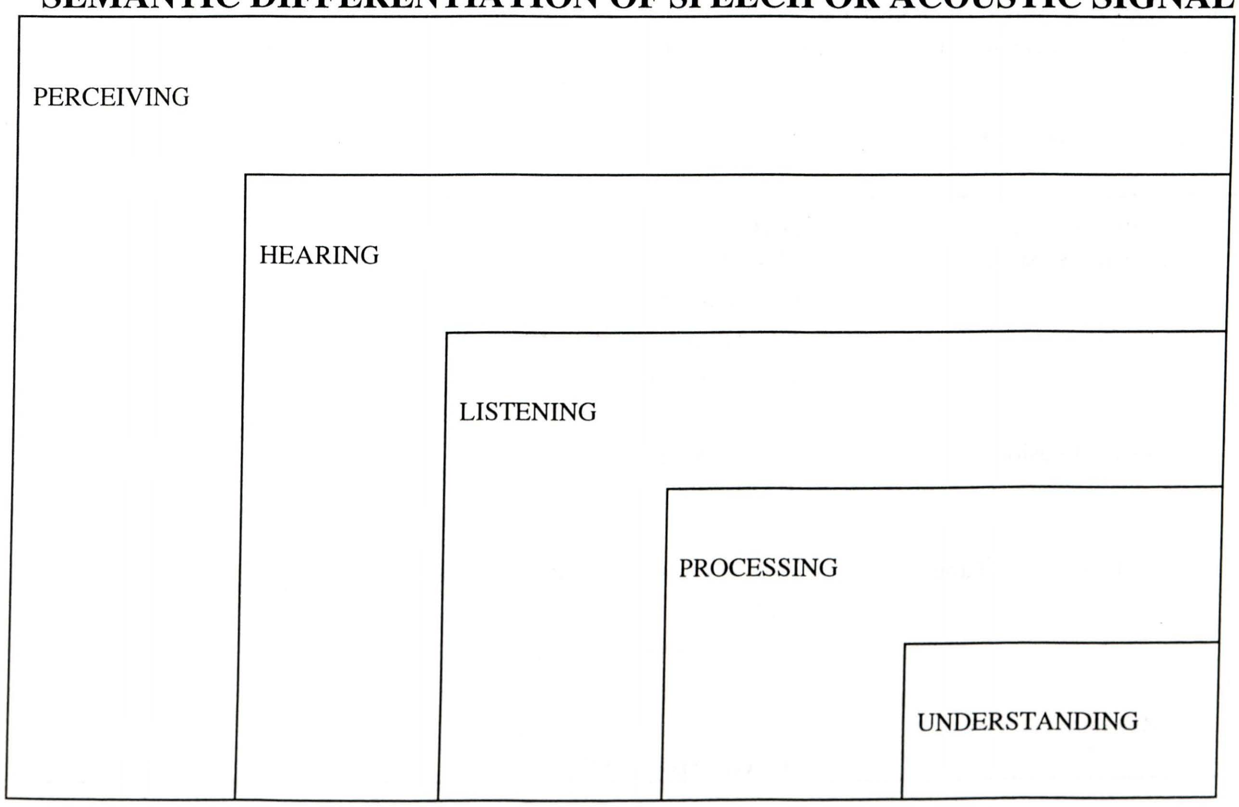
INTERNAL

Cochlear

Brainstem

Cortical

SEMANTIC DIFFERENTIATION OF SPEECH OR ACOUSTIC SIGNAL



Binaural interaction, whether fusion or separation, places significant reliance on the neurological integrity of the two peripheral, brainstem, and cortical pathways for the signals.

Measures of CAP

Behavioural

Physiological

Assuming a "normal hearing" system, what effect does distorted acoustic input have on behavioural aspects of speech perception? The SCAN test was developed to address the following purposes:

1. to determine possible disorders of central nervous (systems) functions by assessing auditory maturation.
2. to identify children who may be at risk for auditory processing or receptive language problems who may require additional audiological and language testing.
3. to identify children who may benefit from specific classroom management strategies to improve auditory and language processing abilities.

Test materials consist of a cassette tape, recording forms and instruction manual. It was designed to be administered on a high quality portable stereo cassette player with two high fidelity headphones or (as is used at Curtin) a high fidelity stereo tape deck, two channel audiometer with matching earphones. The test can be administered in any quiet room in a face-to-face fashion. Presentation is at MCL with bilaterally equal Sensation levels. A test prerequisite is normal and symmetrical hearing levels.

The three subsets of the test are as follows:

Subtest 1, filtered words

Two list of 20 monosyllabic words are low-pass filtered at 1000Hz with filter roll-off of 32 dB/octave. Two words are presented to each ear in alternation, so that each ear has received a list of 20.

Subtest 2, auditory figure ground:

Two lists of 20 undistorted monosyllables are presented in the presence of multitalker babble noise at +8 S/N ratio. Again, two words are presented to each ear in alternation.

Test 3, competing words:

Two lists of 25 monosyllabic word pairs are presented to the right and left ears with simultaneous onset times. The first 25 word pairs are directed right task and the second 25 directed left task.

Scores are norm-referenced, that is, interpretation is based on a comparison of child's performance to age level peers. This can be compared as an overall score or on the basis of subtests. Raw scores are converted to Standard Scores and can then be expressed in terms of percentiles or standard deviations or age equivalents. Confidence levels are provided.

The technical characteristics of reliability and validity measures are included in the manual and are sufficient for this test. Reliability measures consist of Internal Consistency, Test-Retest Reliability and Standard Error of Measurement. Validity assessment included Content, Construction, and Criterion-related validity.

In general for the original study, there was no significant difference between males and females, or matched samples of English and English-as-non-primary language students. This suggested that Australian children would not be adversely affected by the General American accent on the tape. However, as part of a Masters Degree program, the test was normed on a local Western Australian population.

Test results and implications for children in both American and Australian populations are discussed.

Results for producing both norms were based on an approximate stratified sampling procedure. That is, attempts were made to distribute such factors as age, sex, SES, geographics, parents occupation, ethnicity, etc.

Both results showed an age related progression, or developmental attributes, in a positive manner. In terms of the topic today, we can assume that the younger the child, the greater the influence (in a negative sense) acoustic distortion has on the performances of the individual.

In practice, of course, raw scores are converted to standard scores by age, which may then be used to determine where an individual falls within the normal distribution.

The data obtained for Australian children were compared to that of the original sample and significant differences were found in all but 6 of the 32 comparisons. These existence of differences between the samples

Mean Raw Scores and Standard Deviations of One-Year Age Groups Administered SCAN Filtered Words (FW), Auditory Figure Ground (AFG), Competing Words (CW) Subtests, and Composite (Comp).

		Mean Raw Scores				Standard Deviations			
AGE	N	FW	AFG	CW	Comp	FW	AFG	CW	Comp
3	40	20.5	17.2	20.8	58.5	7.6	9.4	18.5	29.7
4	54	25.6	23.2	31.0	79.9	4.5	7.5	22.6	28.2
5	108	27.3	27.1	51.0	105.5	4.7	4.1	18.0	22.9
6	130	30.8	30.9	65.2	126.0	4.4	4.8	14.8	20.0
7	167	31.8	31.0	72.5	135.3	4.3	4.0	14.8	19.7
8	169	33.1	32.5	77.4	143.0	3.9	3.7	11.5	15.2
9	151	33.7	32.6	80.0	146.3	3.4	3.7	7.9	11.3
10	132	34.6	33.6	82.8	150.9	3.4	3.5	10.1	12.5
11	83	35.0	33.5	84.0	152.5	4.1	4.4	9.7	14.9

From Kieth (1986)

would therefore invalidate the use of normative data from the original sample thus supporting the need for local normative data.

There were no significant differences between the samples in isolated instances between the 4 year olds on AFG and CW, 8 year olds on FW, 9 year olds on AFG and CW, and 11 year olds on composite scores. This may have been due to the small number of children in the Australian and the original sample for the four year old age group, hence not enabling a true reflection of the existence of differences between populations. A further possibility may have been due to the general immaturity of this age group. Kieth (1986) cautioned the interpretation of results for this age group because of the inconsistent attending of subjects within his data sample. Similar observations were made in collecting data for the Australian sample.

In general, Australian children scored poorer on the American materials. However, since age group reliability and validity were consistent, there is no reason to discount the use of the SCAN as long as the Australian norms are applied to the test tape.

In summary, this is an example of age related auditory processing skills affected by distortion of the speech signal.

If it follows that learning skills or abilities are related to processing and that processing is related to acoustic environment, perhaps more consideration should be taken into the acoustical design of classrooms and other learning environments.

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The Ling Thing

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Abstract

In the field of education of the deaf and hearing impaired, the "LING THING" has been widely utilised. The Ling Thing has a foundation in specific acoustical properties of sounds and their representation of the entire speech spectrum. This presentation will review the acoustical properties of the the Ling Sound Test and its applications in the areas of audiology and education of children with hearing loss, generally known as the Ling Thing.

This paper is printed at the end of this volume.

Session 4A

Aircraft Noise

Phase-Out Of Older Noisy Jet Aircraft

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Abstract

The Air Navigation (Aircraft Noise) Regulations were promulgated in August 1984. The Regulations applied noise standards to aircraft imported into Australia or constructed in Australia, but did not affect aircraft already on the Australian Aircraft Register. The Regulations have been amended several times since 1984, mainly to specify stricter noise standards. The amendment of December 1991 introduced a phase out procedure for jet aircraft already on the Australian register which do not comply with the latest noise standards. The phase out period commenced on 1 April 1995. This paper considers the effect of this amendment on the Australian Regular Public Transport fleet and on the aircraft noise exposure of residents near major airports.

Regulations

The Air Navigation (Aircraft Noise) Regulations were promulgated in August 1984 (ref. 1). The Regulations applied internationally agreed noise standards, which are detailed in Volume 1 of Annex 16 to the Convention on International Civil Aviation. (ref. 2)

The initial standards, which applied to jet aircraft, are detailed in Chapter 2 of the Annex and specify maximum noise levels measured at three points:

a) **lateral noise measurement point:** the point on a line parallel to and 650 m from the runway centre line, or extended runway centre line, where the noise level is a maximum during take-off;

b) **flyover noise measurement point:** the point on the extended centre line of the runway and at a distance of 6.5 km from the start of roll;

c) **approach noise measurement point;** the point on the ground, on the extended centre line of the runway, 120 m (395 feet) vertically below the 3 degree descent path originating from a point 300 m beyond the threshold. On level ground this corresponds to a position 2000 m from the threshold.

These standards applied to subsonic jet aircraft for which the application for certificate of airworthiness for the prototype was made before 6 October 1977. Jet aircraft noise certificated to Chapter 2 standards include the Boeing 727, the McDonnell Douglas DC9 and the Fokker F28.

Stricter noise standards, detailed in Chapter 3 of the Annex, were applied to subsonic jet aircraft with an application for Certificate of Airworthiness for the prototype on or after 6 October 1977. Aircraft certificated to Chapter 3 include the modern high bypass engine aircraft such as the Boeing 737-300s & 767s, the Airbus A300 & A320 and the British Aerospace 146s.

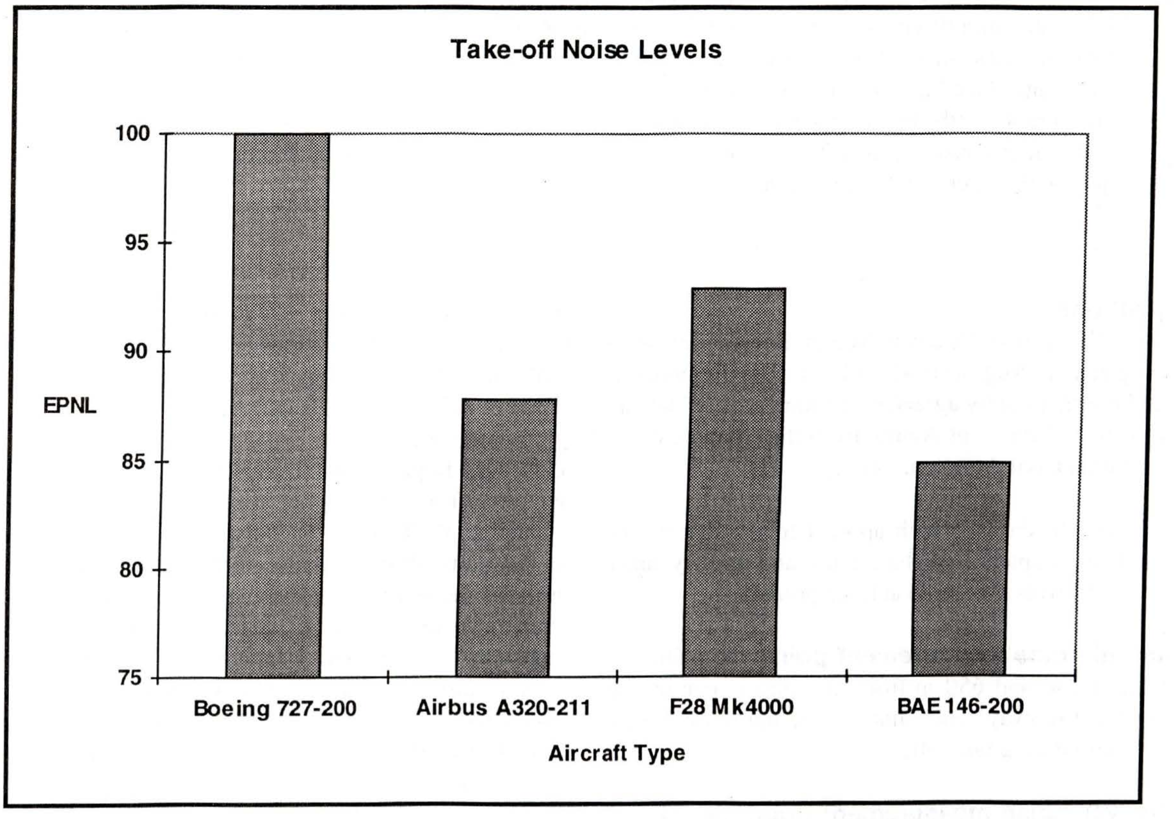
Under the Regulations introduced in 1984 subsonic civil jet aircraft which did not meet the appropriate noise standards of Chapter 2 or 3 were not allowed to operate in Australia. Prior to the Regulations a policy existed to prevent the operation of non noise certificated jet aircraft in Australia, after 1 January 1995 for domestically registered aircraft, and 1 January 1988 for foreign registered aircraft. The Regulations had various minor amendments, but a significant amendment was introduced in 1991. (Ref. 3) The new Regulation 10B introduced a phase out procedure for subsonic jet aircraft that did not meet the stricter noise standards of Chapter 3. The permission for these aircraft to operate in Australia was terminated :

- 25 years after the certificate of airworthiness was issued; or
- 31 March 2002;whichever occurs first. This applies to domestic and foreign registered aircraft.

Noise impact

In March 1995 there were two main types of Chapter 2 noise certificated aircraft in use in Australia; the Boeing 727 and the Fokker F28. The Australian Aircraft Register had seven B727s and 16 FK28s listed on 31 March 1994 and a year later the number of B727s was unchanged, but there was one fewer FK28. The number of Chapter 3 jet aircraft used on domestic transport routes included 4 Airbus A300s, 12 Airbus A320s, 57 Boeing 737s, 6 Boeing 767s, and 25 BAE146s.

Aircraft Type	Passenger Numbers	Take-off EPNL	Sideline EPNL	Approach EPNL	Noise Certificate
Boeing 727-200	146	100.0	102.2	103.2	Chapter 2
Airbus A320-211	144	87.8	94.3	96.4	Chapter 3
Fokker F28 Mk4000	65	92.9	101.7	101.4	Chapter 2
BAE 146-200	73	84.9	87.3	95.6	Chapter 3



The difference in noise emitted by a Chapter 3 aircraft with similar passenger capacity to a Chapter 2 jet aircraft are illustrated by the attached graph. (Ref. 4 & 5)

The modern A320 is 12 dB less noisy on take-off than the older B727, while the BAE146 is 8 dB less noisy than the FK28. The difference in arrival noise is close to 6 dB in both cases.

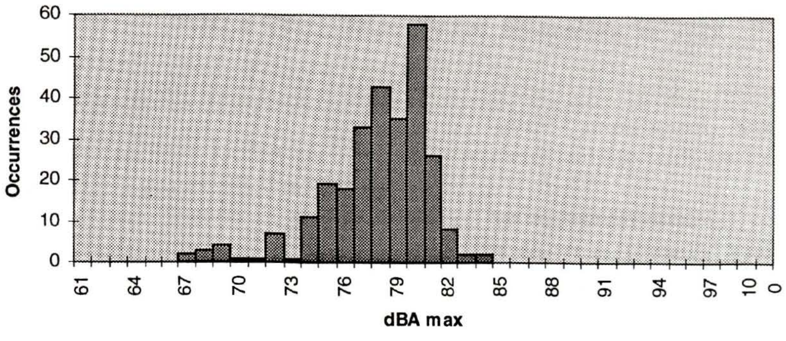
If we look at the number of operations by Chapter 2 noise certificated jets at Perth Airport, an even better picture emerges. In July 1994 there were 35 operations by B727 aircraft and 536 by FK28s. These figures had fallen by July 1995 to no operations by B727s and 380 operations by FK28s. While operations by FK28s had fallen by 42%, operations by BAE 146 aircraft had increased significantly.

Perth Noise & Flight Path Monitoring

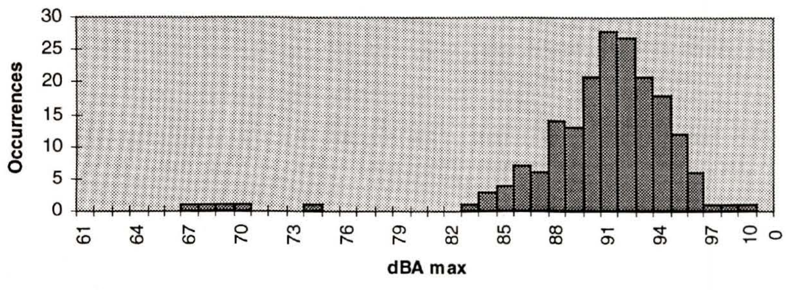
Airservices Australia predecessor, the CAA, installed a Noise & Flight Path Monitoring System (NFPMS) in Perth in 1994. Outputs from this system can be used to determine if the noise from different aircraft types, as measured by the system, displays the same reduced noise from the Chapter 3 aircraft, compared with Chapter 2 aircraft, as predicted by the noise certification figures.

For departures on runway 21, there is a Noise Monitoring Terminal at Cannington which can be used for this purpose. For B727 departures, the average LAmax was 90 dBA; while for A320s the corresponding figure was 80 dBA. The difference is slightly less than the noise certification figures, but still shows very significantly reduced noise from the more modern aircraft. At Perth there are no Noise Monitoring Terminals on the extended centre line of the runway and close to the airport. It was not, therefore, possible to obtain reliable figures

BAE146 Take-off Noise, CAIRNS NMT 3, July-Sept 95



FK28 Take-off Noise, CAIRNS NMT 3, July-Sept 95



for the noise level of BAE 146 aircraft on take-off from the Perth Noise and Flight Path Monitoring system.

I, therefore, used noise levels from the Cairns Noise & Flight Path Monitoring System and in particular from the Noise Monitoring Terminat the hockey field some 1.2 kms from the main runway. This corresponded to 4.7 km from the start of roll for departing aircraft on runway 15. The figures for departures were FK28 90.5 dBA and BAE146 79.2 dBA. The difference of 11 dB is slightly higher than the difference in the noise certification levels.

The above charts show the frequency of the maximum noise level at the NMT north of Cairns Airport. Factors affecting noise level include weight of aircraft (load), head or tail wind and air temperature. For this site, another factor is the take-off procedure which results in a left turn soon after departure in order to avoid flying over the city of Cairns. The aircraft are therefore not all directly over the NMT.

Conclusions

The great majority of Australia's domestic jet fleet are modern aircraft which meet the Chapter 3 standards. For Perth residents, there has been a marked reduction in the numbers of Australian registered Chapter 2 noise certificated jet aircraft using the local airport. The modern aircraft used by the Australian airlines are significantly quieter than the older jet aircraft they replaced.

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A New Monitoring System For General Aviation Fields

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Overview.

Since the first 1990's technology airport monitor, the Cirrus Research CRL 243/1, was installed at Mascot and Brisbane about 5 years ago, interest in airport noise monitoring has spread. In the early 1990's, following the Sydney installation, many large international airports installed new noise monitoring systems and found that they had a useful tool to help control the impact of their operations on the exposed population. Many of the new systems, like Sydney, had connections to the airport secondary radar so that the aircraft tracks could be monitored and stored. In this way a correlation between flight track deviations and excess noise at a particular source could be made.

It was not long before citizens groups, local authorities and even some operators were asking why smaller 'feeder' fields could not have systems installed. The obvious reason was the cost of the very complex and hugely expensive UNIX based systems such as were typically chosen by big airports, much of this cost being partly due to features more like computer games than serious noise measurement. To reduce this cost Cirrus Research plc introduced the first MS-WINDOWS program called RASP in 1991. RASP was a simple system, having every possible acoustic index and measurement but having none of the 'bells and whistles' associated with the majority of UNIX offerings.

A Windows system has always been typically one third the cost of its UNIX equivalent and now with the introduction of Windows 95, is has adequate functionality making it just as fast and capable as UNIX in this application. A system similar to that at Sydney can easily cost over $1million, while an MS-WINDOWS systems such as at Dresden in Germany, would be about one third of this. However, a reduction in price of three to one was not enough for the GA fields and while $330k may seem small to large publicly owned airports, it is totally out of the reach of a small General Aviation operation. The task then given to our designers was to produce a system with measuring accuracy as good as a big system, but at a budget price; the target being a system costing less than $100,000, or £50,000, so obviously many compromises had to be made.

In a technical paper parameters such as "cost" should not normally figure as a main parameter, but in the case of these General Aviation airfields, this is the only constraint which stops them installing noise monitoring. This then automatically becomes the most important parameter and the yardstick by which to judge success.

The compromises

The design team started with the specification of the Cirrus Airport monitor CRL 243/1 and tried to decide what features were 'commercial' and which were essential for operation. Clearly, it is a hard commercial world and many features on the large monitor were there either mainly or solely because of competitive pressures. For example, no reasonable acoustician would think that in 1995 'D' frequency weighting, third octave analysis, 'I' time weighting and Peak response are needed on a dedicated airport monitor. However, one or more of our commercial competitors fitted these and to make sure that Cirrus were never excluded from a tender, these were all fitted to the CRL 243/1. Their removal would certainly lead to reduced cost and lower power consumption and improve reliability. The current working group document ISO TC43 -WG43 recommends that all measurements should be in dBA and thus confirms this view, so all these functions were removed from the specification for GA fields.

The next significant cost was the security built into a Cirrus Noise Monitor. There are 20 different alarms, auto-dial to specified telephone numbers after an alarm, as well as dual sealing to make sure that water would never get into a Cirrus instrument. The case and internal covers of the CRL 243/1 are stronger than any competing unit and are, as far as can be, vandal proof. However, at a GA field the operators are very happy to provide their own external security if this limits the initial cost. Finally some 'nice touches' were deemed not needed. For example, the mast is a beautiful tilting assembly and can be dropped by a single person easily. For the new system, a fixed mast would be used, either mounted near ground level - with the appropriate acoustic corrections - or mounted on a suitable existing structure. As well the very complex and totally waterproof, microphone back venting was removed in favour of a simple system.

Finally, the method of connection to the public supply could be simplified. Currently, a separate 'termination' unit is needed to meet the requirements of various national power and telephone authorities. This allows the installing engineer to mount the system and then the public utilities staff can come along and simply connect the external services without interfering with the system. To avoid all this, the new unit was to be simply plugged in like a domestic appliance, using a residual circuit breaker plug. This obviously requires waterproof connectors, but these are now readily available at low cost.

The next parameters to be investigated were the communications and the software. Typically, Airport Noise Monitors can communicate with their host computer by radio links, dedicated telephone lines, either tie line or 'dial-up' as well as Infra red links, cell phones as well as direct RS232 connection etc. All these options added more code to the internal program, meaning larger memory sizes as well as added cost in supporting very large embedded codes. In addition, each of these, except direct connection, requires a MODEM which in itself not expensive, but which doubles the size of any batteries needed and also adds to the complexities of the system. For a GA field, most users are happy to visit the monitor once a week and collect the data by means of a laptop computer. An option offered by many potential users was to site the new monitor inside a building and then use a direct permanent connection to the host computer via line drivers. This is a far less expensive option and while a MODEM can be specified as an option, it would operate only in 'dial-up' mode with a fixed protocol, all other methods being ignored.

The host computer software did not need the features of a larger system and significant costs could be saved here. Both the UNIX and MS-WINDOWS systems are built round a map of the airport showing the current noise levels as well as the flight tracks. For a GA field this is too elaborate and a simple command line MS-DOS program is more suitable and far less expensive, both to write and maintain. Using an MS-DOS based program also meant that the computer could be a much reduced device and even a simple, low cost, old fashioned 80286 will operate fast enough.

However, there was one feature that was not going to be compromised - the measuring accuracy-. The Cirrus NMT has been tested by several 'Pattern Approval' laboratories and has the unique record of never having been failed on accuracy; this is one feature that had to be kept. To this end the same processor techniques were used where the short Leq is generated by the Wallis Holding method where only after Leq computation is the signal digitised.

Design brief

Given the compromises on specification, the first task was to find a suitable case, which would be adequate in terms of weatherproofing and at the same time be affordable. A similar problem had been solved by the announcement at the 1993 Glenelg meeting of the new CR:245, a dedicated environmental monitor and the same format would be almost ideal for the GA field monitor. Indeed, many of the internal blocks were logically very similar so in fact the basic CR: 245 was modified to be a dual purpose instrument, capable of being configured in production to either role.

The basis of the data acquisition is a 120dB dynamic span circuit which takes a new Leq value every 62,5mS. These short Leq values are used in the same way as the CRL 243/1 to produce the type of data that would be needed by a GA field. For example, it is not sensible to have the instrument fully configurable as is the larger unit. Most GA operations do not have the luxury of a permanent acoustician on site and thus any attempt to re-configure the unit could result in either a logical error or a parameter being chosen that make no real acoustic sense. On a conventional Cirrus airport monitor, a configuration error is not very important as the unit can be re-configured remotely via the phone lines. However, this new unit was not intended to have a modem and thus any re-configuration would mean a site visit. Such a visit would not be part of a maintenance contract and would therefore have to be paid by the client.

Because re-configuration on site was not a user arrangement, the number of 'standard' measurements had to be carefully considered. For example, the normal Cirrus Noise monitor has 6 environmental noise groups operating at different times or periods. All these periods and times can be set by the user and the 'recognised' aircraft can be included or not. In other words, the environmental groups can measure the actual noise or the noise with the aircraft events removed; thus giving the direct effect on the noise climate over any time period the operator desires. In the new unit, these would be fixed at installation to fit the country regulations and locked.

The data taken by any noise monitor is then passed to some means of accessing and reporting the results and for the RASP system, a Paradox database is used. The early RASP systems had an MS-DOS version of Paradox as MS-Windows was not then available and a few airports such as Wellington, New Zealand still use this DOS database system. It was therefore a simple task to make the new MS-DOS program file data under the older conventions, which could still be read by the newer Windows version. Cirrus Research did have a program available to operate a single NMT, called 243FATS, originally written to exercise an NMT rather than to operate it. This program could performs most of the basic functions of down-loading data etc., but was unusable to all except very computer literate users. It required the user to enter the memory locations in hexadecimal code and this limited it to a few engineer users. However, the basic code could be utilised in a new program much more user friendly and this was called 3D (Data Download and Display). 3D could transfer data from an NMT automatically or manually and file the data into the standard report format from which it could be printed. Obviously at this stage, deviations form standard can be accommodated, but the additional cost of modifying software is often beyond the means of small airports, so it was very important to make the reports as universal as possible.

The final major task was to re-think the microphone system and installation problems. The CR: 245 has a beautiful locking microphone mount which fits into a travel case. For this system, an older microphone system the MK425 was to be utilised as the basis for the new unit. MK425 units have been installed in the north of Scotland for over seven years and not one has ever failed due to water ingress, so this unit is clearly suitable for a GA field. The new microphone is either mounted on a conventional tripod or it could be strapped to an existing pole or even wall mounted using standard television type mounting brackets, available world wide at very low cost. Recognising that factory engineers cannot travel all over the world to install these low cost units, they are intended to be 'self installed' using no more expertise than is available at an electrical workshop. However, what the manufacturer believes is "easy to install" can seem very difficult when the language, training and background of the user is quite unlike that of the designers. For this reason, exceptionably clear manuals for each stage of the operation were needed and time alone will tell if these are adequate outside the English speaking countries.

The final system

Most GA fields have a single runway and with intelligent placing two monitors can be used, one at each end of the runway. These monitors have seven day battery operation in case of failure of the 220v, so the user does not have to check on them every day. The microphone is mounted within 15 meters of the monitor to avoid having to use high current cable drivers or loosing high frequency, high level signals. Connection between the microphone and the NMT are via plugs and sockets, so that a unit can easily be removed. This is a new departure for an outdoor monitor as Cirrus have always made their connections 'hard wired'. This has always caused arguments with service personnel who did not like to make hard wired joints in the field. However, the much improved reliability this gave was worth the argument.

The main unit, looking very like a CR: 245 without its microphone system is simply plugged into the microphone and a suitable source of 220 volts and the installation is complete.

In operation, the typical GA user has little to do. He has to visit each of the NMT once a week and transfer the data to a laptop. The diskette from the laptop is then put in the host computer and the data taken in by 3D. If the user is lucky enough to have one NMT connected by a cable, only 1the data from the other NMT need be transferred. After transfer, the program automatically prints out the desired standard reports, typically including a daily, weekly and monthly report on both recognised aircraft events as well as the overall environmental noise climate. Usually, hourly L10, L50 L90 and Leq is chosen, but any parameters from the full airport reports can be included if so ordered.

For routine maintenance, the user, or a local service engineer, should visit twice a year and make sure that all the weather seals are intact and ideally once every two years, the system should be sent back to an approved laboratory to be re-calibrated. However, with the number of standard sound level meters in service which have never been recalibrated, this advice is probably going to be ignored.

Summary

The main objection expressed by GA fields against having a noise monitor have been demolished. A system fully meeting IEC 651Type 1 and the 1995 draft ISO and IEC standards has been produced at a cost under the initial target. A small field can now install a full system for less than $75,000, including full service.

No compromises whatever in the measuring accuracy have been made, although some of the extras loved by salesmen have not been fitted.

The prototypes for the system were installed at sites other than airports and have so far proved to live up to the expectations. The first airports to have the new instrumentation are being installed in October 1995.

Insulating Houses Against Aircraft Noise: A Pilot Study

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Introduction

In November 1994, the federal Minister for Transport announced a voluntary sound insulation package for existing residences located within the ANEF 30–40 contours (year 2010 contours) around the Sydney Kingsford Smith airport. It is estimated that more than 4000 houses are located in this zone. The acoustic treatments to be adopted for the houses were to take account of the results of a pilot study on a relatively small number of houses.

The responsibility for the pilot study was given to the Department of Administrative Services (DAS): the CSIRO in conjunction with DAS carried out the pilot study in the first six months of 1995. A total of eighteen (18) residences were chosen and studied in the pilot project, and this paper presents and discusses some of the important acoustic results from this work.

Selection of houses

The 18 houses were divided into two groups. The first group of eight houses chosen for the study were uninhabited, while the remaining 10 houses were occupied. The unoccupied houses permitted maximum freedom and flexibility to undertake various insulation upgrading measures and study the resulting improvements. These houses were studied using the step-by-step approach, i.e. the acoustic improvement was measured after each step was completed.

The work on the occupied houses began later so that the knowledge gained by treating the unoccupied houses could be applied to these houses, which were noise monitored on the basis of 'before' and 'after' acoustic treatment only. The occupied houses represent the normal situation that is likely to occur in the large-scale aircraft noise insulation project and it provided useful feedback of residents' reaction to the acceptability of the acoustic treatment. The costs of the various acoustic insulation improvement measures were also recorded to provide projections for the large-scale aircraft noise insulation work planned over the next three years, but the financial aspects are not covered in this paper.

Aircraft noise

Aircraft noise is different from many other noises in that the noise source is high in the air. Problems with aircraft noise arise largely from the incompatibility of having airports and residential buildings in close proximity. AS 2021[1] provides guidelines on the appropriateness of land for various uses based on the ANEF zoning, and for new residential construction, the Standard classifies the land as unacceptable if the ANEF is greater than 25. In the case of Sydney airport, many existing residences fall in the unacceptable ANEF zone. The focus of the pilot study was to study the sound insulation measures that may be applied to the existing residences to make their indoor noise environment satisfactory against the intrusion of aircraft noise.

Because aircraft noise levels at a particular site can vary due to factors such as loading, operating conditions and flight path variations, it is appropriate to use the average maximum aircraft noise level data tabulated in AS 2021 for different aircraft types as external design sound levels. These values can be calculated if the landing and take-off distance coordinates and site elevation information is known. As the Sydney airport runways are practically at sea levels, almost all sites will be higher than the runway levels. On the advice of the Department of Transport, the Boeing 747-200 series of aircraft was chosen as the relevant aircraft type because other noisy aircraft such as the Boeing 727 and DC9 aircraft are soon to be phased out.

The recommended indoor design levels in AS 2021 are 50 dB(A) for sleeping and relaxing areas and 60 dB(A) for normal domestic areas. The difference between the calculated external design sound level and the recommended indoor sound level gives the Aircraft Noise Reduction (ANR) that is required for a particular dwelling. The difference was calculated for both take-offs and landings and the higher difference between the two was taken as the design goal.

The residences were chosen using the criteria that they should be representative of the housing stock and be geographically distributed fairly evenly across the suburbs in the ANEF 30–40 contours. A large proportion of houses in this area are of late Victorian and Federation style and are between 80 and 100 years old. Most are of brick construction with a terracotta tile, slate or corrugated steel roof and have timber-framed double-hung or casement-type windows.

Measurement methodology

The ANR provided by all the residences prior to any acoustic treatment was measured to establish base reference data. For those houses that were studied using the step-by-step approach, the noise measurements were carried out after each step was completed. Maximum A-weighted sound pressure levels of actual aircraft fly-overs were measured, simultaneously indoors and outdoors, using the S time-weighting characteristic. Three internal noise monitoring locations were chosen and these normally comprised: (1) main or master bedroom; (2) living room; and (3) kitchen.

The outdoor noise levels were measured by a Brüel and Kjaer Outdoor Microphone Unit type 4921 connected to a Brüel and Kjaer real-time frequency analyser type 2143. The outdoor microphone, placed at 1.2 m above ground level, was set up either in the backyard or near the kerbside in front of the residence concerned, depending on whichever location provided minimum obstruction between the microphone and the flight path in the sky. The A-weighted indoor sound levels were measured using Brüel and Kjaer sound level meters connected to Brüel and Kjaer level recorders with a microphone height of approximately 1.2 m above the floor. Noise emissions from jet aircraft were recorded while smaller propeller-type aircraft were ignored. Measurements suspected of being adversely affected by traffic noise or other extraneous noise sources were discarded. Typically noise levels from more than 30 aircraft fly-overs were measured and the results averaged. Typical standard deviation of the measurements was about 1 dB, although higher values (about 2 dB) were also found. The range of individual flyover noise reduction could be as high as 5–6 dB. For occupied houses, typically about 10–20 aircraft fly-overs were measured and the results averaged.

The first acoustic insulation enhancement step was improvement of the weakest links in the chain which included: sealing gaps around doors and windows; and sealing other openings, e.g. wall vents. The installation of fibrous ceiling insulation was also generally included in this step. The next weakest link that could be easily upgraded was the windows, which generally were initially improved by installing a secondary glazing comprising 6.38 mm thick laminated glass separated by a nominal 100 mm airspace. The existing external front doors were generally left unchanged (except for new perimeter seals) so that the street appearance of the houses was not affected. Further insulation improvement measures were undertaken as necessary, including upgrading the attenuation characteristics of the roof/ceiling construction, which was carried out without replacing the existing roofs and ceilings. The aim of the sound insulation measures was that the residences would achieve the design noise reduction prior to the installation of the mechanical ventilation or air-conditioning systems. These were installed as separate items, with the goal that the systems' ductwork would not cause any noticeable deterioration in the sound insulation already achieved. It should be noted that some form of mechanical ventilation or air-conditioning is essential for the well-being of the occupants after the house has been acoustically insulated.

Results and Discussion

Existing ANR external design levels

The space available here does not allow a detailed description of the insulation improvement steps and the corresponding benefit achieved for all residences; these details are available elsewhere in a pilot project report[(2)].

The existing average ANR of the bedroom, living and kitchen/dining areas of the 17 houses studied in this work, rounded to the nearest dB(A), are given in Table 1. The 18th residence did not have a kitchen or living area as it was a part of a larger youth centre and is not included. For houses with open kitchen/living or kitchen/dining areas, the same ANR is assumed for both areas.

The results show that for bedrooms, the initial ANR was in the range 20–30 dB(A), except for House 16 which had 1 dB(A) higher than the above range and House 13 which already had some insulation improvement work carried out by the owner. These ANR values may be taken as representative of the ANR expected from other houses in the affected suburbs. Given that the external design sound level in the ANEF 30–40 zone can easily exceed 85 dB(A), it suggests that houses without any

Table 1. House Nos 1–17: measured aircraft noise reductions before insulation improvement

House no.	Initial aircraft noise reduction (dB(A))		
	Bedroom 1	Living	Kitchen
1	21	18	18
2	24	21	17
3	27	20	16
4	24	25	25
5	27	19	11
6	28	21	18
7	29	27	27
8	24	26	31
9	20	25	21
10	28	20	24
11	23	25	21
12	26	21	26
13	37	30	30
14	25	32	27
15	26	23	18
16	31	28	25
17	25	24	22

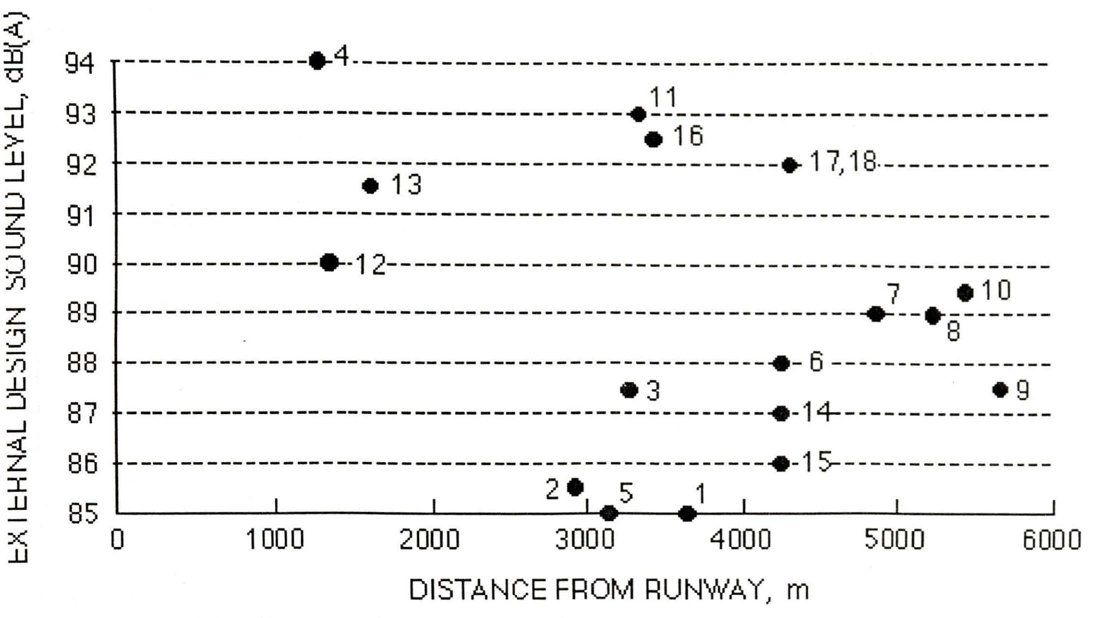

Figure 1. External design sound level as a function of distance from runway.

acoustic treatment in this zone are likely to fall short of the internal design sound levels recommended for bedrooms in AS 2021. The living rooms tended to have ANR a few dB(A) lower than the bedrooms except where they were in a shielded position in the house. The kitchen/dining areas had ANR that were a few dB(A) lower still than those for the living areas or bedrooms. The factors that can contribute to this poor acoustic performance are: (1) the presence of hard non-absorptive floors in these areas; and (2) the presence of exhaust fume-extraction systems to remove cooking odours.

The calculated external design sound levels of all the houses are plotted in Fig. 1 as a function of distance from the nearest end of the Sydney airport runway 34L. As can be seen, the lowest external design sound level is 85 dB(A). The figure also shows that the design sound levels do not decrease linearly with distance from the runway. This apparent anomaly can be easily explained if one realises that the sideline distance DS from the extended runway centre-line plays an important role in determining the noise level exposure of a given site, and for a given distance from the runway, the aircraft noise level is maximum for locations directly underneath the flight path and decreases as the DS is increased.

Improvement by low-cost measures

The ANR measurements showed that after making low-cost acoustic improvements such as installing seals on external doors and windows, closing wall vents and installing ceiling insulation batts, only small improvements in ANR are likely, say about 3–5 dB(A), which on their own will not be sufficient. The results from several houses also suggested that fixing door seals to internal doors is generally not necessary unless a separation of treated/untreated areas is being considered or unless the external design sound level is very high.

Improvement by secondary glazing

The ANR improvement in bedrooms by installing secondary glazing (after low-cost measures have been undertaken), is shown in Fig. 2 for three residences. The acoustic benefit is highest for House 8 which was a mid-floor flat in a modern high-rise block. In this case, the ceiling and floor were concrete and the walls were cavity-brick, so the predominant path for the entry of sound into the bedroom was through external windows. The secondary windows installed in this particular case comprised 10.38 mm thick laminated glass. The lower acoustic benefit observed for Houses 2 and 3 is more representative of the benefit likely to be achieved in houses where other sound entry paths exist.

Improvement by upgrading roof/ceiling

For brick houses, after low-cost improvement measures and secondary glazing, the next weakest component is likely to be the roof/ceiling construction. For all houses included in the study, existing roofs and ceilings were not altered and the same acoustic treatment was used in both pitched roofs and flat roofs. The improvement measures consisted of installing flexible barrier sheeting such as Wavebar™ and Acoustiflex™ on top of the ceiling joists over fibrous insulation. The material was laid in one or two layers by overlapping (by 100 mm) and taping adjacent sheets, stapling to the ceiling joists at 900 mm centres, and extending to the edge of the ceiling over the top of external walls. Measurements on one house showed that this upgrading measure is not likely to be acoustically effective unless secondary windows have been installed first. The target internal sound level for bedrooms (namely 50 dB(A)) was achieved or nearly

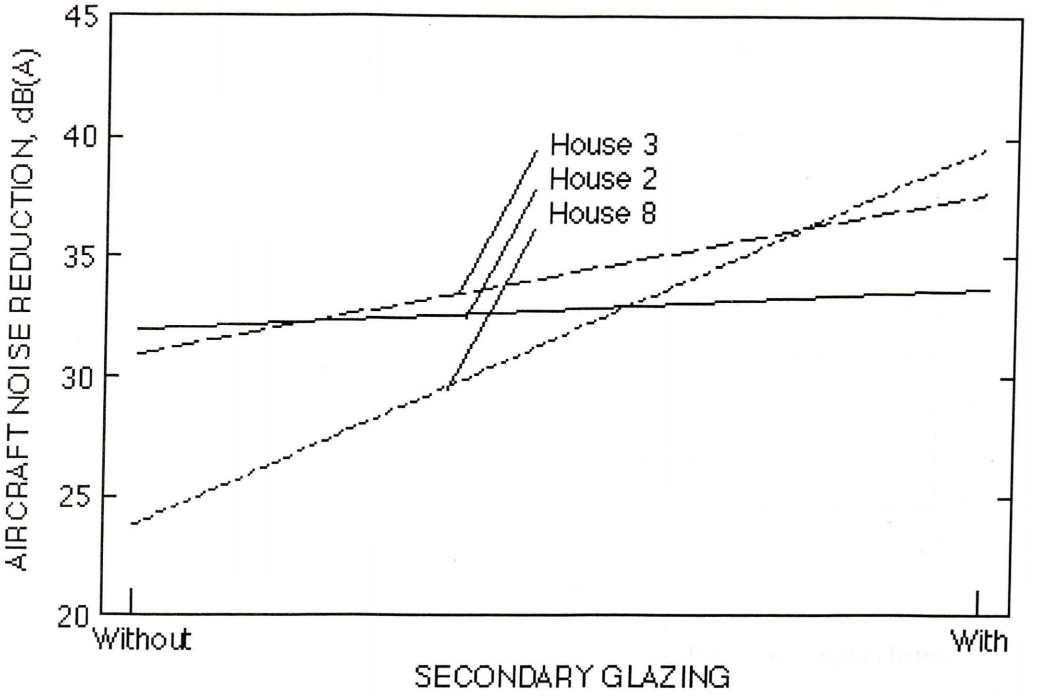

Figure 2. Aircraft noise reduction for Bedroom 1 of Houses 2, 3 and 8 with and without secondary glazing.

achieved after this upgrading measure. The final ANR values for Bedroom 1 of the houses studied are given in Table 2. It should be mentioned that in some cases additional specific measures were undertaken and their effect is also included in the final ANR results given in the table. For House 6, the acoustic improvement achieved is small because of the weatherboard construction, practical difficulties and the reluctance of the owner to have the house's internal lining replaced. In House 17, the existing roof/ceiling was not changed at all as the owner had a plan to add a second storey in the near future which would shield the existing roof/ceiling, and thus explains the reason for the apparent poor improvement recorded.

The step-by-step ANR improvements achieved in Bedroom 1 of Houses 2, 3 and 7 are shown in Fig. 3, where base attenuation representing open bedroom windows is assumed to be 10 dB(A).

The measured results indicated that shielded bedrooms may provide 3–4 dB(A) higher ANR than a bedroom at the front of a house. The results on one weatherboard/timber house included in this study showed that modest improvements in acoustic insulation are possible by low-cost measures as for brick houses, but the external walls may then become the weakest link and would require upgrading before the potential benefit of secondary glazing can be realised. Skylights were common in many houses to compensate for insufficient natural light. Their acoustic performance was improved by installing a hinged timber sash with seals at the ceiling level and lining the interior of the cavity with sound absorbents. Commercially

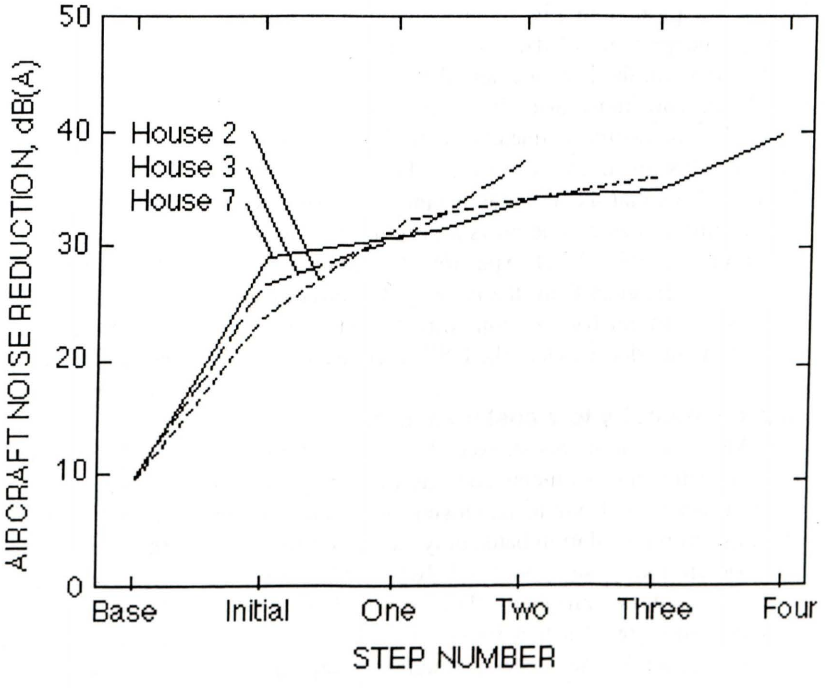

Figure 3. Aircraft noise reduction for Bedroom 1 of Houses 2, 3 and 7.

Table 2. House Nos 1–17: measured aircraft noise reductions before and after insulation improvement

House no.	External design level (dB(A))	Initial ANR Bedroom 1 (dB(A))	Final ANR Bedroom 1 (dB(A))
1	85	21	36
2	85.5	24	36
3	87.5	27	38
4	94	24	41
5	85	27	37
6	88	28	32
7	89	29	37
8	89	24	40
9	87.5	20	40
10	89.5	28	37
11	93	23	40
12	90	26	38
13	91.5	37	40
14	87	25	34
15	86	26	36
16	92.5	31	40
17	92	25	31

available 4.5 mm acrylic snap-on magnetic windows were also found to be useful for this purpose.

Conclusions

The aircraft noise insulation work reported here showed that for houses of brick construction located in areas where the external design sound level is 90 dB(A) or less, it should be possible to achieve 50 dB(A) or 60 dB(A) targets for internal sound levels. The achievement of these levels of noise reduction would require sealing gaps, vents and openings, installing door seals and secondary glazing, and upgrading the existing roof/ceiling construction. For brick houses located in areas with external design sound levels greater than 90 dB(A), it is unlikely that 50 dB(A) internal sound levels will be achieved in all cases by conventional noise attenuation improvement measures (i.e. not replacing existing roof/ceiling structure and keeping secondary glazing thickness within normal domestic construction limits). However, the internal sound level target of 60 dB(A) can be met even for these external levels. In most cases, internal door seals are not likely to be necessary.

Many houses located in the ANEF 30–40 zone are exposed to aircraft noise levels in excess of 85 dB(A). The external design sound levels do not decrease linearly with distance from the runway, but are strongly influenced by the offset from the extended centre-line from the airport runway.

The work showed that it is possible to achieve the desired ANR without the use of additional layers of plasterboard or other rigid sheeting in the ceiling and that the same acoustic upgrading technique can be successfully applied to different roof types. The use of fibrous thermal insulation in the ceiling alone is only likely to give marginal improvement and is not sufficient to provide the necessary noise reduction in areas with high aircraft noise exposure.

Acknowledgments

The extensive support provided by the Australian Operational Support Services (AOSS) for the work reported here, in particular by the AOSS Project Manager, Mr Ken Butler, is gratefully acknowledged. This paper has been published with the permission of the federal Department of Transport who sponsored the pilot study.

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Session 4B

Concert Noise

How Loud is the Orchestra ?

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Introduction

It has long been established that there is potential for noise induced hearing loss for musicians. However, as with all noise exposure, the risk is related to the exposure time and the energy absorbed. There is considerable variability in the noise exposure of musicians which is related to many factors, but the maximum noise exposure will be governed by the output power of the instrument. In this paper I will examine some aspects of the variability of exposure and attempt to establish the likely risks from each instrument group by looking at the maximum output power.

Noise levels in an orchestra

There is considerable difficulty in predicting the average long term exposure of musicians to noise due to the variation in performance schedules, time off and the dynamics of the various pieces being played. This is particularly the case for the Australian Opera and Ballet Orchestra where a cycle of Opera and Ballet works is planned up to 5 years in advance to ensure availability of conductors, lead singers, prima ballerinas and sets.

The extract (fig 1) from a previous study Mikl[2] showed that the variation of noise exposure between performances due to the instrument played and type of opera was significant. The overall exposure could not be determined from simple measurements but rather had to be extrapolated from a series of measurements and calculated from rosters and performance schedules.

Unfortunately opera performances are scheduled years in advance and therefore an estimate of potential noise levels is necessary, to enable future schedules and rosters to be planned. This paper looks at some possibilities available to predict probable orchestra exposure for future events.

Interpretation of past performances

Typical performances were analysed in detail to ascertain if a pattern existed which would enable overall exposure levels to be estimated. Figure 2 shows the relationship of the measured levels for performances of Aida L_{Aeq} 93.7 and the Marriage of Figaro L_{Aeq} 81.0.

It has been widely accepted that the 'Equal Energy Concept' is valid and accordingly that exposures of high energy will only require a short time to deliver a 'daily dose' of noise. What is often overlooked is just how short the time intervals are which can deliver excessive energy to the ear. Figure 3 shows that the time exposures need only be short to not allow recovery to below 85 dB(A) during the rest of the working day. It can be seen that a total of only 15 minutes are required at 100 dB(A) to exceed the 85 dB criteria. By examining Figure 2 it can also be seen that it is not uncommon to exceed 100 dB(A) during a performance.
The music levels throughout any performance are variable and therefore the estimate of the total time at levels above 100 dB(A) is difficult to establish. Further

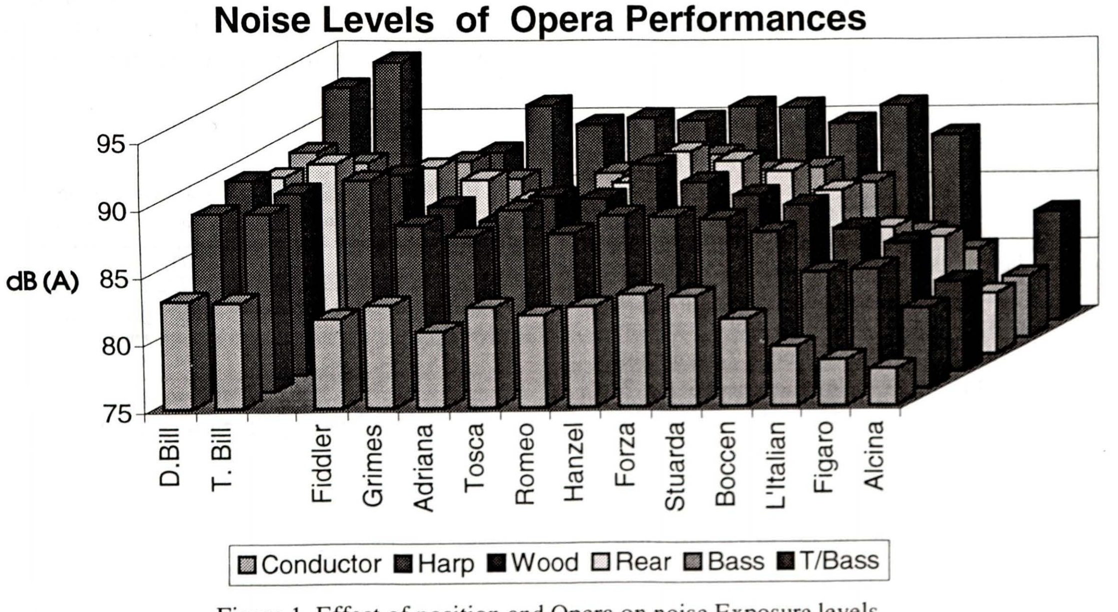


Figure 1. Effect of position and Opera on noise Exposure levels

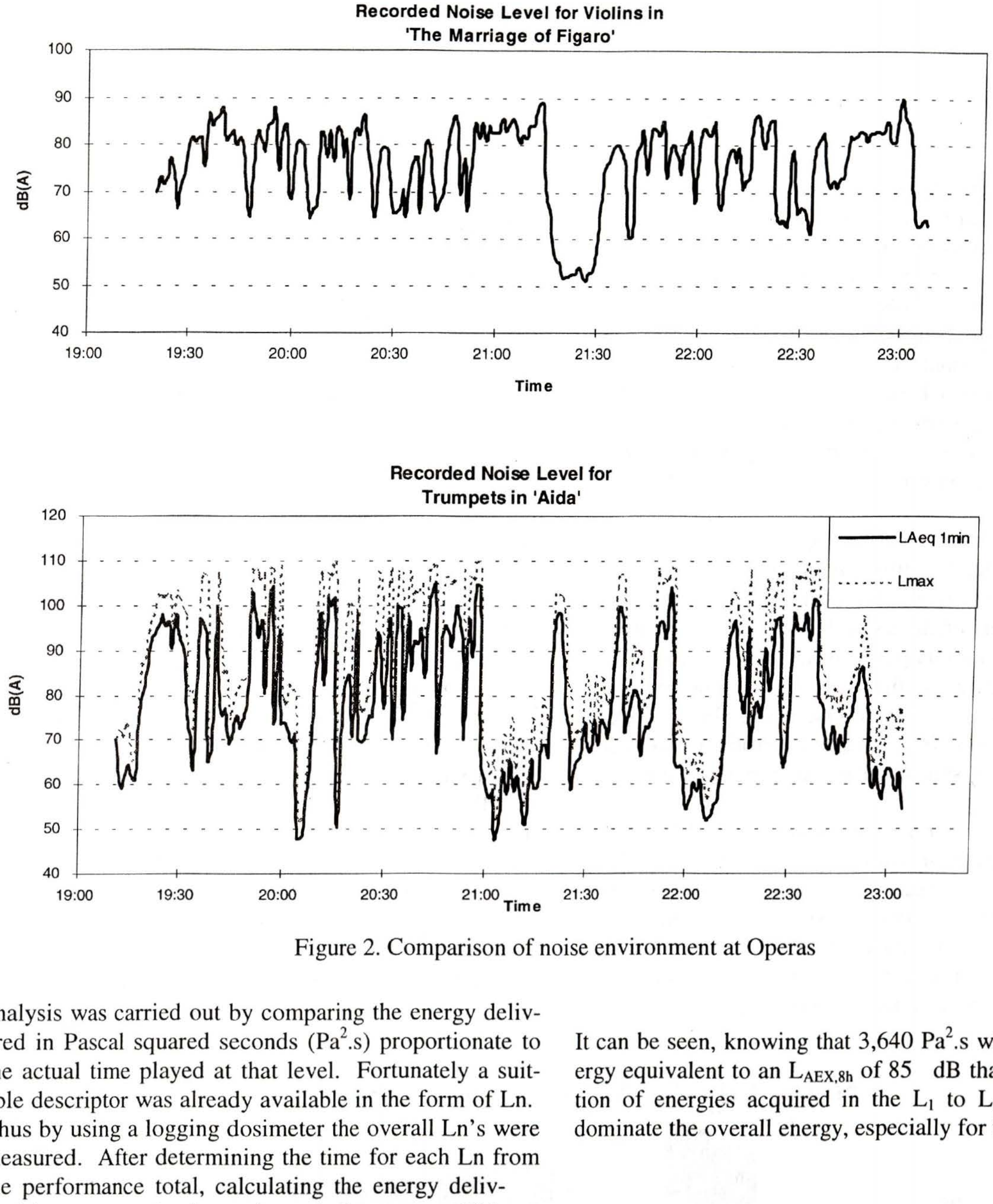


Figure 2. Comparison of noise environment at Operas

analysis was carried out by comparing the energy delivered in Pascal squared seconds ($Pa^2.s$) proportionate to the actual time played at that level. Fortunately a suitable descriptor was already available in the form of Ln. Thus by using a logging dosimeter the overall Ln's were measured. After determining the time for each Ln from the performance total, calculating the energy delivered becomes a simple mathematical exercise. This is shown at Figure 4.

Unfortunately the overall $L_{EX,8h}$ obtained by this method was below the measured $L_{EX,8h}$ consistently by 2-3 dB. This however does not detract from the theory but merely highlights the effect of short duration peaks. Each L_n value spans about 2.5 minutes. Thus the L_1 is the dB(A) level exceeded for 2.5 minutes. For Aida L_1 was 106.5 and it included an L_{max} of 110 dB(A) and an L_{peak} of 130 dB(A). Therefore the underestimation is to be expected and is a result of the descriptor used. However without another parameter available the use of L_n still yields useful results.

It can be seen, knowing that 3,640 $Pa^2.s$ will deliver energy equivalent to an $L_{AEX,8h}$ of 85 dB that the summation of energies acquired in the L_1 to L_{10} bands will dominate the overall energy, especially for the trumpets.

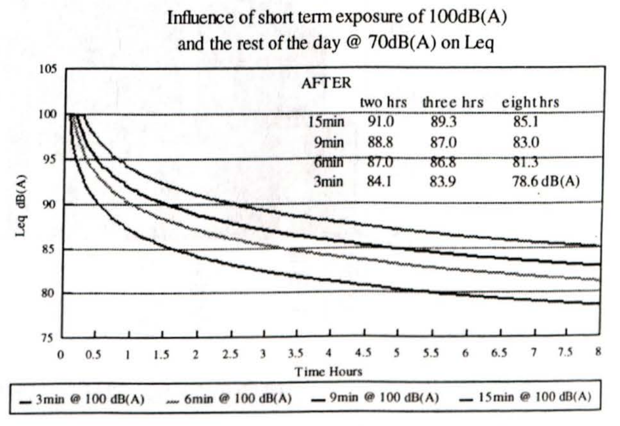
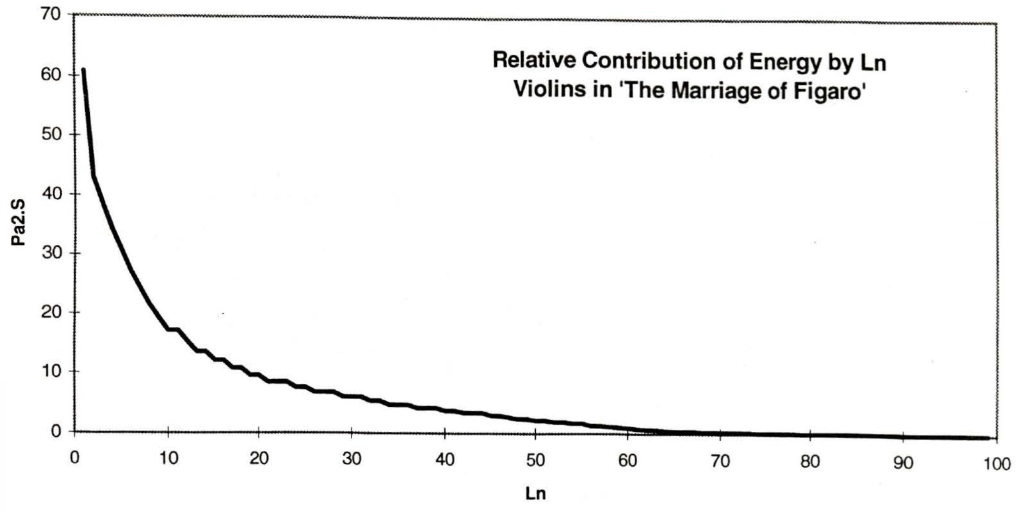


Figure 3. Effect of short term high energy noise



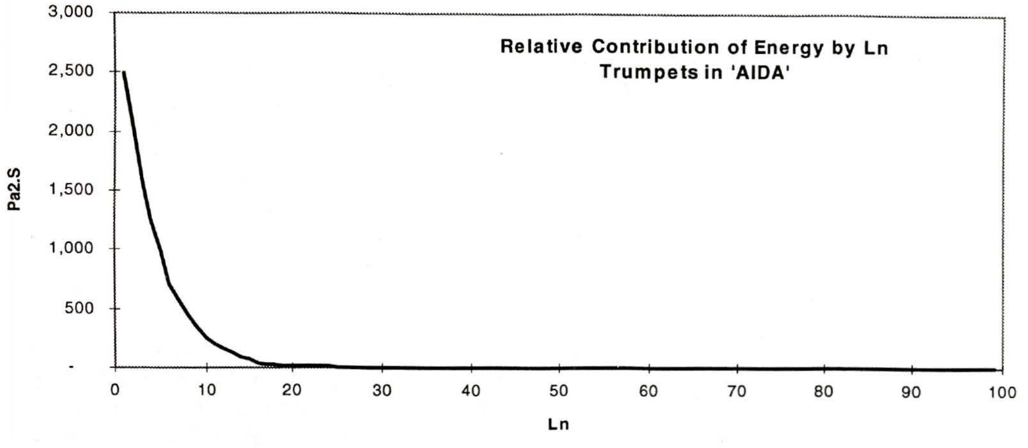


Figure 4. Comparison of Energy output over time during performances.

Prediction strategy

An attempt at predicting the overall L_{EX} of a performance was thought to be possible by standardising the instrumental groups into soft, medium, loud, and maximum then determining the time played at each level from the score and hence predicting the overall L_{EX} . From the foregoing discussion we only need to know the length and loudness of loudest parts to establish the L_{EX}. If the maximum sound power of each instrument was known it would only be necessary to count the total length of time played at or about that level. (This might be established from sections of the musical score.) Multiplication by the number of instruments and taking account of the environment could give an estimate of the overall L_{EX} .

Sound power determination

The sound power of a musical instrument is of course a variable quantity, depending on the method of measurement, the music played, the emphasis of the musician and the instrument itself. To give a realistic power some of the variables need to be identified and taken into account.

The method of measurement

The rehearsal hall was chosen for the measurements. To minimise the reverberation time the front wall was completely covered with heavy drapes, large amounts of curtain were placed behind the players and a number of absorbent office partitions were introduced into the corners of the room. The section was positioned so that it was not on any room axis.

The Survey Method from AS1217.7[1] with the increased number of microphone positions as per AS1217.6[1] was adopted as a suitable method. However, knowing the variability of the source, it was decided to mark out the hemisphere and suspend the 10 microphones at the appropriate positions. The musicians were then placed in the centre of the hemisphere and the noise level was logged each second during play at each microphone position. At the conclusion of the testing all instruments were downloaded and the data for each instrument/dynamic level was extracted and analysed.

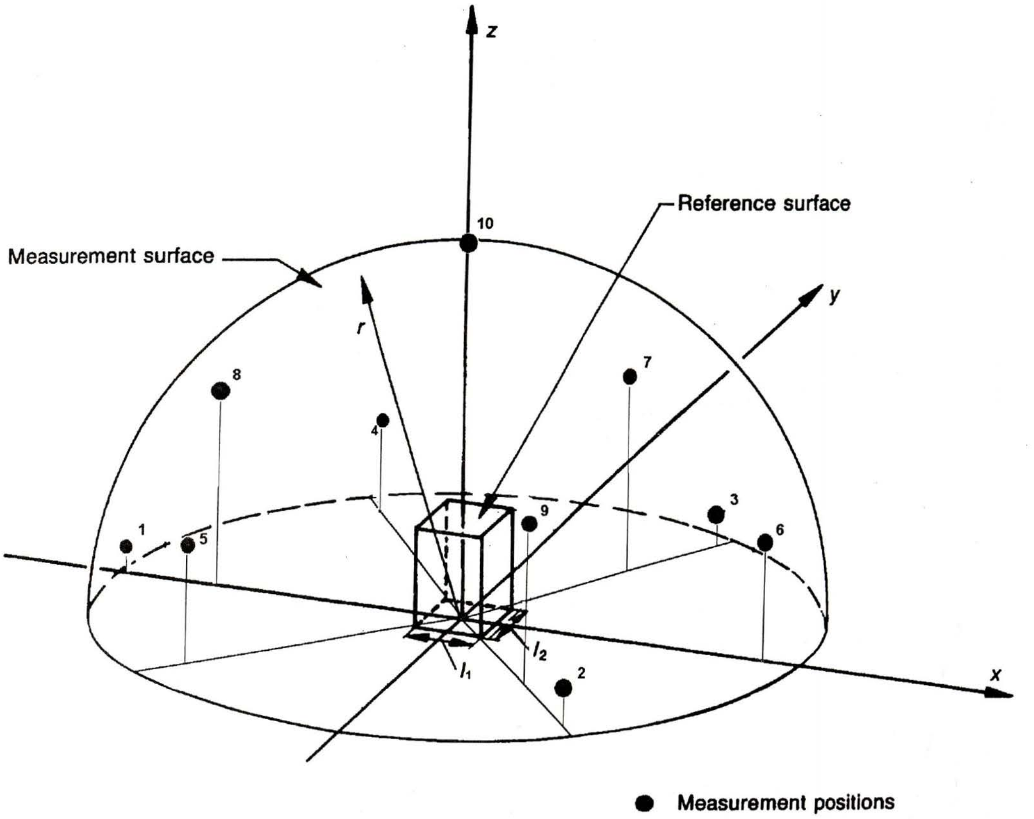


Figure 5. Microphone positions in relation to the players.

The music played

A selection of music was chosen from the opera Tosca. This opera contains the full dynamic range of volumes and was at the time being performed and therefore well rehearsed. For each instrument representative pieces were chosen and designated Soft, Medium and Loud. These were chosen by the conductor and orchestral management.

The emphasis of the musician and the instrument

To standardise the variation of playing, each piece of music was played by three musicians all positioned within the one metre square at the centre of the hemisphere. Each piece was rehearsed and then played. The use of three players from each section was felt to be a good compromise which would average out the playing styles and instrument output for each section. To ascertain the possible maximum sustained output the Loud section was replayed with the players sacrificing clarity for loudness. This section was then used to determine a maximum sustainable effort, and hence a theoretical maximum which would never be exceeded in normal playing.

Results

The Power for each level/instrument was determined by a power summation over the surface area of the hemisphere of the 10 microphone positions for the full length of each played piece. A correction was subtracted for the reverberation time of the hall and the overall power was then divided by the number of instruments (Figure 6). It should be noted that even for the Maximum this is not an instrument L_{max} for a single blast, but the maximum short term effort which a player can produce over a set piece of music.

The figure has been arranged in order of increasing maximum sustainable effort.

Some interesting points to note are:

- The strings have limited dynamic range with the full range of play being from 74-89 dB(A) for continuous music. Of course this is why there are many strings in an orchestra.
- The brass have a better range (79-110 dB(A)) but the maximum sustainable effort is not as loud as many persons expect. This is due to the need to follow real music and not just blow the loudest possible note.

To predict how the orchestra behaves as a group the individual instrument powers were multiplied by the numbers of instruments present in a performance of Tosca. This changes the power significantly as the ratio of the numbers of instruments increases to 18 violins (87-102 dB(A)) vs 4 trombones (85-116 dB(A)).

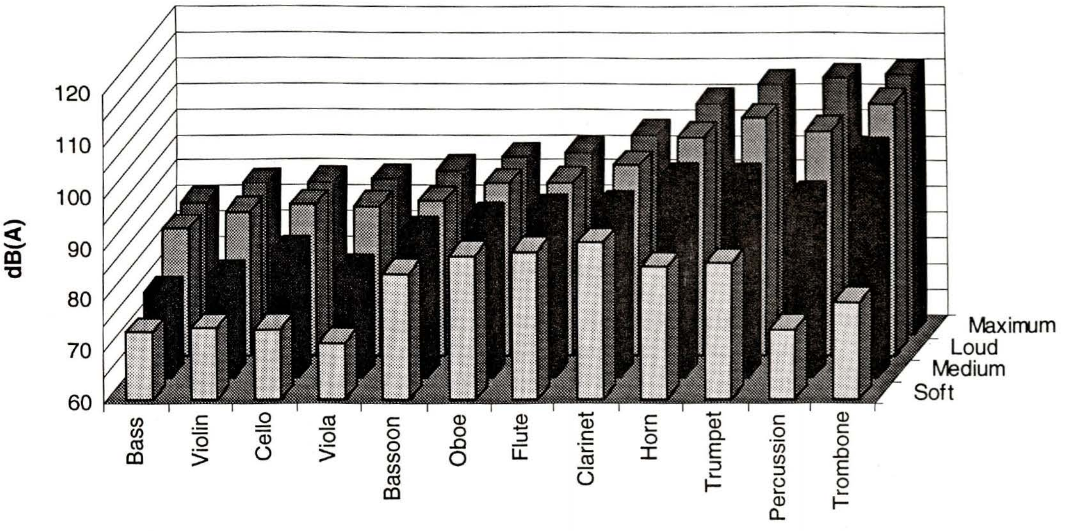


Figure 6. Power output of instruments

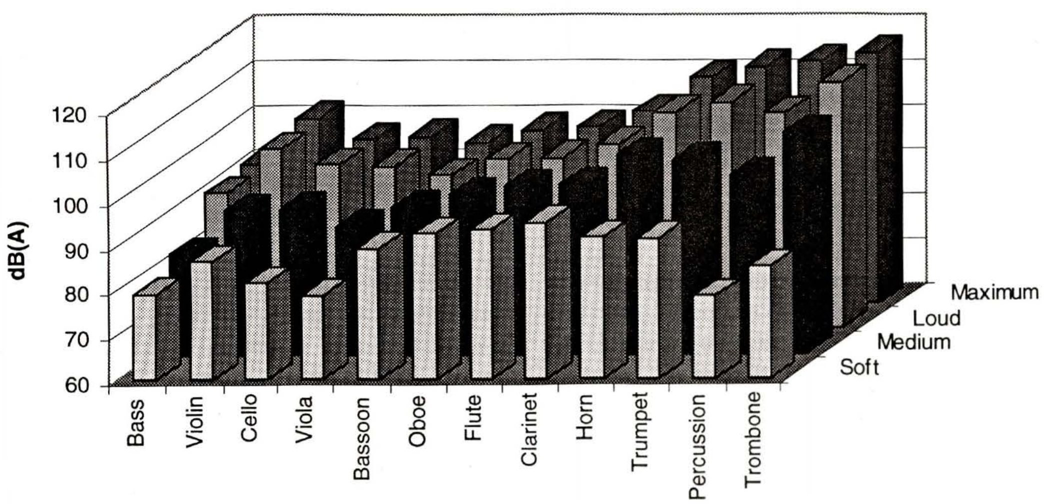


Figure 7. Power output of Sections of the orchestra as required for the opera Tosca

Conclusions

The daily noise exposure due to opera performances is caused by short but loud parts of the opera. These are in the main the overture and the dramatic conclusions to scenes or the end of acts. The maximum possible contributions are from the Horns, Trumpets, Percussion and Trombone. If these could be controlled or modified a significant improvement to overall noise exposure of the orchestra could be gained..

Acknowledgments

The cooperation of the Australian Opera and Ballet Orchestra Management and players without whom this task would not have been possible is gratefully acknowledged. Special thanks also to Klaus Mathiasch of the Acoustics Unit for his help in setting up the experiment and the extraction of the raw data.

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Orchestral Noise ?

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Abstract

In recent years there has been considerable interest in the increasing problem of noise in the work place. The incidence of hearing loss and tinnitus that arise as a result of injury to the ear from excessive noise exposure is of concern with the hearing health of the community. One difficult area of noise exposure to address occurs in orchestras, large and small. Good hearing is essential to orchestra members and hearing damage can be disastrous for their career. What are the typical noise levels to which orchestra members are exposed and just what is the solution to reduce this potentially damaging exposure?

Introduction and background

Over the last few years there has been an increasing awareness amongst musicians that they are extremely vulnerable to noise injury resulting from the constant exposure to high noise levels from their music. The resulting noise injury can lead to a handicap in the form of a hearing loss and/or the presence of tinnitus. Initially it was the rock music industry that was the cause for concern as most uninitiated listeners saw rock music as simply unwanted loud noise. However, it was soon realised that the problem also arises with classical or in fact any form of music.

There have been several studies of the noise levels to which musicians are exposed. These include Rosser (1992), Camp and Horstman (1992) and Sabesky and Korczynski (1995). All high light the difficulties, current and potential, that musicians experience. Problem areas mentioned include the dynamic range of the music, exposure to loud pure tones, impulsive noises and crowded conditions.

For many years the research group involved with Hearing Loss Prevention at the National Acoustic Laboratories has had involvement with musicians who have shown signs of hearing loss and tinnitus. Musicians have found that it affects both their private and professional lives. NAL became involved with the measurement of the noise exposure levels of orchestral members from two organisations. The first was for musicians playing in a "pit" setting for a musical concerning the story of a ghost at a singing venue; while the second was for a large Australian organisation employing several large Symphony Orchestras. The symphony orchestras performed in venues that were both in the open layout and the "pit" style.

There have been several documented studies of orchestral musicians (see references) and there is much anecdotal evidence concerning rock musicians as to the effects of music on hearing. Also many musicians from the percussion section of orchestras can also relate stories of the hearing problems of their teachers.

Measurements

The main measurement was the event A-weighted average sound pressure level over the performance or rehearsal time, the $L_{A,eq}$. The second parameter measured was the maximum peak level during the event, MAXP. Some of these MAXP measurements were linear and some were A-weighted. This depended on the type of instrument being used and the scale(s) available on the instrument.

The measurement period varied depending on what music was being played and the venue. It was felt that the most representative measurements would be those that simply looked at the noise levels over the whole time that members of the orchestra spent at the venue. Thus equipment was set to record the period before play commenced including tuning up and finished at the end of the performance/rehearsal. If there were significant intervals or breaks during the rehearsals instrument readings were recorded at this time in order to minimise any possible loss of information if equipment happened to malfunction for any particular reason.

Instrumentation

The instrumentation used through out the series of measurements consisted of a variety of Sound Level Meters. The meters used were Brüel & Kjær Model 2231 Precision Integrating Sound Level Meters, Cirrus Research CRL 701 Noise Logging Exposure Meters and CRL 702 Integrating Sound Level Meters. All instrumentation was checked for level before and after each performance/rehearsal.

Measurement positions around the orchestras varied from event to event. The single fixed position was that in front of the conductors lectern. This position was taken as a reference point.

Other measurement locations were selected either as the result of a request of members of the orchestras or on a judgement made as to possible "noisy" areas taking into account both the nature of the music being played and the instruments involved. It was initially hoped that the members of the orchestras would wish to be more involved with selecting the measurement locations. However, in the main positions were selected by those carrying out the measurements on the basis of discussions with orchestra members and on the music program and arrangement of the orchestra.

Measuring instruments were either mounted on tripods with the microphone about 1m high or fastened to the musician's instrument stand with the microphone about 1.5m high. Microphones were set at a height which was conducive to optimum performance while remaining as inconspicuous as possible.

The measurements taken at each position were such as to give the approximate total noise exposure level that a musician at that location could expect to receive. It would be a very difficult task to state exactly the noise levels that any particular musician was exposed to for any particular performance/rehearsal. Levels vary depending on many factors such as the conductor, the layout of the orchestra, the enthusiasm of the players and the acoustic environment. Thus all levels expressed in this report should be taken as a guide as to exposure levels and not as definitive statements.

Results

What are the Noise Levels?

Remember here that we are *not* talking about music at a disco or "rock" music venue. Such a venue might be at an RSL Club or local Bowling Club. We are looking at a stage setting playing classical or more "popular" music.

Table 1 summarises some of the measured exposures.

There are some other unofficial results that are well worth quoting here. These are for a more recent stage performance concerning a single Asian lady. Measurements from a B&K noise dose meter worn by a member of the orchestra in the pit produced the following readings: sample time = 2h 55m; $L_{A,eq}$ = 96.7 dB(A); PND = 172%. Comments from members of the orchestra indicate that they consider the music to be much louder than "Phantom" and even after wearing personally moulded ear plugs that nominally supply approximately 25 dB of protection, they still have ringing in their ears the next morning.

Discussion

The figures above confirm that those working in these sections of the music industry receive a significant noise exposure during a performance. Equivalent levels are measured during rehearsals, however, during a rehearsal the levels are usually lower as periodically the playing ceases for discussion and direction by the conductor.

How can musicians protect themselves against this potential hazard? This is a very difficult problem. There are the straight physical or scientific/engineering problems and the "human" problems. The scientific/engineering problems include the difficulty of reducing the noise levels to which the musicians are exposed and/or the use of personal hearing protection. While the human problems include actually being committed to wearing the personal hearing protection, the layout of the orchestra and the desire to reduce the level of noise to which the individual musicians are exposed while not affecting the music or performance in any way.

Reducing the overall noise level of the orchestra is not a very practical solution particularly in the case of a symphony orchestra at a venue where the music is not amplified for the audience. The use of "clear plastic shields" has been tried in the US by Camp and Horstman (1992) and in New Zealand by Rosser (1992). Camp and Horstmann concluded that *"freestanding clear plastic shields provide little protection for the musicians downstream from a given sound-generating source"* (p 92). Rosser found from an earlier study that there could be a reduction in noise level of up to 3 dB but in his 1992 study *"it was interesting to note that the perspex screens were not effective in the new rehearsal studio. This was most probably due to multiple reflections and high levels of reverberation in this studio"* (p 15).

The use of some form of clear screen is regularly suggested for the obvious reason that vision is unimpaired while sound should be blocked. However, the provision of multiple, acoustically hard reflecting surfaces may present many acoustic paths for the sound thus 'trapping' the sound in the vicinity of the musician. This acts so as to raise the noise level from the musicians own instrument while blocking some sound from adjacent instruments. Screens can also cause a "distorted" sound to be presented to the audience and the conductor from the resultant multiple paths. This can clearly be demonstrated by simply listening to an orchestra from the rear as opposed to the normal frontal position can result is a very different perception of the music.

Chassin and Chong (1994) suggested that some degree of protection was provided by the musicians themselves in the brass and woodwind sections. This occurs as a result of the blowing action producing a back pressure in the middle ear. Chassin and Chong estimated that:

"this slight middle ear dysfunction can amount to the equivalence of a 2 - 4 dBA reduction in noise exposure for bassoon and oboe players (double reeded instruments)and a 1 -2 dBA reduction for clarinets and saxophone players (single reed instruments) and brass players" (p 171).

Table 1:

Type of Orchestra	*Venue & Measurement Location*	*Program*	$L_{A,eq,T}$ *dB(A)*	*Exposure time, T (hr:min)*
Small (20 piece)	Pit 1.Trumpets 2.Drums 3.2nd keyboard 4.Conductor.	Phantom of the Opera, Act II	 95.3 90.3 91.8 89.0	 0:54
Symphony Orchestra (~100 piece)	Open stage 1.Bassoon 2.Cor Anglais 3.Violins 4.Double base 5.Conductor	Tchaiovsky, Symphony #5	 95.4 89.3 87.2 84.9 85.3	 0:57
Symphony orchestra (~100 piece)	Open stage 1.Rear of tuba 2.Contra bassoon 3.Horns 4.Conductor	Xenakis, (contemporary)	 95.0 88.9 85.9 88.0	 0:31
Symphony orchestra (~120 piece + 100 pers choir)	Open stage 1.Prime bassoon 2.Horns 3.Woodwinds 4.Harps 5.Conductor	Mahler's Third Symphony	 93.8 96.3 91.7 86.7 84.6	 2:01
Opera orchestra (~50 piece)	Pit 1.Brass 2.Trumpet 3.Horns 4.Centre of pit	 Salome	 92.8 88.6 87.9 89.4	 2:20
Symphony orchestra (~100 piece)	Open stage 1.Centre of orchestra 2.Horns 3.First violins 4.Double bases 5.Conductor	 Respeghi	 90.9 93.3 85.1 84.3 88.9	 1:15
Symphony orchestra (~70 piece)	Open stage 1.Violins 2.Piccolo 3.Flutes 4.Brass 5.Conductor	 Great Classics	 85.2 84.7 87.7 88.6 84.5	 1:20

The use of barriers does not seem to be a particularly successful approach to the problem. Some success has been experienced with the acoustic lining of orchestra pits. While not having a significant, if any, effect, on the direct noise, the reverberant field is minimised. Orchestra members in the pit at "The Phantom of the Opera" found that, subjectively, there was quite an improvement in the acoustic conditions of the pit after the walls adjacent to the percussion section has been lined with a non-reflective acoustic foam.

An orchestra, such as the Sydney Symphony Orchestra, works in very open conditions when performing at a venue like the Sydney Opera House Concert Theatre. The problems here arise with the direct sound from adjacent instruments which are very often very close in a relatively small area and this is a very difficult problem to solve.

The arrangement of the orchestra is a fixed parameter. There seems to be very little opportunity for rearranging the layout in order to reduce the exposure. This appears to be mainly a cultural phenomenon and based in the traditions and history of the orchestra. The layout of the orchestra has a major influence on how the orchestra sounds to the conductor and to significantly vary the layout would affect the sound of the orchestra.

The use of personal hearing protection presently seems to be the "best" solution or at least the only solution of offering the degree of noise reduction required. Some musicians and orchestras find this a satisfactory solution while others think it intolerable. With the choice being between plugs and muffs. The use of ear muffs is usually quickly dismissed the main difficulties being that firstly they physically get in the way when playing certain instruments and secondly the audience would start to wonder "just what is going on here?" The use of ear plugs is at least discrete.

Many musicians dismiss the use of plugs because of the way the resonance of the head is affected and the consequent distortion of that they experience in the sound of their own instrument. This distortion results from two effects. The first being the change in physical characteristics of the ear canal through the insertion of a plug; and the second from the non-linearity in attenuation across the frequency spectrum that is characteristic of any passive hearing protector. Chassin and Chong (1991 and 1994) suggest the use of "Musicians Earplugs" which have a linear attenuation over the frequency spectrum. This alleviates the non-linearity problem but still leaves the physical change in the resonant characteristics of the ear canal (and head) and the subsequent distortion of the music.

For the use of personal hearing protection musicians will have learn to listen to their music in a "different" way. That is, they will need to learn what the music sounds like while wearing the appropriate hearing protection. Currently for a dedicated musician this is an anathema, something almost unspeakable and almost incomprehensible. To implement such a course of action will take considerable cultural change.

Hall and Santucci (1995) suggested the same "Musician Ear" plug solution in conjunction with some form of on stage and/or in-ear monitoring of noise exposure during performances. For a "rock" band with only four or five members, while not being a trivial task, this approach is certainly a practical and workable solution. However, the difficulties of monitoring large orchestras are two fold. Just where do you monitor when the are 50 or 100 orchestra members and do you monitor, in-ear, all members? It would be an overwhelming logistic task for a full symphony orchestra.

Conclusion

The reduction in the noise levels to which musicians are exposed, protected or unprotected, is not as simple as it might at first appear. While noise experts may propose what they see as a satisfactory solution to an orchestra's noise exposure problem, the orchestra members themselves may not have the same good feelings and enthusiasm about it's implementation.

Before a complete solution can be found, there will need to be some further technical developments and extensive negotiations with the musicians themselves as there does not appear to be a single generic solution.

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Session 5A

Architectural Noise Reduction

Acoustical Feature Extraction From Aircraft And Traffic Noise

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Abstract

For the purpose of developing a real-time transportation noise recognition system, a variety of acoustical features and statistical models of signals are examined. Finding an appropriate acoustical criteria to discriminate between several types of environmental noise is the goal of the present paper. The main emphasis is put on the discrimination of transportation noise, music and speech. The results can be useful to operate any noise activated system sensitive to transportation noise such as intelligent noise monitoring and control.

Introduction

A variety of different parameters and signal processing methods have been examined to distinguish between transportation noise and other environmental sound sources such as speech and music. Some parameters are required to distinguish between the low frequency, random broadband nature of transportation noise and more complex spectra of speech, music and other sources of environmental noise. In this study, aircraft noise, heavy vehicle and mixture of traffic in dry and rainy weather are considered as transportation noise sources.

A real-time intelligent system requires real-time data acquisition, detection and pattern recognition. The pattern recognition problem can be divided into several stages of which feature extraction and source classification are two of the most important ones. The goal of this paper is to find some statistical features from transportation noise signals to be used in a noise activated control system.

The feature extraction methods used in seismology, which deal with low frequency vibration recognition, are found applicable to transportation noise recognition. The criteria for automatic discrimination between nuclear explosions and natural seismic activity are also worth examining. Speech and speaker recognition systems have also employed a variety of signal processing and pattern recognition methods. They may be examined to identify transportation noise.

The energy of a signal, the zero-crossing, the linear prediction coefficients and the autocorrelation function are the time domain functions which have been used for successful waveform recognition in other fields [1,2,3]. The frequency spectrum envelope, averaged-peak frequency, maximum-peak frequency in each band and the plot of the first and the second frequencies are the other features that have already been used for transient events and aircraft type recognition [1,2,8]. The Euclidean distance and the maximum likelihood are the two methods of pattern recognition are applied to compare the pattern vector with the sample vector.

In the following sections the characteristics of transportation noise are dicussed and a signal modelling procedure is explained. A pattern classification method is then presented. Finally, a recognition algorithm and its application to a transportation noise monitoring system is presented.

Characteristics of transportation noise

Aircraft noise and traffic noise are non-stationary and time dependent. Transportation noise emitted by cars, heavy vehicles and aircraft is variable in both time and frequency domains but its acoustic energy is mostly concentrated in the lower frequencies. The amplitude and frequency of land transportation noise is strongly dependent on the acceleration, speed, operating mode and the conditions of the vehicle and road. Also, the distance between the source and the observer as well as the weather conditions are very important parameters. In the case of aircraft noise, the type of aircraft and propulsion system, load, take off or the landing position, the angle of flight and the weather conditions all interfere the characteristics of the emitted noise. Many random parameters such as the noise of a vehicle brakes and the sound of car horns and the vibration of trailers are mixed with transportation noise.

Time variation of energy and sound pressure Level

In real-time signal processing, one of the most important parameters to be measured is the energy of signal. This parameter has been used to differentiate between voiced, unvoiced and silence in [3]. Fig. 1(a) is the result of monitoring the energy of the signal from different environmental noise sources. It shows that the energy of background noise is considerably less than the energy of noise event. When the sensing system is turned off the energy is equal to zero. In the present research an energy threshold level is used to activate the recognition system and discriminate between the "Off" position of sensing system and the "Background" noise. Also, the energy is an indication of the beginning and end points of noise events. The sound pressure level of an acoustical signal is another parameter which can be easily measured Fig. 1(b) shows monitoring of sound pressure level from the same events in Fig. 1(a) under the same conditions. As the logarithmic plot is much noisier than the linear plot, particularly for the values close to zero, the linear plot is more useful for pattern recognition.

In Fig. 1(a) and (b) each number represents a separate class of noise events as follow: 1) the sensing system is off, 2) a B&K 4230 calibrator, pure tone 1 KHz, 3) Background noise, 4) Noise of aircraft VH-CHT, 5) A piece of classic music, 6) Noise of aircraft VH-CZT, 7) Noise of aircraft VH-EWI, 8) A piece of speech, 9) A piece of rock music and 10) Noise of aircraft VH-TOO.

Linear Prediction Model of Time Signal

The "Linear Prediction Model" of time signals is one of the most powerful existing techniques available to discriminate between different waveforms. Different characteristics of this method have been used successfully in the field of speech and speaker recognition [3,4,5]. Also, there is a wide range of applications in the field of seismology to classify nuclear explosions and earthquakes automatically [1,2,8]. The basic idea of the linear prediction model of a signal is to model the signal as a linear combination of its past and present values with a hypothetical input to the system. The input of such a model is white noise or an impulse and the output is the given signal. If x_n is the time signal, the general form of such a statistical model can be represented as:

$$x_n = -\sum_{k=1}^{p} a_k x_{n-k} + G \sum_{l=0}^{q} b_l w_{n-1} \qquad b_0 = 1,$$

$$1 \le k \le p, \quad 1 \le l \le q \qquad (1)$$

where, a_k, b_l are the model coefficients, G is the system gain, w_n is white noise sequence, p and q are the orders of model. The output signal, x_n, is a linear function of past outputs, present and past inputs. The estimated coefficients a_k and b_l are useful parameters for pattern classification. The frequency domain representation of equation (1) is:

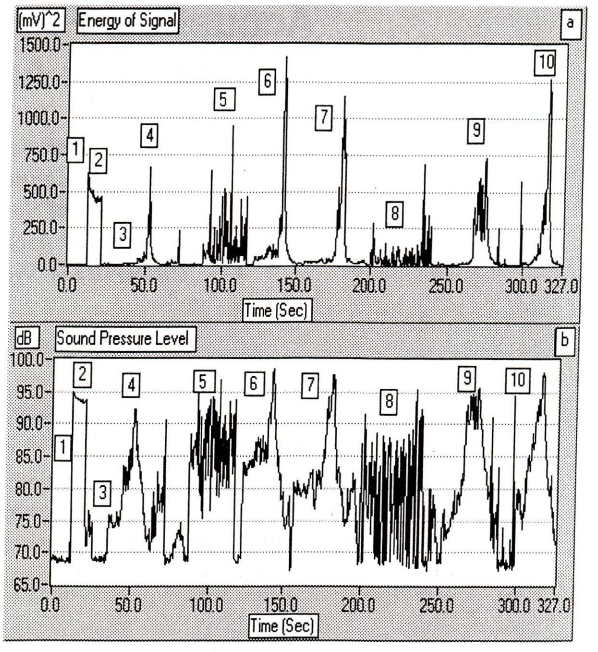

Figure 1: a) Energy monitoring of environmental noise. b) Sound pressure level monitoring of the same noise events shown in (a), sampling frequency is 5 KHz for 512 samples.

$$H(z) = G \frac{1 + \sum_{l=1}^{q} b_l z^{-l}}{1 + \sum_{k=1}^{p} a_k z^{-k}} \qquad (2)$$

where $H(z)$ is the general pole-zero model which is. called an autoregressive moving average model ARMA(p,q). Equation (1) can be simplified as an all-pole model or autoregressive model AR(p), equation (3).

The autoregressive model of a signal, when $b_l = 0$. It can be considered as a recursive filter with feedback as follows:

$$x_n = -\sum_{k=1}^{p} a_k x_{n-k} + G w_n \qquad (3)$$

Where, p is the order of the AR model. The most applicable frequency spectral match for the AR model is found by dividing G^2 by the magnitude squared of the FFT from the sequence of $1, a_1, a_2, \ldots, a_p$. Fig. 2(a) is the result of 60-pole fit (AR) to a FFT signal spectrum computed from the noise of aircraft VH-CZT in landing mode.

Discrimination of AR coefficients

Transportation noise mainly contains low frequency energy and the randomness of the noise is greater than

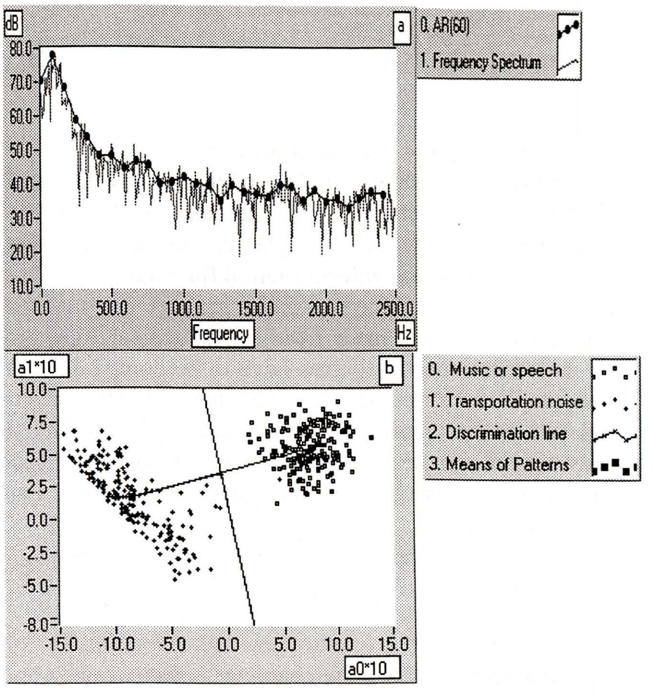

Figure 2: a) Frequency spectrum of autoregressive model with 60 poles, 512 samples from aircraft VH-CZT in landing mode. b) Plot of the first and the second coefficients of 2-pole AR model to discriminate between speech, music and transportation noise.

other acoustic sources such as speech and music. A low order AR model for the 1 KHz filtered data from different aircraft, heavy vehicles and mixture of traffic noise has been examined. There is a strong similarity between the plot of the first and second AR coefficients of aircraft and road traffic noise. The same similarity has been found for classical music, rock music and continuous speech. A second order AR model gives a considerable separation between speech, music and transportation noise with the number of samples less than 128. In Fig. 2(b) the results of 200 data from two pieces of classical music, a piece of rock music and an interview are compared with the same number of data from aircraft VH-CZT noise and a mixture of traffic noise. Fig. 2(b) shows that the second order AR coefficients are useful in recognising the transportation noise. They also have the potential of dealing with low frequency noise recognition.

Acoustical pattern classification of Environmental Noise Sources

After studying different measurable and available features and their abilities to discriminate between different acoustical sources, an attempt is made to fuse the extracted features to make a decision system. As a result of the previous sections, the energy of a signal is a good indication of the start and the end of noise events. In addition the energy of the signal is a discriminating factor between an "Off" signal and "Background" noise. In the present study the pattern recognition task is divided into two classes of recognition:

1. Short-time acoustical source recognition, which is quick enough not to miss the noise event, as the ultimate goal of source recognition is to activate the system as the noise event starts. The data acquisition, data processing, feature extraction and patten recognition must not take more than a portion of a second. In such a short time extracting the detailed information from the acoustical source is impossible. Basically human being's hearing system is also not able to recognise the similar sound sources in a very short time.
2. Long-time acoustical source recognition, which does not take more than 50 seconds duration. This case is useful to identify the particular vehicle or aircraft. To differentiate between different types of aircraft, a satisfactory number of data is required to be compared with the data base already made from statistical manipulation of data recorded from aircraft noise.

Traffic noise in rainy weather contains a lot of high frequency signals that makes it easily identifiable from the other transportation noise. The high rate of zero-crossing in the time signal is the main characteristic of this signal. The 4230 B&K calibrator gives a constant zero-crossing and sound pressure level.

Short-time zero-crossing of time signal

The zero-crossing is the number of zero crossings level by the time signals per duration of sampling. In the case of a pure tone, zero-crossing is a good measurement for frequency, but for broadband signals e.g. transportation noise, it can be a rough indication of frequency concentration. Direct measurement of zero-crossing for a discrete signal is difficult because the value of the signal is rarely zero. The process of detecting zero-crossing at a given time is based on the sign change of the product of multiplication of one data before and one data after that time.

Short-time autocorrelation function

The autocorrelation function has a lot of useful properties real-time signal processing and detection in real-time. An important property of the autocorrelation function for the present study is the considerable difference between geometrical configuration of this function from transportation noise and the other acoustical sources. The main reasons can be the low frequency nature of such a noise and its randomness. The number of peaks and the number of zero-crossing in the case of speech and music, is considerably higher than that of aircraft and traffic noise. The number of zero-crossings of the autocorrelation function in time domain is totally different with the meaning of zero-crossing function of time signals. Fig. 3(a) is the result of 200 plots of the number of zero-crossing of the autocorrelation function with respect to the zeroth element of the autocorrelation function.

The location of peaks

The frequency content of transportation noise is mainly concentrated in low frequencies. Thus, it seems to be reasonable to draw the plot of the first and the second peak frequencies. Fig. 3(b) is the plot of f1 and f2, the first and second frequencies of 200 samples from mixture of heavy vehicle noise and the same pieces of speech and music, that were given in previous sections. The discrimination line shows a good separation between two classes of acoustical sources.

Euclidean distance and maximum likelihood calculation

In the case of short-time data acquisition and pattern recognition the features, eg. AR coefficients or f1 vs f2 from sample data, have to be compared with the related features in pattern space. By considering the discrimination line, which has the same distance from the means of two classes of noise, the distance between the feature from sampled data and the mean of feature from each pattern has to be calculated. The closest distance indicates the type of sample data statistically. One of the simplest distance measurement is Euclidean distance. If the coordinates of the sample feature is *(x, y)* and the means of feature from source-1 and source-2 are located at, (μ_{1x},μ_{1y}), (μ_{2x},μ_{2y}), respectively, then the sample belongs to source-1 if the following condition holds:

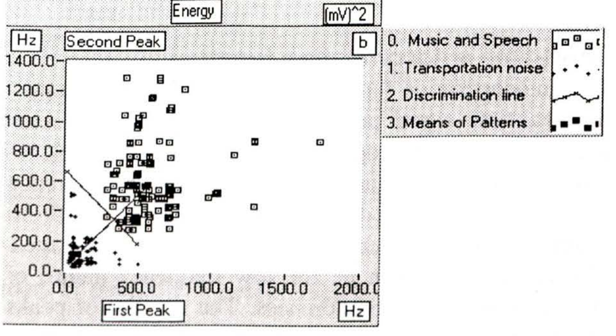

Figure 3: a) Feature extraction from the autocorrelation function for 200 samples of traffic noise, speech and music. b) A plot of the first and the second frequencies of mixture of heavy vehicle noise in comparison with speech and music. The sampling frequency is 5kHz with 512 samples

$$[(x-\mu_{1x})^2+(y-\mu_{1y})^2]^{0.5} < [(x-\mu_{2x})^2+(y-\mu_{2y})^2]^{0.5} \qquad (4)$$

In equation (4) the Euclidean distances between sample and patterns are compared. This method is implemented to discriminate between speech, music and transportation noise based on AR coefficients and the number of zero-crossings in the autocorrelation function.

The pattern vectors made from the acoustical signature of the environmental sources have to be compared with the sample vector made in the same manner. The statistical pattern recognition finds the maximum likelihood between the sample vector and pattern vector. The reference pattern of *m* types of noise source is represented as a vector [6,7]:

$$P_i = (P_{1i}, P_{2i}, \ldots, P_{ni}) \quad i=1,2,..m \qquad (5)$$

The sample pattern of noise character is represented as :

$$S = (s_1, s_2, \ldots, s_n) \qquad (6)$$

The sample vector and the reference patterns vectors will then be compared. To evaluate the similarity between a sample and references, the following set of conditional distance has to be computed:

$$D(S, P_i) = [\sum_{j=1}^{n} (s_j - P_{ji})^2 . W_{ji}]^{0.5}$$
$$i=1,2,..m,\ j=1,\ 2,..n \qquad (7)$$

where W_{ji} is the weight of *n*th parameter for a noise source type and can be defined as the estimated variance of the *n*th parameter for *i*th source. The sample is assigned to a type of noise source where the distance score is the smallest value. The pattern made by the averaged spectrum from 5 types of aircraft have 256 elements per spectrum. The mean and variance of the data are calculated and compared with the same parameters in sample vectors based on equation (7).

Frequency spectral patterns

To identify the type of aircraft or vehicle a short-time data acquisition can not provide enough information to make a recognisable feature. Averaged frequency spectrum is a feature for long-time acquisition. This feature contains the time variation of the frequency spectrum and introduces a stable pattern from the frequency spectrum. An averaged spectrum has been used successfully to identify 5 different types of aircraft. After recognition of transportation noise, 5 types of aircraft will be identified (when the noise event is finished). The main feature to identify the type of aircraft is the minimum Euclidean distance between averaged frequency spectra of sample

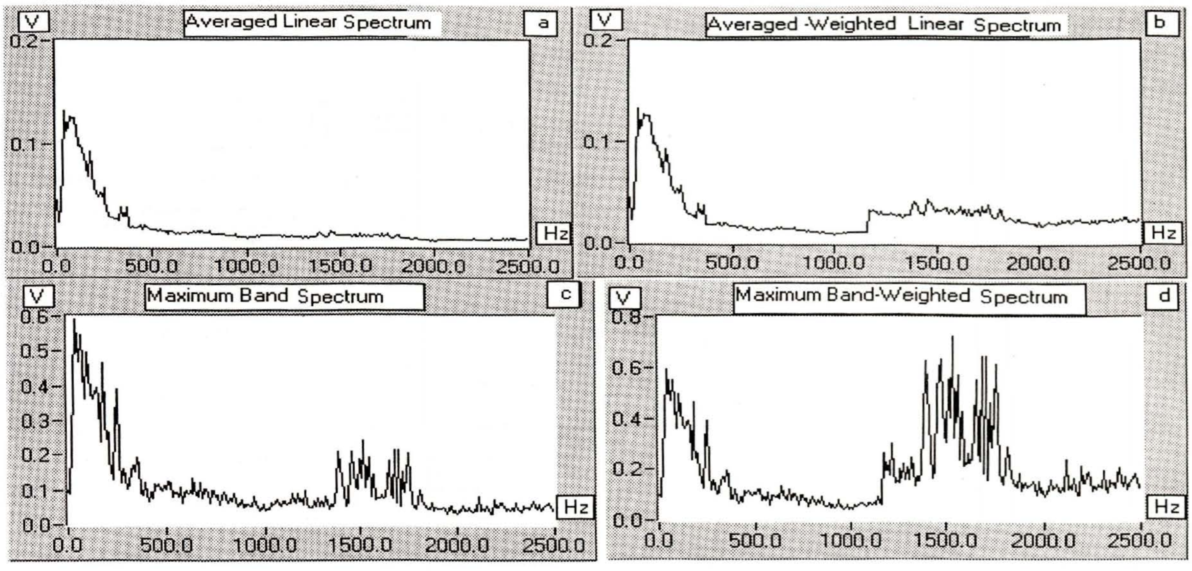

Figure 4: Comparison between different spectral patterns of aircraft VH-TAD noise, a) A pattern of averaged frequency spectrum, b) A pattern of linear averaged-weighted frequency spectrum, c) A pattern of linear maximum frequency spectrum, d) A pattern of linear maximum-weighted frequency.

and patterns. The averaged spectrum of noise can be expressed as follows:

$$X_m = x_{m1}, x_{m2}, \dots x_{mn} \qquad (8)$$

$$Y_j = \frac{\sum_{j=1}^{m} x_{ij}}{m} \qquad i = 1,2,..m, \quad j = 1,2,..n \qquad (9)$$

If the averaged frequency spectrum array is $Y_j = y_1, y_2, \dots y_n$, where n is the number of data in each spectrum array and m is the number of spectrum. Fig. 4(a) illustrates the averaged frequency spectrum pattern made from the overflight duration of aircraft VH-TAD with 512 samples and 2 KHz sampling frequency.

With the aim of searching for the inherent structure of data taken from acoustical noise sources, particularly aircraft, another pattern from the maximum amplitude in each frequency band can be made. This pattern is comparable with the averaged spectrum pattern. The main characteristics of the maximum spectrum pattern is to hold the growth of impulsive frequencies which are not observable in averaged spectrum.

$$Y_j = Max.X_{ij} \quad i = 1,2,\dots m, \quad j = 1,2,..n \qquad (10)$$

The major problem with this feature is its large number of spikes and its instability. We can put emphasis on crucial parts of the averaged or maximum spectrum by weighting the frequencies of interest, e.g. Fig. 4(c) shows the growth of frequencies higher than 1200 Hz and Fig. 4(d) shows the multiplications of those frequencies by 3.

Decision making and aircraft recognition

The order of various features extraction stages and classifiers is crucial in making a right decision about the type of acoustical source. The first thing to know is the correlation between features and sources and realising the stage of decision making based on the extracted feature. The best method is to move from general criteria for a rough classification to specific criteria for a fine classification.

A decision making algorithm based on the discussed features is presented in Fig. 5. The level of energy in short intervals is measured and compared with the threshold levels of background noise and noise events. When a noise event starts, the system will change the sampling rate and start to measure zero-crossing, auto-correlation function and the FFT. Then the AR model of the signal is determined. All information from these features are compared with some properly selected threshold values. After evaluating the type of noise, if it belongs to the transportation noise class the monitoring system will be activated and then the spectral features will be calculated. They are compared with the patterns data base to find whether they are matched to a recognisable aircraft. The acoustical sources to be classified in the present research are: a) Sensing system Off, zero voltage acoustical input, b) Background noise, c) B&K 4230 calibrator, 94 dB, 1 KHz, d) Traffic noise in rainy weather, e) Music and speech, 2 pieces of classic music, 1 piece of rock music and one piece of reading the news by male speaker, f) Transportation noise in dry weather including 5 types of aircraft, a mixture of traffic, a sam-

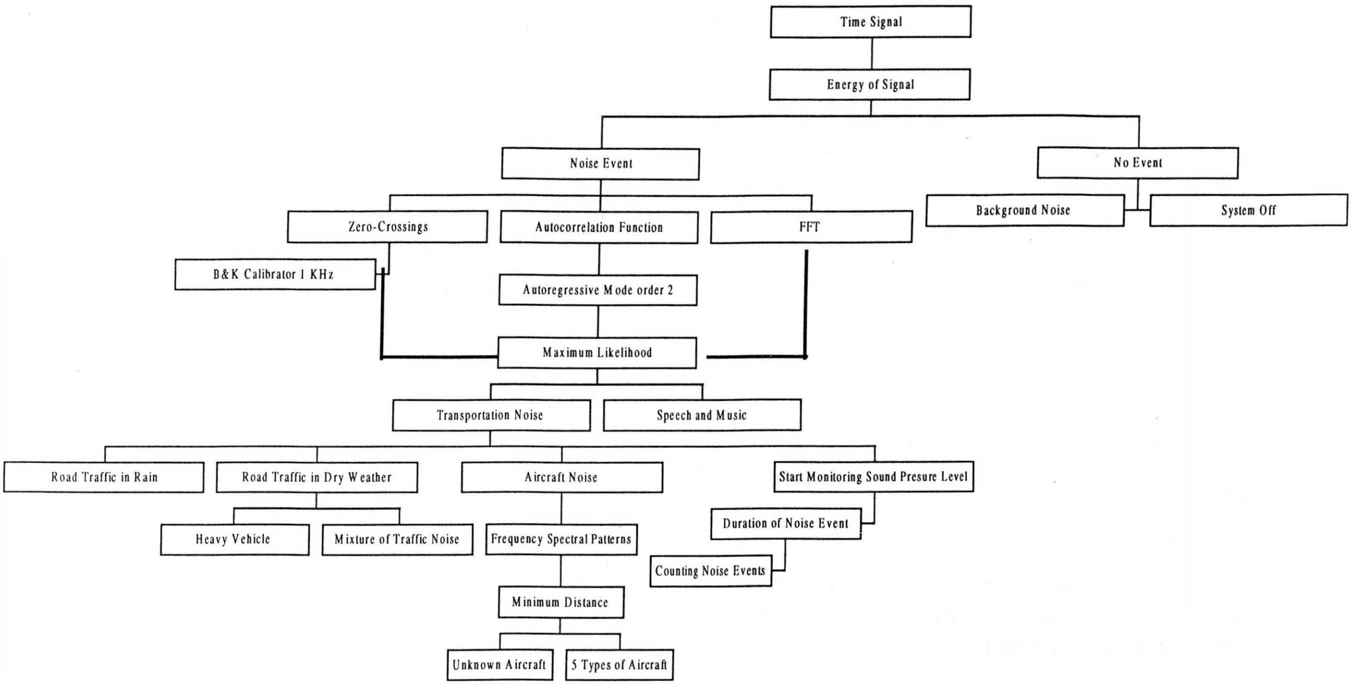

Figure 5: Decision tree for transportation noise recognition and monitoring.

ple of heavy vehicles and a mixture of traffic, train and aircraft noise together.

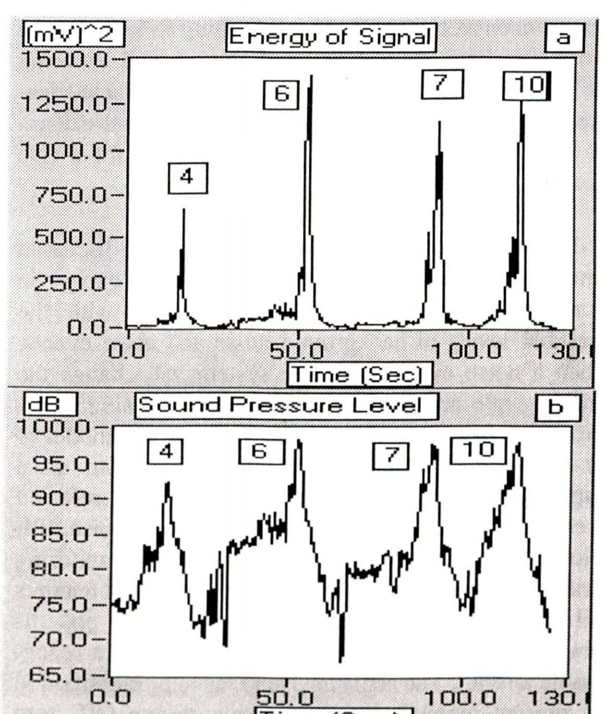

Figure 6: a). The energy monitoring of transportation noise by an intelligent noise monitoring system for the same events in Fig. 1. b). Sound pressure level monitoring of transportation noise by an intelligent noise monitoring system for the same events in (a).

There are two major stages in the algorithm, the first step is training and the second step is operating, which are different from the similar steps in other programs. Both stages are adaptive and will be activated only by transportation noise. The main reason that they are made adaptive is that the duration of overflight is different for different aircraft noise event. Therefore finding exact start and end points of the event can avoid interferences caused in the subsequent stages due to incorrect data. An example of that is the process of averaging the frequency spectrum which has a crucial role in aircraft noise recognition.

Intelligent transportation noise monitoring

A sound pressure level monitoring system can be activated when the traffic noise or aircraft noise dominates the background noise. Such a recording and monitoring system can automatically record the desired noise. This program is able to calculate the time duration of an event and the maximum and the mean of sound pressure level during an event. In a quiet environment this program counts the number of vehicles and aircraft with a noise level more than a proper threshold. The results of operating an intelligent noise monitoring system for the noise given in Fig.1 are presented in Fig. 6. The monitoring system is activated only by transportation noise. When the nature of sound changes or the level of energy and sound falls to less than some particular value the system stops monitoring.

Conclusion

In this paper some useful features to discriminate the transportation noise from other environmental noises such as music and speech have been introduced.

The features used to discriminate the transportation noise from the other sounds are the energy of the signal, sound pressure level, linear prediction coefficients, autocorrelation function, peak frequency and zero-crossing. The AR coefficients and the autocorrelation function provide reliable criteria to discriminate between transportation noise, music and speech in short-time data acquisition. These features have been used successfully to activate an intelligent noise monitoring system. There is also some false recognition when the speech segments are not continuous.

Spectral features for long-time data acquisition are used to recognise the type of aircraft. The result of this research is likely to be applicable to other acoustical noise recognition, such as intelligent noise control [9], counting vehicles, acoustical diagnosis of defects in machines and even to medical diagnosis systems. In the future research the model of other types of environmental noise such as bird sounds and thunderstorm, will be studied.

Acknowledgment

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A Noise Activated Control Approach to Reduce The Transportation Noise

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Abstract

There is a conflict between the need for supplying fresh air and passive noise control by conventional methods, particularly near airports, railway lines and beside the major trucking routes. A "noise activated system" sensitive to transportation noise is an automatic system which controls the outdoor noise intrusion only when it occurs. Such a control system can be linked to a programmable ventilation system that closes the windows and ventilators when a noisy vehicle or an aircraft approaches and opens them when the noisy event passes. The window could be closed automatically, at certain times of a day, week, month and a year when the noise level is higher than standard level.

Introduction

For the majority of people the only way to reduce aircraft and traffic noise is to close windows in busy hours. Opening and closing a window is the simplest way to supply the required fresh air and moderate the temperature inside a building. The window transmits the outdoor unwanted noise to the inside space and disturbs the comfort conditions. In most noisy situations there is a conflict between the need to supply fresh air and the need to control noise penetration from aircraft, trains and heavy vehicles. A programmable system to control ventilation would permit the occupants to specify the type of noises that they do not wish to hear and the degree of required ventilation. The window could be automatically closed at the approach of a noisy event and opened when the noise source had passed. Based on interest of the user, the window could be automatically closed at the certain times of a day, week, month or year. Such a system would have to discriminate between the sounds as wanted or unwanted and based on reasonable criteria make a decision about its state of opening. A major constraint is that the method must be fast enough to be used to activate the window before the noise level peaks. In this paper the concept of a noise activated control system to operate a window has been studied for two different classes of noise sources; single acoustic source and a mixture of noise sources.

Conventional methods of noise reduction inside buildings involve using two distinct methods, passive and active noise control. In passive noise control, sound absorbent materials are used to reduce the level of sound inside the building. To achieve high noise insulation the wall should not have any opening such as windows. Double glazing the windows and using acoustic insulating material are the most popular methods in passive noise control. In this situation, when the windows must be sealed and always closed, it is pointless to think about natural ventilation. Passive noise control is effective for high frequency noise. In reducing low frequency noise levels the bulk and cost of sound absorbent materials increases exponentially and makes these methods expensive to implement. In practice the windows are considered as important elements of a building, allowing for natural lighting, ventilation and visual contact with the outdoor environment [1].

Active noise control uses the superposition of acoustic waves to generate destructive interference. A stable destructive interference in active noise control is possible only for frequencies less than 500 Hz. A complete cancellation of unwanted noise by generating the secondary acoustical wave is dependent on the acoustical response of the medium and the response characteristics of the electronic equipment used in the system. It is also strongly related to the acoustical parameters of the primary source [2]. Active noise control is effective when the sound is steady or changing slowly, but outdoor noise is broadband and time-varying. Therefore, the active noise control is not a good solution for residential places. An active noise control system in a residential building can cancel the low frequencies of speech and music and the other low frequency sounds that people want to hear.

In this paper we try to clarify the concept of a noise activated control system to operate window and determine how useful and applicable it can be. Transportation noise emitted from heavy vehicles and aircraft are considered as the unwanted noises and the control system has to identify these sources. Several signal processing and acoustical pattern recognition methods have been examined to find a suitable criteria to discriminate transportation noise and ordinary environmental sounds such

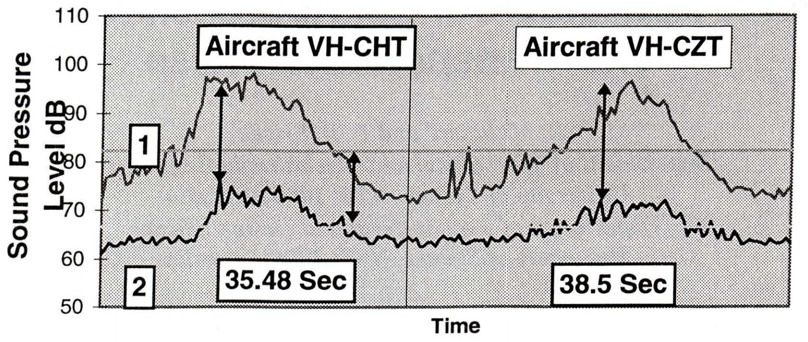

Fig. 1. : A time history of sound pressure level from two aircraft. Graph 1, when the window is open and graph-2 when the window is closed.

as speech and music [3]. The calculation of the minimum permitted time delay for a control system is presented. The proposed algorithm and the results are discussed and it is concluded that significant reductions in peak noise level could be obtained.

Linear prediction coefficients, autocorrelation function, zero-crossing, energy of signal and sound pressure level can be employed as features to be extracted to make an adaptive algorithm for pattern recognition. This algorithm is capable of training and operating the control system. As the result of the noise recognition process a decision can be made about the position of the window.

Noise attenuation

To realise how much a closed window can reduce the intrusion of transportation noise, an experiment has been arranged. The noise monitoring system was tested by being exposed to noise emitted from aircraft VH-CHT and VH-CZT and the sound pressure level for open and closed window was monitored from inside the building in vicinity of a window. The thickness of glazing is 5 mm.

Fig. 1 displays the time history of sound pressure level for an open and closed window in graph 1 and 2 respectively. To record the result of closed window a loud speaker has been used. Fig. 1 shows that closed window reduces the maximum sound pressure level from 98 dB to 77 dB for the aircraft type VH-CHT and from 96 dB to 73 dB for aircraft type VH-CZT. The average of sound pressure level during the flyover of these two aircraft is reduced from 82.26 dB to 66.63 dB. As glass attenuates the higher frequencies of noise more than its lower frequencies, the result of noise reduction in the case of some transportation noise which contents high frequency signals will be more satisfactory than that of aircraft noise.

Smart ventilation

One of the major problems in insulating buildings against outdoor noise such as transportation noise is natural ventilation. Air conditioning systems are expensive to install, run and maintain. The use of natural ventilation negates any attempt to add acoustic insulation to the windows. Smart ventilation as a part of an automated building would permit the occupants to specify the type of outdoor noises they wish to respond to. After a proper response of the system, the outdoor noise entering the building is attenuated more than 10 dB. Such a system to would identify the appropriate property of unwanted noise, based on the reasonable criteria and make a decision whether the window should be opened or closed. Speech intelligibility or sleep disturbance limits could be the boundaries of the operating system.

In the design of a smart ventilation system, thermal comfort, quality of air and the occasions that human being decides to open or close the window such as strong winds have to be considered. Such a ventilation system could also be programmed to hold the window closed at certain times, such as when the television is on or a telephone conversation is being held. Also, such a system could close the window in rush hours and when a known noise event such as garbage collection is taking place. Therefore, the acoustical signals of these sources and the thermal conditions of air are the inputs of the system. The output of the system is the position of the window. Other inputs such as thermal parameters of indoor and outdoor air, time and the residents voice commands could also be contemplated.

Transportation noise categories

Transportation noise emitted from aircraft and traffic flow is a non-stationary signal and the frequency spectrum is time dependent. The amplitude and frequency of road transportation noise are strongly dependent on the acceleration, speed, operating mode and the condition of car and road. In the case of aircraft noise, the type of aircraft and its propulsion system, load, take off or landing position, the angle of flight and the weather conditions can affect the characteristics of the emitted noise. A lot of random parameters such as the noise of vehicle brakes and sound of car horns and the vibration of trailers are mixed with transportation noise. In this paper we try to find applicable methods to identify approaching noise sources. These methods can be used to activate the control system sufficiently fast and accurate. As a first step traffic noise emitted from heavy vehicles and aircraft are considered as the unwanted noises. Transportation noise can be classified into two categories: "single noise event" and the "mixture of noise events". When a single vehicle comes into a quiet background, the time history of the signal can be considered as a "single noise event". Therefore, there is only one source of noise that can affect the measuring and control system. Aircraft noise and the noise emitted from the heavy vehicles at night time are examples of single noise events. Although the amplitude and frequency of signals change in different instants and positions of the vehicle, detection and recognition of single source in a quiet background is much easier than the mixture of noise. To have a better understanding of the difference between the single and mixture of acoustical events, we take the example of the crowd in a classroom. Each word is a regular set of acoustical signals which can be recognised and comprehended. But when different speakers start pronouncing different words, the combination of acoustical signals is difficult to comprehend. The only judgment about the crowd is the level of loudness and noisiness. Another important point in relation with the single noise event is that in the majority of pattern recognition situations the decision about the class of sample is made after the event has occurred. But in a real-time noise activated system, a decision needs to be taken at the very beginning of the event to avoid any disturbance.

In a noisy background, it is difficult to distinguish between different vehicles. In spite of having a lot of methods for reconstruction and detection of the known signal in a noisy background, detection of the time-variant noise of vehicle with different characteristics and conditions is very difficult, especially when the background noise is traffic noise which usually has the same nature as the noise event.

Minimum reaction time

There is a time-delay between the instant at which the approaching vehicle is detected and the final reaction of the window controller system. This time delay consists of two parts. The first part is related to the algorithm computational time and the second part is the result of the inherent delay of the actuators and the mechanical system.

The software system includes many stages such as: data acquisition, data calibration, indexing and bundling of data, logarithmic conversion, feature extraction, pattern classification and comparison between the various conditions and inputs to make the right decision about the position of the window. A stepper motor is the best programmable actuator to control the angular position of the window. Therefore, after decision making, a sequential coding is needed to operate the stepper motor.

The data acquisition part is set up based on the required sampling period and the maximum frequency to be measured. The time needed to perform the data acquisition and processing for half a second sampling time with the rate of 5 KHz is 0.7 seconds. The time requirement for sequential coding to indicate the angular position of the stepper motor's shaft is the major time consuming stage. The total time delay of control system with respect to the beginning of noise event can be formulated as follows:

$$t_{delay} = t_s + t_p + t_a + t_i \tag{1}$$

where, t_{delay} is the total time delay, t_s is the sampling time, t_p is the time requirement for data processing and pattern recognition, t_a is the time of sequential coding to operate the actuator and t_i is the inertial time delay because of the mechanical equipment.

In a successful noise control experiment the total time delay must be much shorter than half of the duration of the whole noise event. It is assumed that the peak of noise event occurs at the half time of noise event.

$$t_{delay} < T/2 \tag{2}$$

where T is the duration of the noise event. An experiment was arranged with a single noise event from a point source such as a heavy truck, with constant acoustical property with respect to time. The vehicle was driven with the maximum permitted speed of 60 Km/h, and the acceleration is assumed to be zero. The road is considered as a straight line.

As it is shown in Fig. 2, the observer is located at 5 m beside the road and the threshold of sound pressure level for detection is 70 dB. The maximum sound pressure level of heavy vehicle at the observation point is 95 dB. The whole detection time is T, and the length of the road which falls within the detectable area is X. The relation between sound intensities in observation point and detection point can be describe as equation (3). The sound attenuation through absorption by air is negligible. In the case of traffic noise, for 60 percent relative humidity, the sound absorption is less than 1 dB in 500 m.

$$\frac{I_0}{I_1} = \frac{r_1^2}{r_0^2} \tag{3}$$

By replacing the equivalent values from the logarithmic form of sound pressure level and sound intensity, we have:

$$SPL_0 = 20\log\frac{\overline{P_0}}{2\times10^{-5}} \tag{4}$$

$$SPL_1 = 20\log\frac{\overline{P_1}}{2\times10^{-5}} \tag{5}$$

$$\frac{I_0}{I_1} = \frac{\overline{\dfrac{P_0^2}{\rho . c}}}{\overline{\dfrac{P_1^2}{\rho . c}}} \tag{6}$$

$$r_1 = r_0 \times 10^{\frac{1}{20}(SPL_1 - SPL_0)} \tag{7}$$

$SPL_0 = 95(dB)$ and SPL_1 is decided to be $70(dB)$, hence, $r_1 = 88.9\ m$ and $v = 16.7\ (m/s)$,

$$x = r_1 \cos(\alpha) \tag{8}$$

$X = 177.8\ m$

$$X = v.T \tag{9}$$

which yields $T = 10.63$ (sec).

The above values show that a heavy vehicle with a sound pressure level of 95 dB located at the distance of 5m from the road is detectable only during 10.63 seconds of the event. In other word for 10.63 seconds in circular area with diameter 177.8 *m*, the sound pressure level is higher than or equal to 70 dB. In this calculation the Doppler effect is not considered. In a program which was developed in a virtual instrumentation software, the sampling time, $t_s = 0.5$ sec, the time required for data processing and pattern recognition $t_p = 0.2$ sec, the time of sequential coding to operate the actuator $t_a = 2$ sec and the inertial time delay $t_i = 0.6$ sec because of small size of prototype. Therefore the total delay is $t_{delay} = 3.3$ sec. When the noise of heavy vehicle reaches its peaks after 5.32 sec, the window is closed.

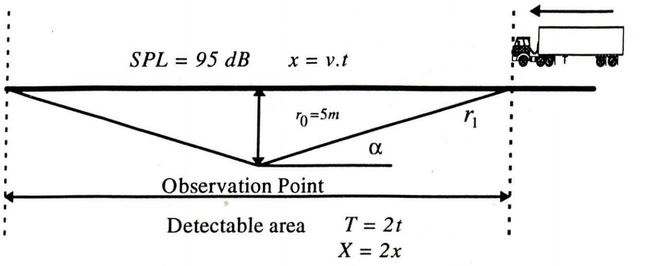

Fig. 2: The schematic diagram of the experiment for noise measurement.

Noise classification

To operate the window, sound pressure level and the energy of signal are good criteria. Therefore, as soon as the sound pressure level and the energy reach to certain levels, the control system will be activated. But these criteria are not dependent on a particular source of noise. The control system may be activated by any other acoustical source. The ventilation system should be activated only by transportation noise with a sound pressure level and energy of more than certain levels. In this study aircraft noise, heavy vehicle and mixture of traffic in dry and rainy weather are considered as transportation noise sources. A major system constraint is the time limit imposed on the recognition task. It would not be a practicable system if the response to the noise event happened after or at the end of the event. Therefore, the acoustical features have to be selected that are extractable in a short-time during data acquisition process. Identifying the approaching unwanted source of noise at far distances is crucial. The first step in the design of a control system sensitive to transportation noise is to recognise the type of noise. The real-time control of a controllable window requires real-time data acquisition, pattern recognition and decision making. The pattern recognition problem can be divided into several stages of which feature extraction and source classification are the most two important ones. Since there is no unique method to distinguish between different noise sources, a variety of parameters with different signal processing methods have been used. A discriminating parameter has to be used to distinguish between the random broadband transportation noise and complex spectra speech, music and other sources of environmental noise as such. This is the main purpose of acoustical feature extraction in the present work. The extracted features have to give an appropriate measure of the observed data while retaining most of information that is useful for recognition. The major method used in this work is based on statistical feature extraction. The proper statistical function to treat the acoustical data has to be correlated to the inherent structure of data observed from the noise source.

The low order linear prediction model of transportation noise and the autocorrelation function are useful to differentiate this type of noise from speech and music. The zero-crossing rate of transportation noise signal is another applicable criterion. The processes of detection and recognition of a particular source of traffic noise have a lot in common with what is done in diagnosing defects in machines, speech and speaker recognition and passive sonar tracking. All the sample patterns should be compared with a reference template pattern. Selecting the most effective reference should be based upon an appropriate statistical method. The maximum likelihood ratio is a useful discriminating scheme. In order to do

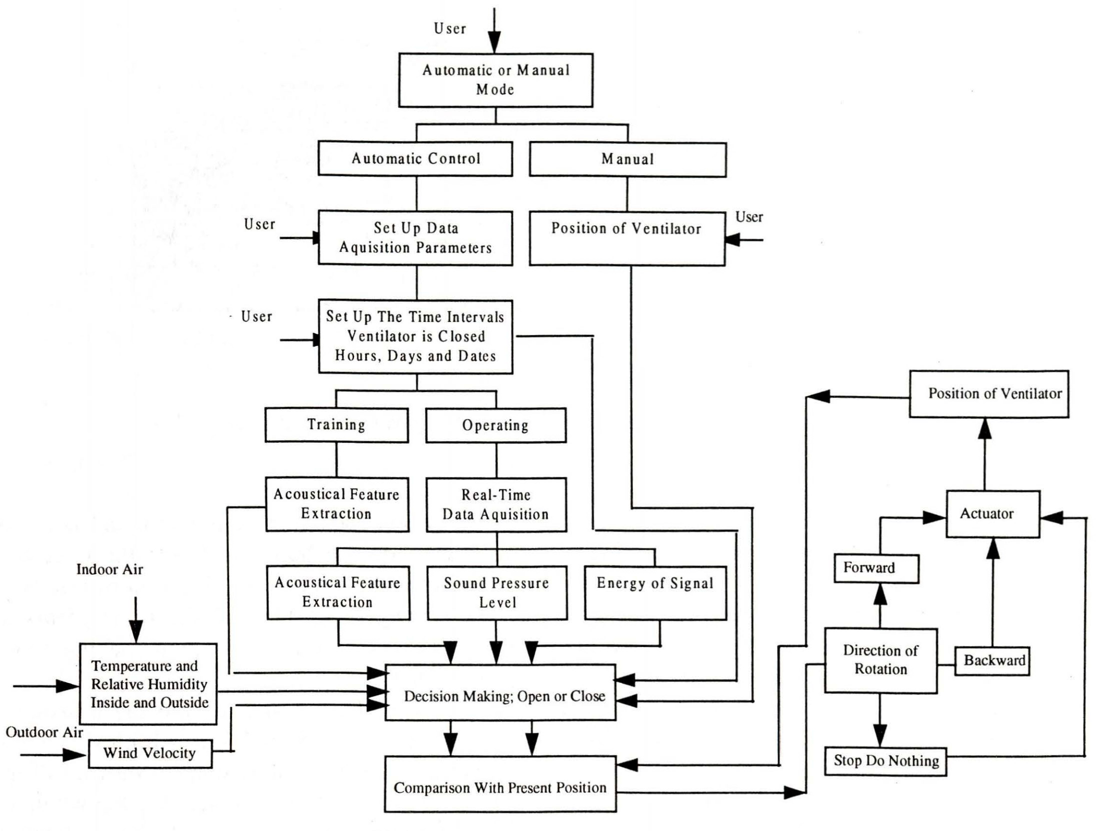

Fig. 3: The block diagram of a noise activated ventilation system.

that a decision boundary should be defined as a criterion to compare a sample with the reference pattern.

Noise activated ventilation control

A significant characteristic of a noise activated control for ventilation systems is to operate at the start of the event rather than at its completion. In spite of the simplicity in the operation of a ventilation system the acoustical detection and pattern recognition programs of such events can be as complex as that of a guided torpedo detector.

Fig. 3 shows the block diagram of a control algorithm to operate the ventilation system. It works in two modes of "Manual" and "Automatic" operations. The time and date of closing the window can be set up by the user. The criteria to activate the system are divided into two groups: primary and secondary criteria. The primary criteria such as sound pressure level and the energy of signal are more important. The secondary criteria are the linear prediction coefficients, autocorrelation function, plot of first and second frequencies and zero-crossing. When the system passes the boundaries of primary criteria, the decision system will start to calculate the secondary criteria. The control system would be activated when the sound pressure level reached to 70 dB and the energy of signal for 512 samples with the rate of 5 KHz exceeded 10 units of energy. If the result of recognition favours "transportation noise", the system sends a window shut command. On the other hand if the recognition favours other environmental sources such as speech and music an open window signal will be issued. The "Automatic" mode of this program is implemented in two stages, "Training" mode and "Operating" mode. In the training mode the system is exposed to a set of training transportation noise signals in order to set the decision boundaries. In operating mode, during a real-time data acquisition process, feature extraction and classification take place.

Based on the temporal regularities that exist in some environmental phenomena such as traffic flow, the system can be predicably programmed with respect to time. These occasions can be some particular times of year, week or day. The user can set up the date and time of closing period in training mode.

The comparison between indoor and outdoor air conditions interferes the decision system. In some occasions thermal comfort priority has to be considered.

Simulation

The interpretation of noise and other events that happen during the operation of noise activated window is shown in fig. 4. This figure is the time history of sound pressure level of aircraft vh-ewi flyover in landing mode. The first graph is the noise monitoring when the window is

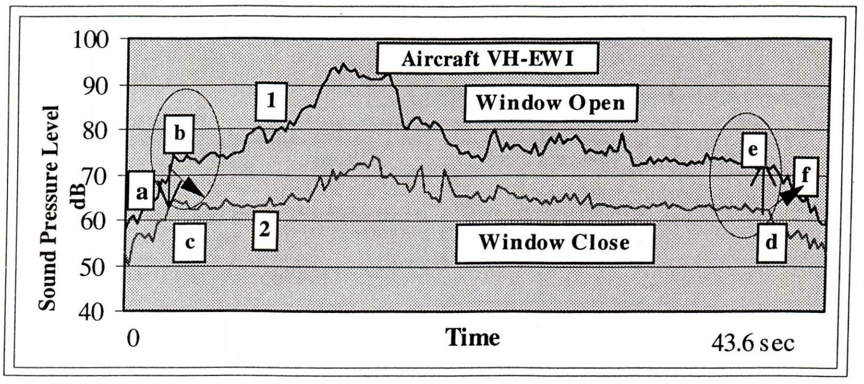

Fig. 4: The predicted result of monitoring the aircraft noise during closing the window.

open. Graph-2 is the result of monitoring the same noise with the use of loud speaker when the window is closed.

The actual received noise during the window operation is the path a-b-c-d-e-f. We have assumed that the user did not wish to receive transportation noise higher than 70 db. At the beginning of the aircraft noise event, the recognition system identifies an approaching transportation noise. The start of noise event is at point (a) in graph-1. The sound pressure level increases from point (a) to point (b) where it exceeds the sound pressure level threshold. At point (b) the control system starts to close the window. If this process is supposed to happen immediately then the noise level will jump from point (b) to (c). Otherwise there is a delay which prolongs the system reaction. When the outside noise is reduced back to 70 db the control system will start to open the window. In fact when the outside sound pressure level goes to point (e) the noise level will jump from point (d) to point (e). As a result the sound pressure level is reduced from 94. 85 db to 74.37 db at the peak with a time average reduction from 78.07 db to 65.2 db. There are two transient areas in the time history of sound pressure level in fig. 4, which are shown inside ellipses. These two transient states are important as mentioned in section 5.

Graph-1 in Fig. 5 shows the noise level when the window is open and Graph-2 is the actual noise signal received inside the building during the operation of the noise activated system. The location of points a, b, c, d, e and f are comparable with the same points in Fig. 5. Point (c) is the instant at which the window is completely shut and point (e) is the instant the window is open. The paths b-c and d-e are the transient parts of noise reduction in which window is closing and opening. The transient part of noise control shown in Fig. 5 takes about 6.5 (sec) out of 25.5 (sec) for the whole event. As a result, there is a considerable noise attenuation for almost 75% of noise event duration. Also the noise peak would not be as annoying as it could possibly be. The distance between two lines passing points b and c indicates the time delay of the actuator, which is equal to $t_a + t_i$.

Conclusion

This paper explained a possible solution to overcome the existing conflict between natural ventilation and passive noise control. A noise activated control system can protect the indoor space against the intrusion of the outdoor unwanted noise if it detects and operates fast enough. In the case of single noise event in a quiet background such as aircraft noise event, this method is more efficient.

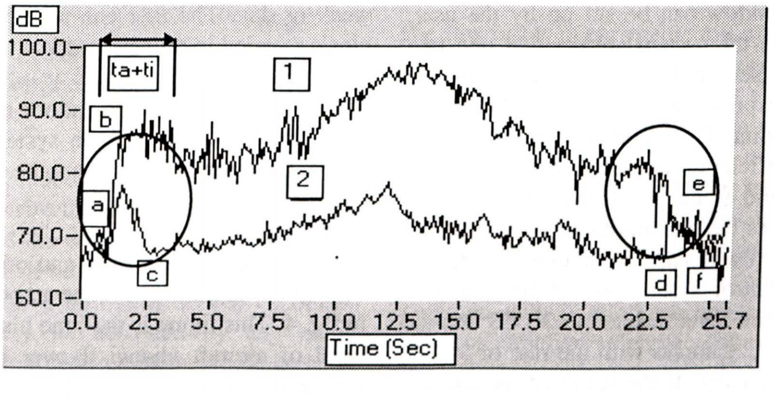

Fig. 5: The result of operating a noise activated system to control the window position, the noise of aircraft VH-TOO.

The major problem to use the noise activated system is the accuracy in recognition. The results of this research can be used in other fields of acoustical source recognition system. In spite of the simplicity in the operating of a ventilation system the acoustical detection and pattern recognition algorithms can be very complicated.

Finding a criteria to judge about the priority of thermal comfort or silence is helpful. Making a prototype from a noise activated ventilation at this stage is necessary for further studies with consideration of public demands.

Acknowledgment

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The Attenuation of Noise Entering Buildings Using Quarter-Wave Resonators

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Abstract

This paper presents an alternative way of achieving satisfactory noise levels in buildings located in noisy environments. It is a continuation of preliminary research carried out into the use of quarter-wave resonators located at a building ventilation opening to achieve a reduction of noise entering a building. The level and nature of the attenuation of noise entering a building by a multiple resonator is determined, indicating that in addition to resonator action, there are diffraction effects similar in nature to that of a barrier which aid in the attenuation of noise entering the building. Future work will involve the use of "diffractors" located outside a window in a building to attenuate noise entering the building.

Introduction

The need for a cost-effective technique of reducing noise entering buildings through ventilation openings such as windows is becoming increasingly urgent as more people are becoming exposed to environmental noise. The use of quarter-wave resonators outside ventilation openings of buildings is a new application of a simple, well known noise reduction technique. Preliminary research carried out into the possibilities of this new application (Field 1995) involved sound intensity measurements of noise entering a building with a resonator located under a window. A comparison of intensity measurements with and without the resonator present indicated a substantial reduction in sound intensity in a narrow frequency range close to the resonant frequency of the resonator. The present research continues this work with the determination of the attenuation provided by a multiple resonator located outside a building ventilation opening.

Experiments

The Multiple Resonator

The resonator used in previous research (Field 1995) was converted to a multiple resonator by the insertion of two partitions, allowing for a three resonator system. The new multiple resonator was primitive in design, with no calculations carried out to check the suitability of its geometry except that the length of the respective cavities corresponded to a quarter wavelength of the noise to be attenuated. These lengths were achieved by filling the respective cavities with sand up to the required height corresponding to a quarter wavelength. The main aim was to observe any attenuation provided by the multiple resonator which would warrant further detailing of another resonator device.

Initial Experiments

The room used in all experiments was a 6:1 scale model based on a room used for determining the shielding provided by building facades (Lawrence et al 1983). In one wall there was a rectangular opening corresponding to 1% of that particular wall area with a ratio of opening width to opening height of 0.85. The volume of the room was 0.14m^3. The resonator frequencies chosen for the first experiment were 1250 Hz, 3150 Hz and 6300 Hz which would translate into relatively low frequencies for the full scale room. The nature of noise used in all experiments was white noise produced by a Bruel and Kjaer Type 1405 noise generator in the range of 20 Hz to 20k Hz. The source consisted of a horn driver with a tube attached to simulate a point source as closely as possible. The receiver was a 1/2 inch Bruel and Kjaer Type 4133 microphone. A smooth, rigid and reflective ground surface was used between the source and receiver in all experiments.

Initial experiments involved determining the effect of the multiple resonator on noise entering the building through the opening. In each experiment the sound pressure level was measured just inside the model room at the opening in 1/3 octave bands from 1k Hz to 10k Hz (using a Bruel and Kjaer Type 2034 Dual Spectrum Analyser and a computer program used to calculate 1/3 octaves) when the resonator was not present outside the opening and then when the resonator was present. Measurements were not made inside the room because of reflections from the inside walls which would make the effect of the resonator more difficult to assess. The source was located 1040 mm from the receiver at a height of 200 mm. The receiver was located just inside the opening in the model room at a height of 180 mm. This source-receiver geometry was kept constant. The difference between the two levels gave the attenuation, in dB, due to the presence of the resonator. The action of the

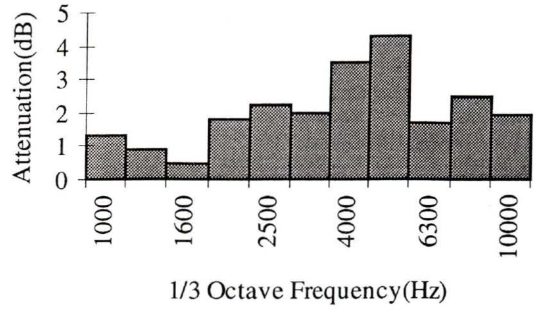

Figure 1(a) Attenuation vs 1/3 Octave Frequency (normal multiple resonator)

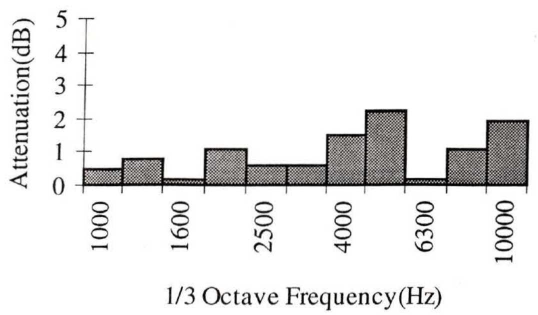

Figure 1(b) Attenuation vs 1/3 Octave Frequency (resonator full of sand)

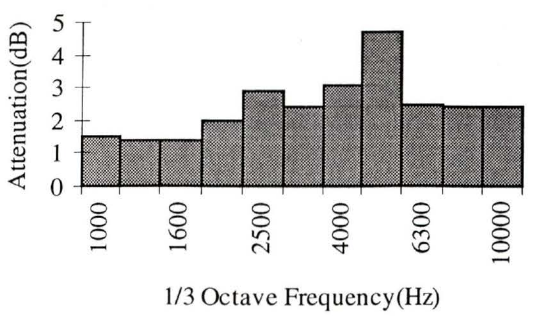

Figure 1(c) Attenuation vs 1/3 Octave Frequency (damped resonator)

resonator was observed under different conditions. Firstly the resonator was used in its normal state outside the opening with sand used to achieve the required cavity lengths for the resonators. The resonator was then filled completely with sand to simulate a thick barrier (and no resonator) and finally the sand was poured out of the resonator and filled with absorptive material so it became a type of multiple barrier with damping in between the partitions. The idea behind using these different conditions was to observe the nature of the resonator action. The two possible mechanisms of noise reduction by the resonator outside the building would be the intended action of the resonators resulting in attenuation at the particular resonant frequencies and also diffraction over a wide range of frequencies due to the presence of an obstacle near the building opening. The different conditions under which the resonator was tested would indicate the relevant mechanism. The results for the attenuation provided by the resonator under various conditions are shown in Figures 1(a) to 1(c).

Figure 1(a) indicates that the multiple resonator is not functioning as expected since the attenuation at the resonant frequencies of the resonator (1250 Hz, 3150 Hz and 6300 Hz) is not significantly higher than in other 1/3 octave bands. A comparison between Figure 1(a) and Figure 1(b) shows that diffraction is the only mechanism of attenuation provided by the multiple resonator. Diffraction is occurring at the tops of the partitions in the resonator and at the leading edge of the resonator exposed to noise (acting as a finite width multiple barrier). When the resonator is filled with sand, the partitions are no longer exposed. The attenuation achieved in Figure 1(b) is then only due to the outside edges of the multiple resonator casing. Figure 1(c) confirms that the resonator is not functioning properly. The addition of absorptive material should damp the resonator response but in some frequency bands there are increases in attenuation. A comparison of Figure 1(a) and Figure 1(c) shows that the undamped resonator gives similar results to the damped resonator. This indicates that the results are most likely due to diffraction at the tops of the partitions and outer edge of the multiple resonator.

These initial experiments indicated that there were two main problems with the multiple resonator. Firstly, after testing the resonator with normally incident white noise as opposed to grazing incidence in the model room experiments, it was found that the sand used to achieve the correct height for the quarter wave resonators was damping the resonator response significantly. It was also found that the cross-sectional areas of the resonators were too large compared to the quarter wavelength of sound they were meant to be attenuating. These two factors were inhibiting the formation of standing waves along the length of the resonator.

Experiments with a New Multiple Resonator

A new resonator was designed which satisfied the necessary conditions for good resonator response. This required the use of a rigid end in the resonator to facilitate the formation of standing waves and imposing constraints on the resonator geometry to prevent the formation of cross-modes while sustaining a high Q factor. The constraints were satisfied by solving the following optimisation problem for resonator diameter D (in metres) for a required Q factor (f is the required resonator frequency in Hz):

$$\pi f D << 1 \qquad \text{(Beranek and Ver 1992)}$$

$$Q < 1.455 \times 10^{-2} D\sqrt{f} \qquad \text{(Ingard 1994)}$$

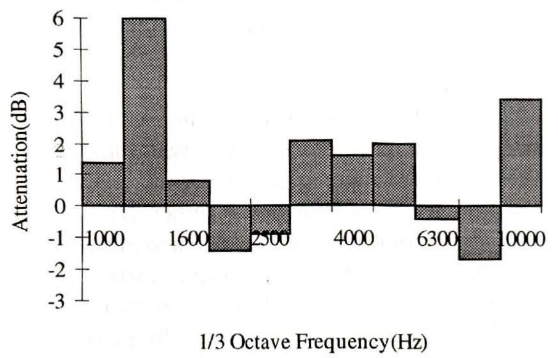

Figure 2(a): Attenuation vs 1/3 Octave Frequency(normal undamped resonator)

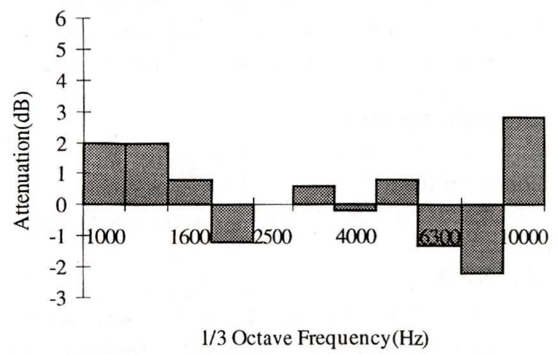

Figure 2(b): Attenuation vs 1/3 Octave Frequency (sand filled resonator)

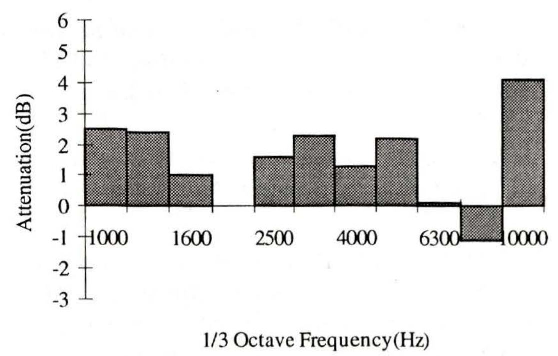

Figure 2(c): Attenuation vs 1/3 Octave Frequency(damped resonator)

The resulting diameter was then converted to an equivalent square cross-section. The new multiple resonator consisted of a 1250 Hz resonator and two 3150 Hz resonators.

The new multiple resonator was then used in a series of experiments similar to the initial experiments. For eight different source positions, the attenuation due to the insertion of the resonator under the window of the model room was measured in 1/3 octave bands at the window of the model room. The multiple resonator was then filled with sand and the experiment repeated. Finally the experiment was repeated with absorptive material added to the normal resonator in order to damp its response. A comparison of the three sets of experiments would pinpoint the mechanism of attenuation by the resonator, whether it be diffraction from the multiple resonator acting as a multiple barrier, the intended action of the resonator itself, or a combination of the two effects.

Typical results from the three sets of experiments are shown in Figures 2(a) to 2(c). The particular source-receiver geometry for these results included a source height of 185 mm, a receiver height of 180 mm, the resonator height of 175 mm (the top of the resonator was in line with the bottom edge of the opening in the model room, hence there was a line of sight between source and receiver) and a source-receiver distance of 850 mm.

From Figures 2(a) to 2(c) it can be concluded that both resonator action and a diffraction effect are present.

The response of the 1250 Hz resonator is clear by a comparison of the attenuation levels in the 1250 Hz 1/3 octave bands in Figures 2(a) and 2(c). The undamped resonator achieved 6.0 dB attenuation while the damped resonator achieved 2.4 dB attenuation. The response of the 3150 Hz resonator was less impressive. The undamped resonator achieved 2.3 dB attenuation while the damped resonator achieved 2.1 dB attenuation. This can be explained by considering the total energy in the respective resonators. The total energy in each resonator is proportional to the volume of each resonator (Ingard 1994). The 3150 Hz resonator, requiring a shorter length and smaller cross-section than the 1250 Hz resonator, therefore has a smaller total energy than the 1250 Hz resonator. It would therefore require several 3150 Hz resonators to achieve the same magnitude of attenuation as the 1250 Hz resonator.

The diffraction effect is evident by the similar trend in attenuation across the 1/3 octave bands not including the resonator frequencies for both the undamped and damped resonators. Since the experiments were carried out with a rigid, reflective ground surface between the source and receiver, the attenuation pattern will be influenced by the interaction between the direct and ground reflected rays arriving at the receiver. At specific path differences between the direct distance from source to receiver and the distance from the source to receiver including a ground reflection, there will be constructive and destructive interference between the direct and reflected rays. For propagation over reflective ground, the phase difference corresponding to this path difference can be given as:

$$\Delta\phi = \frac{2\pi}{\lambda}(L_r - L_d) \qquad (1)$$

where λ is the wavelength of sound, L_r is the reflected distance between the source and receiver and L_d is the direct distance between the source and receiver. Equation (1) can be solved to find frequencies at which destructive and constructive interference occur.

For the geometry corresponding to the results of Figures 2(a) to 2(c) destructive interference is expected to occur at 2300 Hz (π radians out of phase) and 6900 Hz (3π radians out of phase). Constructive interference is expected at 4600 Hz (2π radians out of phase) and 9200 Hz (4π radians out of phase). The presence of the resonator, however, disturbs this interference pattern. This is similar to the insertion of a barrier between a source and receiver. The barrier can disturb the interference pattern formed between direct and ground reflected sound without the barrier present, resulting in increases in sound level at some frequencies instead of reducing it.

The effects of diffraction can be seen by a comparison of the attenuation levels in the 1/3 octave bands not including the resonator frequencies of Figures 2(a) to 2(c). Similar to the results obtained in the initial experiments (Figures 1(a) to 1(c)), diffraction occurs at the partitions within the resonator casing and at the outer casing of the resonator itself. Figures 2(a) and 2(c) indicate diffraction at both the partitions and casing while Figure 2(b) shows the diffraction of the casing alone (since the resonator cavities were filled with sand). It should also be noted that the attenuation due to diffraction for the damped resonator (Figure 2(c)) is higher in most 1/3 octave bands than that obtained by the undamped resonator (Figure 2(a)). From these results the resonator with damping material in it could possibly be perceived as a multiple barrier with an absorbent coating on the barrier surfaces. Maekawa(1965) found higher values of attenuation provided by barriers with a sound-absorbing covering on the source side and Fyfe et al (1995) found an increased performance by absorptive barriers above that of a standard barrier which was a function of the source to barrier distance. In order to test this possibility, the damping material was removed from the resonator cavities. A thin layer of absorbent material was then added to the front casing of the multiple resonator. The attenuation experiment was then repeated with the same source-receiver geometry. It was found that the resonator with the absorptive lining attached achieved higher attenuation levels (between 0.5 dB and 1 dB depending on frequency) in all 1/3 octave bands not including the resonator frequencies with a similar trend to that of the resonator filled with damping material. Hence the multiple resonator filled with damping material can be thought of as a type of multiple barrier with lined with absorbent material.

Currently work is being carried out to develop a computer program that predicts the diffractional effects of the resonator based on theory developed for barriers including a reflective ground surface. The difference in path length of the direct and various ground reflected sound rays plays an important role in this program.

Conclusions

It has been found that a model quarter-wave resonator can achieve 6.0 dB attenuation of white noise in the 1/3 octave band containing the resonator frequency while providing further attenuation in other 1/3 octave bands due to diffraction at the outer edge and at partition edges within the resonator. For a multiple resonator system, it has been found that several higher frequency resonators would be required for each lower frequency resonator according to their volume ratios in order to achieve the same level of attenuation. Future work will include different source locations, interactions of the resonators, developing theory for the prediction of attenuation in complex situations such as near building facades and the use of "diffractors" located around the window of a room to achieve attenuation of noise entering the room.

Acknowledgment

This work has been supported partly by a research scholarship from the New South Wales Roads and Traffic Authority and partly by an Australian Postgraduate Award.

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New Sophisticated Absorption Materials. DECI-TEX® Acoustic Textiles

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Abstract

A new range of sophisticated sound absorption materials, known as Deci-Tex®, have been successfully developed. These engineered fabrics absorb sound effectively, efficiently and economically. Without the bulk normally associated with traditional sound absorbing materials, Deci-Tex acoustic textiles are installed with an air gap between them and the panel behind or even as freely suspended banners. The depth of the air space matched with the impedance of the Deci-Tex results in sound absorption equivalent to the same thickness of many traditional acoustic materials. The absorption mechanism is proposed. Experimental test results are displayed, as well as the results obtained in two previously noisy commercial applications.

Introduction

Traditional sound absorption materials include polyurethane foams and fibrous materials, such as fibreglass, polyester and so on. These traditional materials absorb and dissipate sound energy through friction in the open cell structure of foams, or between the fibres of the porous fibrous materials. Generally, as the thickness of these traditional materials increase, the sound absorption also increases[1]. However, a new range of sound absorption materials have been successfully developed at I.N.C. Corporation. These new acoustic textiles have found to be a practical alternative to foam and fibre sound absorption materials, eliminating the bulk normally associated with traditional materials.

Background

The project aim was to develop high flow resistivity membranes that were thin, inexpensive, good sound absorbers with aethestic appeal to be used in internal sound absorption applications. This resulted in engineered, fabric based sound absorbing materials, now known as Deci-Tex acoustic textiles. As the sound passes through the textile, it is forced through a series of tiny pores which provide resistance to the flow of air. The flow resistance of a porous material is the ratio of the air pressure differential divided by the normal air velocity at the surface of the the sample[2]. The flow resistance of Deci-Tex has been acoustically optimised to ensure maximum sound absorption.

To achieve their remarkable performance, Deci-Tex acoustic textiles are installed with an air gap behind them and the wall as shown in Figure 1. The depth of the air space matched with the impedance of the Deci-Tex results in sound absorption comparable to the same thickness of some conventional acoustic materials. Recent experiment results also show that Deci-Tex P16 performs when freely suspended as banners. In this study, the performance of two main types of Deci-Tex were evaluated by laboratory testing. Results are also shown for the commercial applications of Deci-Tex P16.

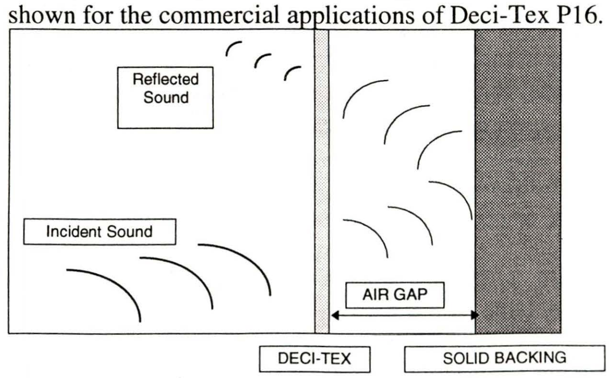

Figure 1: The sound wave passes through Deci-Tex and is absorbed.

Results

Deci-Tex P16

Deci-Tex P16 is a thin non-woven acoustic textile, designed for use as a ceiling or wall lining. It can be used as an acoustic finish in it's own right. The sound absorption of Deci-Tex P16 at various cavity depths is shown in Figure 2. These results show that as the cavity gap increases, the maximum sound absorption peaks at lower frequencies.

In addition to the advantages mentioned above, Deci-Tex P16 can be laminated to a wide range of decorative fabrics. The acoustic performance is guaranteed by the acoustic textile, whilst the facing fabric can be almost any pattern or style. This provides huge freedom to interior acoustic design, which can be easily updated or changed.

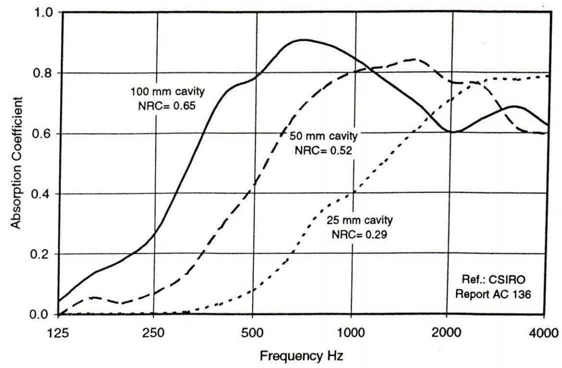

Figure 2. Sound absorption for Deci-Tex P16 at various cavity depths.

Adding a thin layer of non-woven polyester in-fill provides damping of the cavity resonance, significantly increasing the sound absorption, as shown in Figure 3.

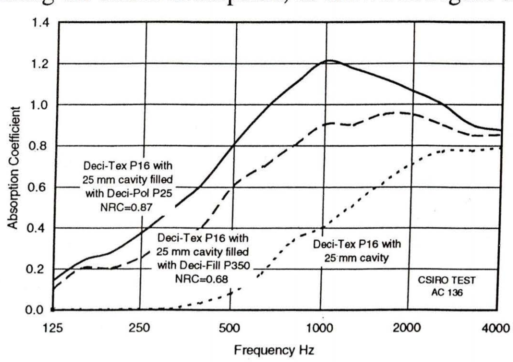

Figure 3. Comparison of Deci-Tex P16 and a 25mm cavity with various backings.

Earlier applications utilised the sound absorbing properties of Deci-Tex P16 stretched a certain distance from a reflecting plane. From an architectural point of view it is not always convenient nor as creative to install the textile fabric next to the wall or ceiling. The most recent performance tests conducted at RMIT Acoustic Laboratory displayed very exciting results. Deci-Tex P16 was tested as freely suspended, vertical banners. Results showed that banners separated 1m, 1.5m and 3m retained the sound absorption properties per square metre of acoustic textile. The acoustic performance of 3 freely suspended banners (1.8m x 2m, separated 1.5m from each other), proved to be similar to that of Deci-Tex P16 installed with a 50mm air cavity from a reflecting plane, shown in Figure 4. There is an encouraging performance improvement of the banners in the low frequency range and in the high frequency range resulting in a slightly higher NRC index.

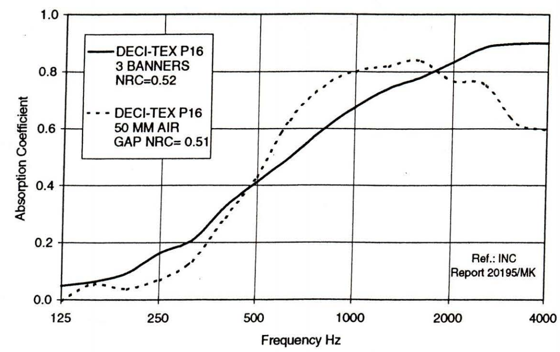

Figure 4. Comparison of 3 Deci-Tex P16 Banners and Deci-Tex P16 with 50 mm air cavity

Deci-Tex P52

Deci-Tex P52 is a thin, rigid non-woven acoustic textile designed for use in suspended ceilings, behind perforated panels. Deci-Tex P52 is easy to install, non-toxic and contains no synthetic mineral fibres. The sound absorption for Deci-Tex P52 behind a thin perforated metal panel of 30% open area at various air gaps, is shown in Figure 5. The results again show that as the air gap increases the maximum sound absorption peaks at lower frequencies.

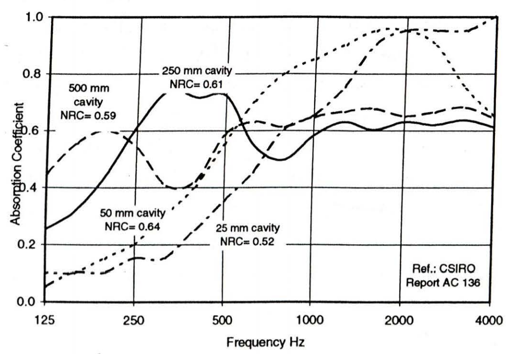

Figure 5. Sound Absorption for Deci-Tex P52 installed behind perforated metal panels tested at air gaps of 25, 50, 250 and 500mm.

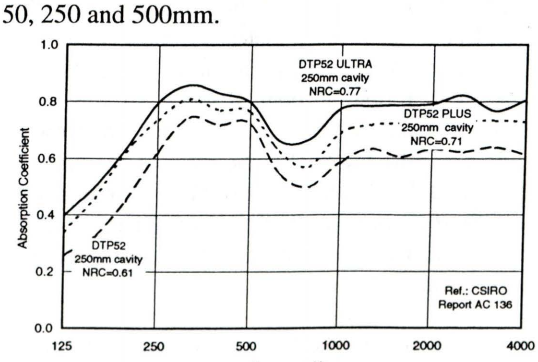

Figure 6. Sound Absorption of Deci-Tex P52 with diferent polyester backings at 250mm air gap, installed behind a perforated metal panel.

As with Deci-Tex P16, the acoustic performance can be increased further by the addition of a thin layer of non-woven polyester in-fill within the air cavity (refer to Figure 6).

Case studies

(1) Application of Deci-Tex P16 for an indoor swimming pool.

The pool owner's business of teaching young children how to swim, was at risk. After numerous complaints about excessive noise from the neighbours, the owner of the private indoor swimming pool in the Melbourne suberb of Bentleigh, desperately required an economical solution to her noise problem.

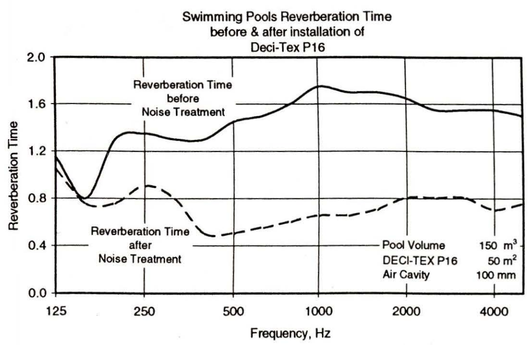

Figure 7.

The volume of the pool enclosure measured $300m^3$. $50m^2$ of Deci-Tex P16 was installed on the ceiling with a 100mm air gap between the acoustic textile and the hard ceiling surface behind. The reverberation time was measured before and after the installation of Deci-Tex P16 and the results are shown in Figure 7.

From this the sound absorption coeffiecients at various frequencies were calculated and the graph is presented in Figure 8. The performance of Deci-Tex P16 exceded expectations, reducing the external noise level by 5 - 6 dB(A), which stopped the neighbours complaining.

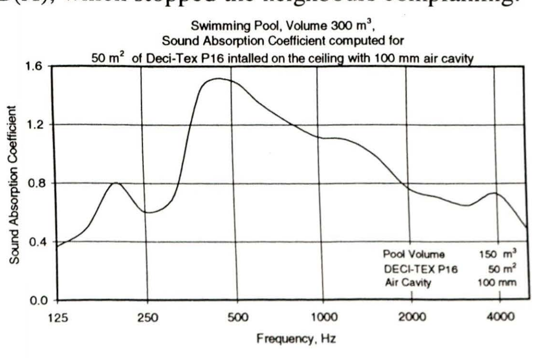

Figure 8.

(2) Application of Deci-Tex P16 in a restuarant.

The recently opened Via Mare Restaurant in Frankston, Victoria, was stylish but somewhat noisy. The L-shaped restaurant has large windows, tiled floors, plaster board walls and other such sound reflecting surfaces. The owner contacted an I.N.C. distributer for a practical and attractive solution.

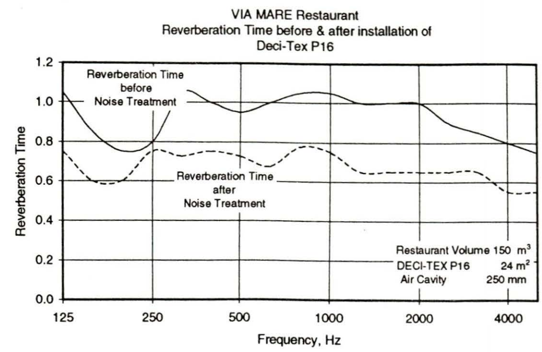

Figure 9.

$24m^2$ of Deci-Tex P16 laminated to an olive coloured fabric chosen by the owner, was subsequently installed with a 250mm air cavity. The area of the acoustic textile was limited due to the location of the down lights in the ceiling. The results were pleasing to the owner, although not as outstanding as those obtained for the swimming pool installation. Figure 9 shows the difference in reverberation time before and after the Deci-Tex P16 installation, while Figure 10 reveals the calculated absorption coefficient based on the reverberation results. The acoustic textile has effectively reduced the reverberation time especially around 500 - 2000 Hz, which is in the vicinity of the human speech range.

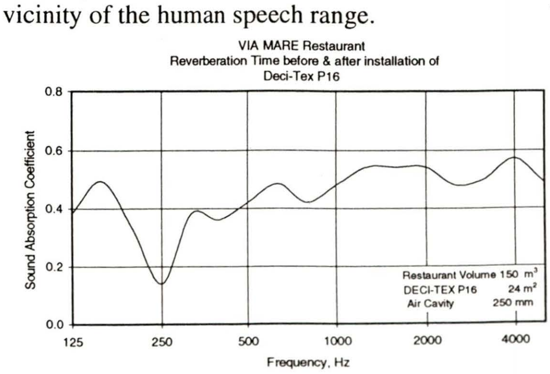

Figure 10.

Conclusion

Deci-Tex acoustic textiles P16 and P52 are practical alternatives to traditional sound absorption materials offering high sound absorption, without the bulk. The Deci-Tex range of materials are safe, easy to handle, decorative and suitable for use in all public buildings under the Building Code of Australia. Even though the remarkable performance of the sophisiticated acoustic textile has not been well understood, Deci-Tex represents the latest innovation in room acoustics offering maximum results with the minimum of cost and fuss.

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Session 6A

Sound Propagation

Outdoor Sound Propagation - An Overview

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Introduction

The quantification of sound propagation outdoors may range from relatively simple to very complex, depending upon the nature of the source and distribution of the affected surrounding areas. If the source is composed of many individual component sources, as would often be the case with an industrial plant, and the surrounding area is extensively affected then the use of a computer to carry out the analysis associated with noise level prediction is essential and a number of schemes (Tonin, 1985) have been designed for the purpose. These schemes generally rely heavily upon empirical information determined from field surveys but gradually empiricism is being replaced with well established analysis based upon extensive research. At present the most successful schemes rely upon a mixture of both theory and empiricism as exemplified by the procedures outlined in the draft International Standard, ISO/DIS 9613-2. The work outlined here presents a more rigorous approach to the problem of predicting outdoor sound propagation than presented in the standard and is considered suitable for inclusion in a computer model.

Outdoor noise level prediction procedure

In all cases the method of predicting community noise levels proceeds as follows:

1. Determine power levels L_w of all sources.
2. For a given environment calculate the individual components of excess attenuation A_{Ei} for all sources i = 1 to N.
3. Compute the resulting sound pressure levels at selected points in the environment for each of the individual sources.
4. Compute the predicted sound pressure levels produced by all of the individual sources at the selected points in the environment by converting the sound pressure level due to each source to pressure squared, adding the squared pressure contributions and converting the total to sound pressure level.

The question of accuracy of prediction is always of importance and experience suggests that ±5 dB may be expected (Marsh, 1976). Alternatively an accuracy of ±2 dB has been claimed but in the latter cases measured data were used to refine the prediction scheme (Marsh, 1982; Delany, 1972). The major problem which makes accurate prediction of total sound pressure levels at community locations difficult is the variability of weather conditions. In practice, the sound pressure level corresponding to the least favourable weather conditions is often calculated and cited as a worst case. In addition, the average sound pressure level L_{Aeq} at various hours during the night and day is calculated assuming the most commonly occurring weather conditions for those times. Overall night-time and daytime L_{Aeq}'s are then calculated on a squared pressure (or energy) average basis. In many cases L_{10} and L_{90} are also calculated. Fortunately it can be shown that if the error in sound level prediction for any source is to be expected, the error for 100 such sources combined randomly will be no larger (Tonin, 1985).

The expression relating sound pressure level, L_p, and sound power level, L_w, for a single source is

$$L_p = L_W - K + DI_M - A_E \tag{1}$$

where K is the geometrical spreading factor, DI_M is a directivity factor to account for the directional radiation properties of the source and A_E is the excess attenuation due to all other effects. While it is generally recognised that the various components of attenuation may be interrelated and not simply additive, investigations have not proceeded as yet to the extent that it is possible to quantitatively express all of the possible inter-relations in one encompassing algorithm. Rather, attenuation of sound propagating out-of-doors is approximated as a linear sum of the component effects embodied in A_E when carrying out the computations of step 3 above.

In general there will be many sources contributing to the sound field. Incoherent sound addition is usually assumed, in which case the sound pressure level at the observation point in the field due to N sources may be computed where L_{pi} is the sound pressure level due to the i^{th} source as the sum of contributions as follows:

$$L_p = 10\log_{10}\sum_{i=1}^{N} 10^{10L_{pi}/10} \tag{2}$$

Geometrical spreading factor, *K*

For a point source in free space, the spreading factor K is given by

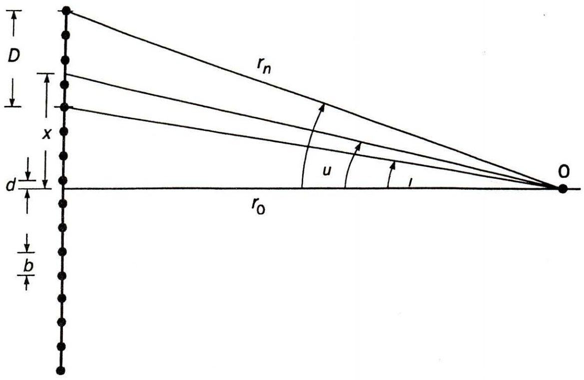

Figure 1: Arrangement for a line source, showing the angle subtended by the source at the observer.

$$K = 10\log_{10}4\pi + 20\log_{10}r \tag{3}$$

where r is the distance from the source to the point of observation.

For a line source, the spreading factor K is given by,

$$K = 10\log_{10}(4\pi rD/\alpha) \tag{4}$$

In the above equation D is the length of the line source, r is the distance from the source centre to the point of observation and α (= α_u - α_l) is the angle subtended by the source at the point of observation as shown in Figure 1.

For a wall modelled as a plane incoherent radiator, K has the following form.

$$K = 10\log_{10}2\pi + 20\log_{10}r - 10\log_{10}F(\alpha,\beta,\gamma,\delta) \tag{5}$$

The dimensionless variables of Equation (5) are; α = H/L, $\beta = h/L$, $\gamma = r/L$ and $\delta = d/L$ (see Figure 2). In the case of the near field, where (γ/10) < α,β,δ, the function F takes the following form.

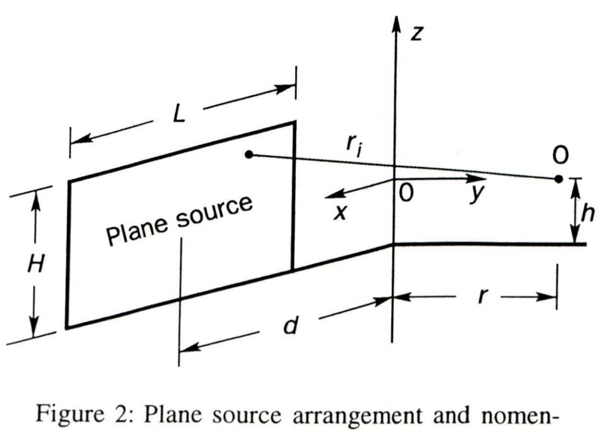

Figure 2: Plane source arrangement and nomenclature.

$$F = \frac{\gamma^2}{\alpha}\left[\begin{array}{l} \tan^{-1}\dfrac{(\alpha-\beta)(\delta+{}^1/_2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta+{}^1/_2)^2+\gamma^2}} \\ +\tan^{-1}\dfrac{\beta(\delta+{}^1/_2)}{\gamma\sqrt{\beta^2+(\delta+{}^1/_2)^2+\gamma^2}} \\ -\tan^{-1}\dfrac{(\alpha-\beta)(\delta-{}^1/_2)}{\gamma\sqrt{(\alpha-\beta)^2+(\delta-{}^1/_2)^2+\gamma^2}} \\ -\tan^{-1}\dfrac{\beta(\delta-{}^1/_2)}{\gamma\sqrt{\beta^2+(\delta-{}^1/_2)^2+\gamma^2}} \end{array}\right] \tag{6}$$

Alternatively, in the case of the far field, where (γ/10) > α,β,δ, the function F = 1 and the corresponding term in Equation (5) is zero.

Directivity factor, DI_M

The directivity factor, DI_M excludes reflection in the ground plane but includes reflection from other reflecting surfaces (Bies and Hansen, 1988, Ch.5) as well as the source inherent directivity.

Excess attenuation factor, A_E

The excess attenuation factor, A_E, is defined as the sum in decibels (dB) of five separate terms as follows.

$$A_E = A_a + A_b + A_f + A_g + A_m \tag{7}$$

The terms of Equation (7) are A_a, the attenuation due to air absorption; A_b, the attenuation due to barriers; A_f, the attenuation due to forests (if present); A_g the attenuation (which may be a gain rather than a loss) due to reflection in the ground plane and A_m the attenuation due to meteorological effects such as wind and temperature gradients (which may also be either a gain or a loss). Each of these terms will be discussed in the following sections, except for barrier and forest effects (A_b, A_f) which are discussed in detail in Bies and Hansen (1988).

Air absorption, A_M

An extensive review of literature on sound propagation in the atmosphere is provided in the report Conservation of Clean Air, Water and Energy (CONCAWE) (Manning, 1981). The author recommends the method of Sutherland (Gill, 1980) as being the best available scheme for calculating air absorption and quotes an accuracy within ± 10% from 0 °C to 40 °C.

Air absorption, A_a, is dependent upon temperature and relative humidity. Calculated values of absorption rate m (Gill, 1980), averaged over an octave band, are listed in Table 1 for the frequencies shown for representative values of temperature and relative humidity. For propagation over distance X, the absorption A_a is

Table 1: Attenuation due to atmospheric absorption (calculated using Sutherland's method (Sutherland et al. 1979).

Relative Humidity (%)	Temp (°C)	m (dB per 1,000 m)							
		63Hz	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
25	15	0.2	0.6	1.3	2.4	5.9	19.3	66.9	198.0
	20	0.2	0.6	1.5	2.6	5.4	15.5	53.7	180.5
	25	0.2	0.6	1.6	3.1	5.6	13.5	43.6	153.4
	30	0.1	0.5	1.7	3.7	6.5	13.0	37.0	128.2
50	15	0.1	0.4	1.2	2.4	4.3	10.3	33.2	118.4
	20	0.1	0.4	1.2	2.8	5.0	10.0	28.1	97.4
	25	0.1	0.3	1.2	3.2	6.2	10.8	25.6	82.2
	30	0.1	0.3	1.1	3.4	7.4	12.8	25.4	72.4
75	15	0.1	0.3	1.0	2.4	4.5	8.7	23.7	81.6
	20	0.1	0.3	0.9	2.7	5.5	9.6	22.0	69.1
	25	0.1	0.2	0.9	2.8	6.5	11.5	22.4	61.5
	30	0.1	0.2	0.8	2.7	7.4	14.2	24.0	58.4

$$A_a = mX \tag{8}$$

Ground effects, method 1, A_g

If it is assumed that the reflection from the ground is incoherent with the direct wave (as is probable at very large distances from the source, then the ground effect may be determined by calculating the ground reflection coefficient to determine the attenuation of the reflected wave and combining the result incoherently with the direct wave (that is, adding pressures squared).

The plane wave reflection loss in decibels given by $A_R = -20\log_{10}|R_p|$, where R_p has been calculated using the following equation.

$$R_p = \frac{Z_m\cos\theta - \rho c\cos\psi}{Z_m\cos\theta + \rho c\cos\psi} \tag{9}$$

where

$$\cos\psi = \sqrt{1 - \left(\frac{k}{k_m}\right)^2 \sin^2\theta} \tag{10}$$

Reference to Equation (10) shows that when $k_m = k_2 >> k_1 = k$, the angle ψ tends to zero and Equation (9) reduces to the following form,

$$R_p = \frac{Z_m\cos\theta - \rho c}{Z_m\cos\theta + \rho c} \tag{11}$$

which is the equation for a locally reactive surface.

In the preceding equations, k_m and Z_m are the complex propagation constant and complex impedance of the ground respectively and may be calculated using the procedure outlined by Bies and Hansen (1988, Appendix 3). The quantity ρc is the characteristic impedance of air and the angle θ is the angle of the incident wave measured from the normal to the ground surface.

The excess attenuation A_g is calculated as follows.

$$A_g = -10\log_{10}\left[1 + |R_p|^2\right] = -10\log_{10}\left[1 + 10^{-A_R/10}\right] \tag{12a,b}$$

The quantity A_g will vary between 0 and -3 dB, depending on the value of A_R (which is always positive).

Ground effects, method 2, A_g

A more accurate (and more complex) means of determining the ground effect is to assume spherical wave reflection including the effects of turbulence, to calculate the corresponding reflection term Γ, defined later in eq. 35. The excess attenuation, A_g, may then be calculated using the following equation.

$$A_g = -10\log_{10}[1 + \Gamma] \tag{13}$$

The problem of determining the reflection coefficient, R_s, for a spherical wave incident upon a plane interface between two media which is produced by a point source above the interface has been considered by Rudnick (1947) and more recently by Attenborough (1994a,b), who gives the following expression for the spherical wave reflection coefficient.

$$R_s = R_p + BG(w)(1 - R_p) \tag{14}$$

In Equation (14), R_p is the plane wave reflection coefficient discussed previously. For the general case that the reflecting interface is extensively reactive, B is defined as follows.

$$B = \frac{B_1 B_2}{B_3 B_4 B_5} \qquad (15)$$

where

$$B_1 = \left[\cos\theta + \frac{\rho c}{Z_m}\left(1 - \frac{k^2}{k_m^2}\sin^2\theta\right)^{1/2}\right]\left[1 - \frac{k^2}{k_m^2}\right]^{1/2} \qquad (16)$$

$$B_2 = \left[\left(1 - \frac{1}{\rho_m^2}\right)^{1/2} + \frac{\rho c}{Z_m}\left(1 - \frac{k^2}{k^2}\right)^{1/2}\cos\theta + \left(1 - \left(\frac{\rho c}{Z_m}\right)^2\right)\sin\theta\right]^{1/2} \qquad (17)$$

$$B_3 = \cos\theta + \frac{\rho c}{Z_m}\left(1 - \frac{k^2}{k_m^2}\right)^{1/2}\left[1 - \frac{1}{\rho_m^2}\right]^{-1/2} \qquad (18)$$

$$B_4 = \left[1 - \frac{k^2}{k_m^2}\sin\theta\right]^{1/2} \qquad (19)$$

$$B_5 = \left[1 - \frac{1}{\rho_m^2}\right]^{3/2}[2\sin\theta]^{1/2}\left[1 - \left(\frac{\rho c}{Z_m}\right)^2\right]^{1/2} \qquad (20)$$

where θ is defined in Figure 3.

The argument, w, of $G(w)$, called the numerical distance, in equation (14) is calculated using the following equation.

$$w = \frac{1}{2}(1 - j)[2k(r_1 + r_2)]^{1/2}\frac{B_3}{B_6^{1/2}} \qquad (21)$$

In equation (21), B_3 is defined above by equation (18) and B_6 is defined as follows.

$$B_6 = 1 + \left[\frac{\rho c}{Z_m}\left(1 - \frac{k^2}{k_m^2}\right)^{1/2}\cos\theta + \left(1 - \frac{\rho c}{Z_m}\right)^{1/2}\sin\theta\right]\left[1 - \frac{1}{\rho_m^2}\right]^{-1/2} \qquad (22)$$

The term $G(w)$ in equation (14) is defined as follows.

$$G(w) = 1 - j\sqrt{\pi}\,w g(w) \qquad (23)$$

where

$$g(w) = e^{-w^2} erfc(jw) \qquad (24)$$

For small w ($|w| < 3$)

$$g(w) = e^{-w^2} - \left[\frac{2jw}{\pi^{1/2}}\right]\sum_{n=0}^{\infty}\frac{(-2w^2)^n}{1x3x\ldots x(2n+1)} \qquad (25)$$

For values of w where the real part is greater than 3 or the imaginary part is greater than 2 and either is less than 6,

$$g(w) = -jw\left[\frac{0.4613135}{w^2 - 0.1901635} + \frac{0.09999216}{w^2 - 1.7844927} + \frac{0.002883894}{w^2 - 5.5253437}\right] \qquad (26)$$

For real or imaginary parts of w greater than 6

$$g(w) = -jw\left[\frac{0.5124242}{w^2 - 0.275255} + \frac{0.05176536}{w^2 - 2.724745}\right] \qquad (27)$$

As shown by Rudnick (1947) the numerical distance, w, becomes very large at large distances from the source and the function $G(w)$ tends to zero. Reference to equation (14) shows that as $G(w)$ tends to zero the reflection coefficient for spherical waves becomes the reflection coefficient for plane waves. Alternatively, w approaches zero close to the source and $G(w)$ approaches 1.

For the case that the frequency of concern, f, satisfies the following relation, the porous surface is essentially locally reactive.

$$f < 10^{-3} R_1 \qquad (28)$$

When equation (28) is satisfied and the porous surface is essentially locally reactive the following simplifications are possible.

$$B_1 = B_3 = \cos\theta + \frac{\rho c}{Z_m} \qquad (29a,b)$$

$$B_2 = (1 + \sin\theta)^{1/2} \qquad (30)$$

$$B_4 = 1 \qquad (31)$$

$$B_5 = (2\sin\theta)^{1/2} \qquad (32)$$

$$B_6 = B_2^2 \qquad (33)$$

Equation (21) becomes

$$w = \frac{1}{2}(1 - j)[2k_1(r_1 + r_2)]^{1/2}\left(\cos\theta + \frac{\rho c}{Z_m}\right)(1 + \sin\theta)^{-1/2} \qquad (34)$$

Effects of turbulence

Turbulence in the acoustic medium containing the direct and reflected waves has a significant effect on the effective surface spherical wave reflection coefficient. This effect will now be discussed with particular reference to sound propagation outdoors over the ground. Experience suggests that local turbulence near the ground is especially important because it introduces variability of phase between the reflected sound and the direct sound from the source to the receiver. Variability in phase between the direct and reflected sound determines whether the two sounds, the direct and the reflected sound, should be considered as adding coherently or incoherently. Coherent reflection requires minimal variability of phase and can result in constructive or destructive interference in which case the variation in level can be very large while incoherent reflection, associated with a large variability of phase, can result in at most a 3 dB variation in observed level.

Solar driven local air currents near the ground, which result as the ground heats up during the day relative to the cooler air above, will cause local convection and turbulence near the ground of the kind of concern here. Sound of wavelength of the order of or less than the turbulence scale will be observed to warble strongly only a short distance away from the source when observed across a paved parking lot. The model proposed here suggests that coherent reflection should be observed more often at night than during the day.

The effect of turbulence on sound propagation over an acoustically smooth surface has been investigated by Clifford and Lataitis (1983) and by Raspet and Wu (1995). The presence or absence of turbulence may be included by a generalisation of their results to give the following general expression for the reflection term, Γ,

$$\Gamma = \frac{r^2 - (1-T)\left(r^2 - (r_1 + r_2)^2\right)}{(r_1 + r_2)^2} |R_s|^2 \qquad (35)$$

$$+ \frac{2Tr}{r_1 + r_2}\left(\cos[k(r_1 + r_2 - r)]\mathrm{Re}\{R_s\} - \sin[k(r_1 + r_2 - r)]\mathrm{Im}\{R_s\}\right)$$

which can be used to calculate the excess attenuation due to the combined ground and turbulence effect as follows,

$$A_g = -10\log_{10}(1+\Gamma) \qquad (36)$$

In equation (35), R_s is the spherical wave reflection coefficient given by Equation (14) and the distances r, r_1 and r_2 are shown in Figure 3.

The exact solution (Clifford and Lataitis, 1983; and Raspet and Wu, 1995) for the term T which appears in equation (35) is very complicated. However, simplifications are possible which lead to the following approximate expression.

$$T = e^{-4\alpha\pi^{5/2}<n_1^2>10^3\Phi} \qquad (37)$$

where

$$\Phi = 0.001\frac{L L_0}{\lambda^2} \qquad (38)$$

In equation (38), L is the horizontal distance between the source and receiver (see Fig. 3), L_0 is the scale of the local turbulence and λ is the wavelength of the sound under consideration. A value of L_0 of about 1 to 1.2 m is suggested if this quantity is unknown or cannot be measured conveniently. When Φ is greater than 1, incoherent reflection can be expected and when Φ is less than 0.1 coherent reflection can be expected. If reflection is incoherent then the spherical wave reflection coefficient given by Equation (14) reduces to the simpler plane wave reflection coefficient.

In equation (37), α has the value 0.5 when $L/k \ll L_0^2$ and the value 1 when $L/k \gg L_0^2$. For values of L/k and L_0 which do not satisfy either of the imposed conditions use of the exact formulation for T may be used (Raspet and Wu, 1995). However, the exact solution is extremely complicated on the one hand and on the other it is reasonable to expect that α will be bounded by its extreme values, 0.5 and 1. Consequently, as the far field is generally satisfied by the value $\alpha = 0.5$, the latter value will be used to evaluate the exponent of the right hand side of equation (37).

The term $<n_1^2>$ is the mean square of the fluctuations in the speed of sound relative to its mean value in the absence of turbulence. A value of 5×10^{-6} is suggested (Raspet and Wu, 1995). Evaluation of the exponent allows equation (37) to be rewritten in terms of the figure of merit, Φ, introduced in equation (38) as follows.

$$T = e^{-0.17\Phi} \qquad (39)$$

As the figure of merit, Φ, approaches 0 indicating little or no turbulence the term T approaches 1, resulting in coherent ground reflection and a large variation in attenuation as a function of distance and frequency. On the other hand, as the figure of merit becomes large, indicating a large turbulence effect, the term T approaches 0, equation (35) reduces to $\Gamma = |R_s|^2$, and at large distances it can be seen from equation (14) that $R_p = R_s$ (as $G(w)$ goes to zero), and the direct and ground reflected waves are combined incoherently, resulting in an expected excess attenuation of between -3 dB and 0 dB.

The excess attenuation, A_g averaged over a one-third octave or octave band is calculated by combining logarithmically the attenuations calculated at several different frequencies (at least 10) equally spaced throughout the band. This band-averaged attenuation is probably more useful for noise predictions in practice, as atmospheric turbulence and undulating ground will result in considerable fluctuations in the single frequency values.

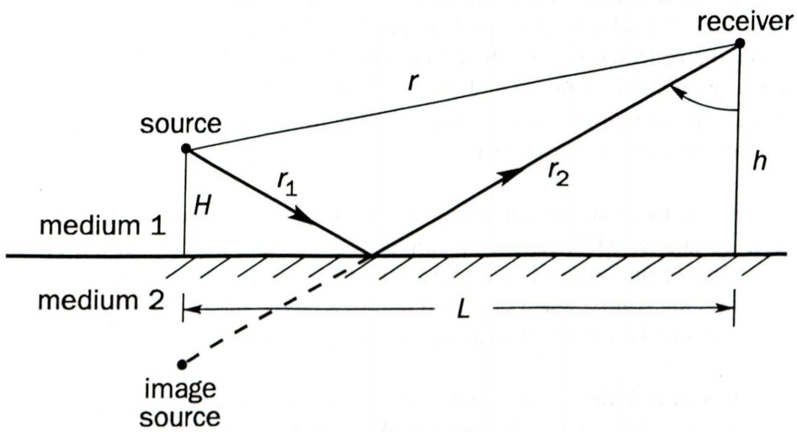

Figure 3: Arrangement and nomenclature for reflection of sound in the ground plane.

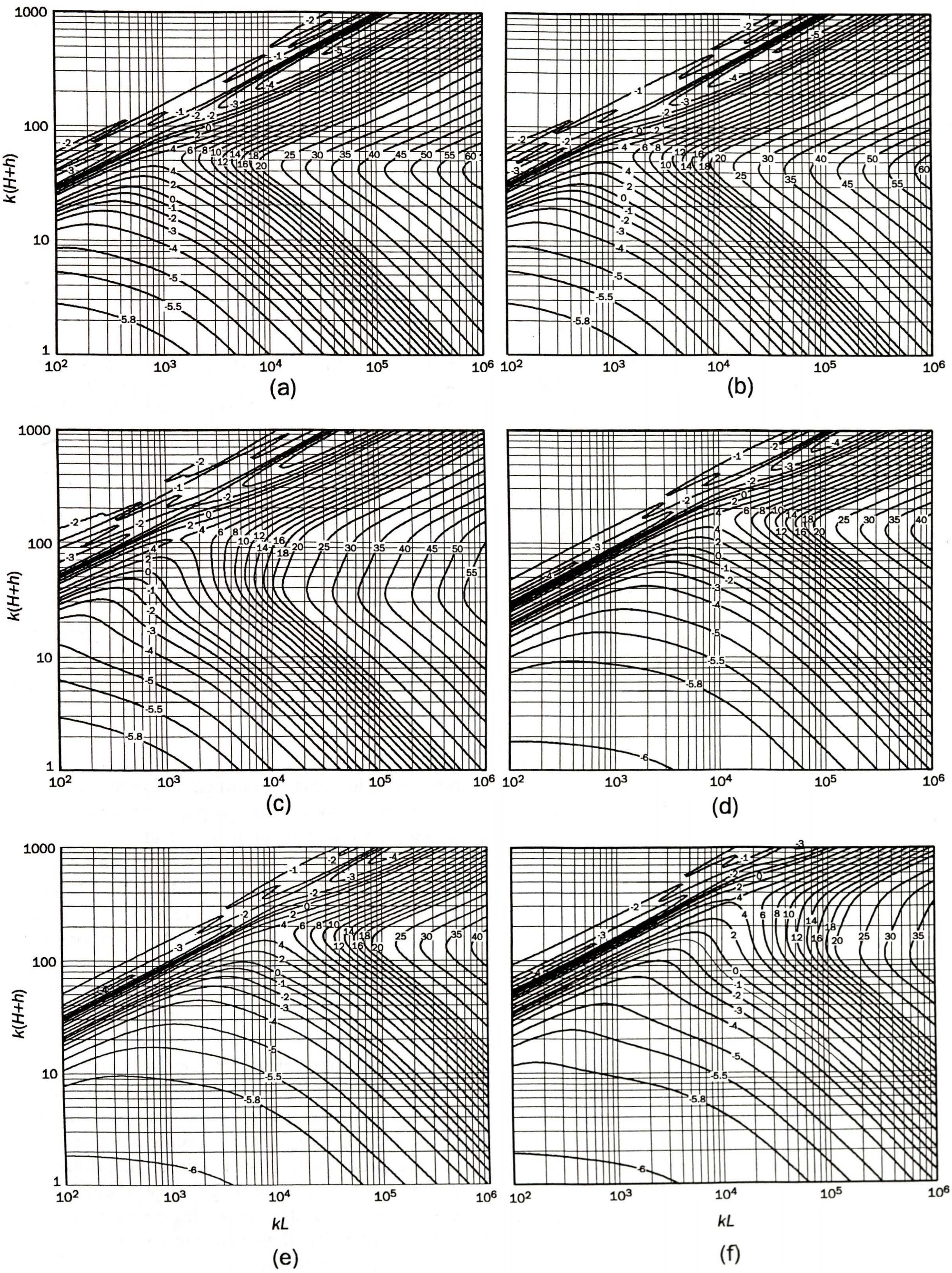

Figure 4: Calculated contours (shown parametrically in decibels in the figure) of octave band excess sound attenuation due to ground reflection and propagation. Source height, H (m), receiver height, h (m), distance between source and receiver, L (m), wavenumber, k (m^{-1}). Surface flow resistivity: (a) 200,000 MKS rayls m^{-1}, H = h. (b) 200,000 MKS rayls m^{-1}, H = 2h. (c) 200,000 MKS rayls m^{-1}, H = 10h. (d) 6,000,000 MKS rayls m^{-1}, H = h. (e) 6,000,000 MSK rayls m^{-1}, H = 2h. (f) 6,0000,000 MKS rayls m^{-1}, H = 10h.

Examples of octave band averaged excess attenuation A_g, calculated assuming coherent combination of the direct and reflected waves using equation (14), as a function of non-dimensional distance kL (wavenumber k, distance L) from the source, are presented in Figures 4a-f for a source height of H and a receiver height of h. The ground flow resistivities used in the calculations were chosen from the mid range of the values given by Embleton et. al., (1983); 2 x 10^5 MKS rayls m^{-1} for grass-covered ground and 6 x 10^6 MKS rayls m^{-1} for hard-packed earth. For each type of ground surface, data for three ratios of source height to receiver height are presented (H/h = 1, 2, 10). As expected, the curves show that for small source heights (relative to a wavelength) and large distances from the source, the relative source to receiver height is not important.

Meteorological effects, A_m

The two principal meteorological variables are wind distribution and vertical temperature gradient. When the temperature increases with height and the gradient is thus positive the condition is termed an inversion. When the temperature decreases with height and the gradient is thus negative the condition is termed a lapse. It has been established (Rudnick, 1957) that curvature of sound propagation is mainly dependent upon the vertical gradient of the speed of sound, which can be caused either by wind or by a temperature gradient, or by both. It has also been shown (Piercy, Embleton and Sutherland, 1977) that refraction due to either vertical wind or vertical temperature gradients produces equivalent acoustic effects which are essentially additive.

A positive temperature gradient (temperature inversion) near the ground will result in sound waves that normally travel upwards being diffracted downwards towards the ground, resulting, in turn, in increased noise levels on the ground. Alternatively, a negative temperature gradient (temperature lapse) will result in reduced noise levels on the ground.

As wind speed generally increases with altitude, wind blowing towards the listener from the source will diffract sound waves downwards, resulting in increased noise levels. Conversely, wind blowing from the listener to the source will diffract sound waves upward, resulting in reduced noise levels. Atmospheric turbulence results in fluctuations of the sound received by the listener; the effect is usually greater during the day than at night.

If the weather conditions are accurately known, then the methods outlined in the literature (Manning, 1981; Tonin, 1985) may be used to predict the excess attenuation due to meteorological effects. Otherwise, it is generally assumed that meteorological effects will result in a sound level variation about the predicted level, as shown in Table 2.

Alternatively, recent work (Stinson, et al, 1995; Di and Daigle, 1994; and Gilbert and Di, 1993) has shown that it is possible to combine meteorological effects with ground effects and turbulence. Although the work represents a great step forward, the input data requirements regarding atmospheric temperature and wind profiles are extensive as is the calculation procedure, thus making it difficult to implement in practice.

Table 2: Variability in sound level predictions due to meteorological influences.

Octave band centre frequency (Hz)	Distance from source (m) 100	200	500	1,000
(dB)				
63	±1	+4, -2	+7, -2	+8, -2
125	±1	+4, -2	+6, -4	+7, -4
250	+3, -1	+5, -3	+6, -5	+7, -6
500	+3, -1	+6, -3	+7, -5	+9, -7
1,000	+7, -1	+11, -3	+12, -5	+12, -5
2,000	+2, -3	+5, -4	+7, -5	+7, -5
4,000	+2, -1	+6, -4	+8, -6	+9, -7
8,000	+2, -1	+6, -4	+8, -6	+9, -7

Conclusions

A procedure has been described for predicting the attenuation of outdoor sound propagation which includes the combined effect of ground and atmospheric turbulence added to other effects. However, the problem of accurately predicting outdoor sound attenuation remains complex with a number of interrelating factors determining the sound level at a particular location. Nevertheless, more complex prediction algorithms which take the interrelationships between various effects into account are gradually replacing algorithms which consider each effect separately, with a corresponding increase in prediction accuracy. However the price paid for increased accuracy is increased detail regarding wind and temperature profiles, ground impedance and turbulence levels. The complexity of the accurate calculations precludes the production of generalised, non-dimensional charts and tables, due to the huge range of combinations possible for the input data, thus relegating future outdoor noise propagation predictions to the computer equipped with sophisticated software.

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Calculation Of Effects Of Wind Speed Fluctuations On Pulse Waveform.

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Analysis of previous impulsive sound studies.

Impulsive sound propagating over distances from 5m to 16m has been studied extensively by the authors[1] and details of the source used for these measurements and a typical experimental layout are described elsewhere[2,3]. The conclusions reached in these studies were:

1. The propagation time can be accurately calculated from the wind speed measurements, especially if ray bending corrections are included.
2. In the time domain it is the peak pressure of the pulse that experiences the greatest variation due to the presence of turbulence. After propagating for a distance of only 16m, the peak pressure varies by less than 3% indoors, however, in the presence of outdoor winds the changes can halve or double the peak pressure.
3. The pulse width can vary significantly, generally increasing with propagation distance. At 5m the received pulse shape is relatively unchanged.
4. There is no evidence of an increase in energy present in the tail of the pulse, indicating that scattering of energy back into the pulse from alternate distant paths is not a major contribution.
5. Changes in the time domain acoustic parameters such as peak pressure, width and energy content do not correlate with the wind speed fluctuations (eg. wind speed difference between source and receiver, mean wind speed or integrated wind speed over the propagation path).

Present Work

The lack of correlation obtained from time domain analysis of the received pulses suggests that turbulent effects may be better understood by frequency analysis. Individual pulses were Fourier transformed and the component amplitudes at various frequencies were compared to the peak pressure. Figure 1 shows the results at 1kHz and 10kHz for propagation distances of 5.3m and 16m, where 1 on the vertical and horizontal axis corresponds to the amplitude of pulses measured indoors. The changes in amplitude at 1kHz show a slight correlation with the peak pressure at 5.3m, however, by 16m there is little correlation. At 10kHz the reverse occurs: there is a weak correlation between amplitude and peak pressure at 5.3m but good correlation at 16m. It should be noted that the majority of the pulse energy is concentrated around 1kHz, however, there is still significant energy at 10kHz and higher, so that changes in these components will affect the pressure peak of the pulse. The data in Fig. 1 indicate that the range of component amplitudes at 1kHz is independent of the increase in distance, whereas the 10kHz component shows a significant change.

Background theory

Effects of turbulence on sound have, in the past, been described in terms of the standard deviation in sound pressure amplitude[4]. Using the theory of Tatarski[5] the standard deviation in pressure amplitude is dependent upon the square of the acoustic frequency (f), the propagation length (x), the standard deviation for time averaged wind speed and the correlation length of the wind speed (L), as given by Eq. 1. The braces in Eq. 1 indicate time averaging over several minutes, corresponding to the period of the wind speed.

$$\left\langle \left|P - P_o\right|^2 \right\rangle = (2\pi f)^2 \left\langle \left|u - u_o\right|^2 \right\rangle Lx \qquad (1)$$

Note Eq. 1 suggests that the ratio of changes in one frequency compared to another should be independent of distance. This contradicts the observations from Fig.1. Tatarski identified the appropriate wave equation for the acoustic pressure including the effects of turbulence. He then solved it by expanding the wavefunction, P, into a non turbulent part, P_o , and a perturbed (to first order) part. A Greens function method was used, comprising the free Greens function G_o and the perturbed Greens function, G_1, to obtain the following integral for the perturbed part containing the turbulence effects:

$$P(x,t) = \int \left[G_o(x,t,x',t') + G_1(x,t,x',t') \right] P(x',t')\, d^3x'\, dt'$$

$$= P_o(x,t) + \frac{1}{2\pi c^2} \int \left[\frac{\mathbf{u}(x',t')\nabla}{|x-x'|} \frac{\partial}{\partial t'} P_o(x',t') \right]_{t'=t-\frac{|x-x'|}{c}} d^3x'$$

$$= P_o + \int \frac{u(x',t)H(x',t)}{|x-x'|} d^3x' \qquad (2)$$

This integral does not have an analytical solution as the wind speed, u, is generally a random function of distance and time. An expression for the time averaged standard

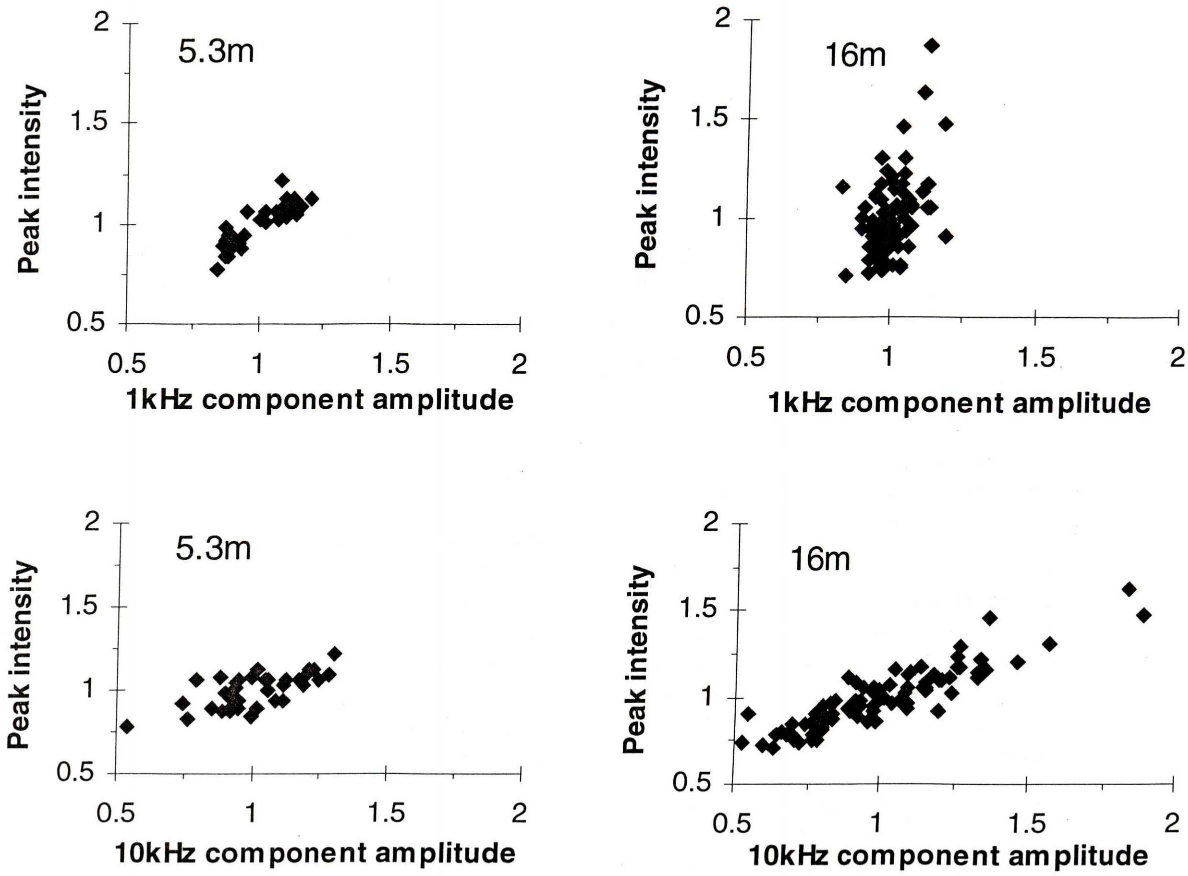

Figure 1. Frequency information derived from pulses for two different source-receiver distances.

deviation of the sound pressure about the average pressure, which corresponds to the non turbulent sound, can be derived and has the form

$$\left\langle \left|P - P_o\right|^2 \right\rangle = \int\int \left\langle u(x')u(x'')\right\rangle \frac{H(x')H(x'')}{\left|x - x'\right|\,\left|x - x''\right|} dx'dx'' \tag{3}$$

This can have an analytical solution by assuming a suitable form for the autocorrelation of wind speed. A Guassian correlation function for the time averaged wind speed, Eq. 4, was used in Eq. 3 to obtain Eq. 1.

$$\left\langle u(x)u(x')\right\rangle = \left\langle |u|^2 \right\rangle e^{-\left(\frac{x-x'}{L}\right)^2} \tag{4}$$

In this work, the original theory of Tatarski was used, avoiding several assumptions implicit in Eq. 1 to obtain a result which can be applied to impulsive sound data. The important facts concerning Eq. 1 are:

- the effects have been integrated over all space.
- the change in the wind speed over distance is assumed to be adequately described by a correlation function.
- the calculation is valid for averages taken over several minutes, which corresponds to several cycles of the wind speed structure.

Equation 1 has the advantage of requiring only two parameters, the correlation length and the variation in time of the wind speed: which can be determined by two wind speed probes separated by a fixed distance. However, the impulse sounds under consideration only occur for a few milliseconds, invalidating point 3 above. Thus, for impulses, it is necessary to solve Eq. 2. This requires the wind speed at all positions over a large area to be known, which implies a multitude of probes, especially if large propagation distances are involved.

Because the flight time of an impulsive sound over a distance of, say, 20m is only 59ms, the atmosphere can often be regarded as "frozen in time" during this interval, since little change in wind speed may occur even in a 4 or 5 second interval. Therefore, there is no need to

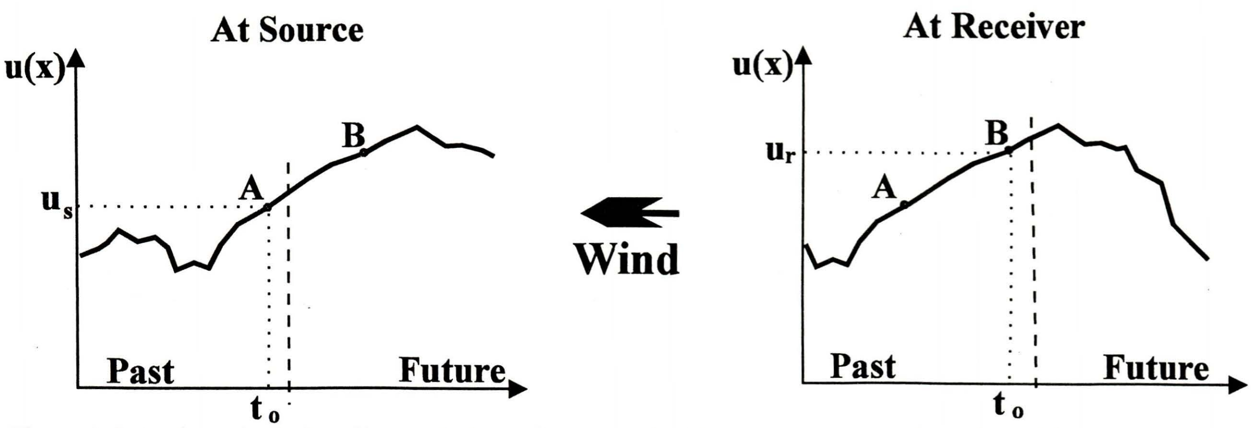

Figure 2. Selection criteria for a linear wind speed pattern. The interval between t_o and the dashed line represents the flight time of the pulse.

undertake the time averaging implied by the theory. If it is possible to determine the wind speed pattern over a short propagation path, then Eq. 2 can be solved. Here, this has been done by modelling the wind speed as a linear function in distance and afterwards comparing the predictions with data from pulses propagated under appropriate conditions. From previous field measurements, large changes in peak pressure have been observed for 1ms long acoustic pulses over 16m. This means that the perturbed part in Eq. 2 can be significant for an individual pulse, though when averaged over hundreds of pulses the standard deviation in pressure is small.

The calculation

The acoustic pressure was calculated in sufficiently small steps that the perturbed term in each step was less than 20% of the unperturbed part. For the purpose of these calculations we neglect any ray bending and assume the pulse travels in essentially a straight line with no energy scattered in from distant parts of the atmosphere. $P_o(x,t)$ is taken to be a plane wave, or more correctly the pulse is Fourier decomposed into plain wave components, of which $P_o(x,t)$ is the one at frequency f. Assuming u is a linear function of displacement, Eq. 2 gives

$$P(x,t) = P_o(x,t)\left(1+\frac{(2\pi f)^2 \Delta u}{2\pi c^3}\right) \tag{5}$$

where $\Delta u = u(x) - u(0)$ with the source and receiver located at 0 and x respectively..

To understand how a particular wind speed condition was determined to be linear in distance, consider Fig. 2 which represents typical wind speed time traces recorded simultaneously at the source and receiver. Time t_o corresponds to the moment that the pulse leaves the source, labelled A and B on both time traces of Fig. 2, and it arrives at the receiver 15ms later. This is indicated by the dashed line in Fig. 2, the shift is exaggerated for clarity. The wind speed effectively does not change during the pulse flight time. Under the conditions sought here, the wind speed pattern as a function of distance, *u(x)*, moves at a mean speed, u_{av}, so that a time trace at the receiver will resemble that of the source but shifted in time, the shift being the source receiver distance divided by the mean wind speed. If the wind speed is reasonably linear in distance between source and receiver then the rate of change in wind speed , $\frac{\partial u}{\partial t}$, at the source should be close to that at the receiver. The concept that the wind speed pattern moves, without a marked change, at a constant speed is a reasonable approximation over short distances such as 16m[6]. However, it is less likely that a linear section will occur just when the pulse is propagated. Thus, relatively few of the measured pulses meet the above criteria.

Equation 5 was evaluated at 1kHz and a propagation distance of 5.3 m for various Δu values and the results are plotted in Fig. 3. Also in Fig. 3 are the measured 1kHz components from the pulse data plotted against measured wind speed difference For the range of measured Δu values it was necessary to divide the space between source and receiver into 4 regions and calculate the value for each region separately, permitting the result of the previous region to be modified by the next. The single step result is shown as the broken line in Fig. 3,

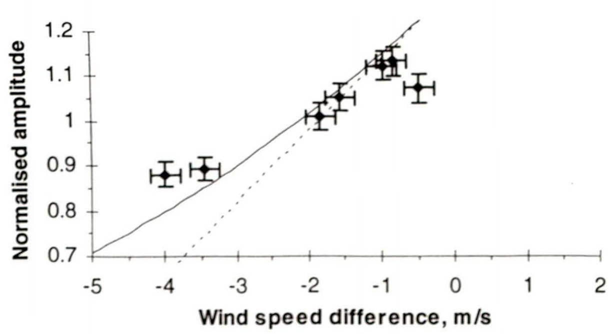

Figure 3. Comparison between theory and experiment at 1kHz.

while the result for four separate regions is the solid curve. The latter differs from a 3 step calculation by the width of the line, indicating that the limit had been achieved..

The agreement between data and theory in Fig. 3 is quite good. Unfortunately, the few pulses propagated under the type of wind structure required by Eq. 5. For greater distances the calculation of turbulence effects will require more complex expressions for the wind speed profile.

Conclusion

A better understanding of the effects of turbulence on impulsive sound may be achieved by considering the pulse behaviour in the frequency domain. At larger distances, around 16m, it is the higher frequency components (around 10kHz) which display effects due to turbulence. Such changes are responsible for the majority of variations observed in the peak pressure. At shorter distances the 1kHz and 10kHz show similar behaviour. Calculations assuming a linear wind speed profile over a propagation distance of 5.3m gave reasonable agreement with the measured data at 1 kHz. However, this type of wind speed profile does not often occur and certainly over larger distance, such as 16m, a more realistic wind speed function need to be considered. What is shown here is that even short distances require the propagation path to be divided into a number of small regions for calculation purposes.

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Acoustic Propagation Over Barriers With Caps and Apertures

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A hole or a crack in a barrier can alter the insertion loss of the barrier, typically decreasing the attenuation at some frequencies and producing enhancement at others. The prediction of such effects requires an appropriate model which, in turn, needs a full understanding of the various mechanisms which may contribute to the resultant sound. Using a continuous wave acoustic source, it is virtually impossible to separate these mechanisms as the different contributions simply add, with appropriate phase shifts, to give another wave of altered amplitude[1]. However, if a short duration pulse is utilised, different paths produce different delays, so the resulting waveform is a succession of pulses, which may partially overlap. By sorting out the resulting pattern, there is the potential to determine the mechanisms involved and, moreover, the relative importance of the mechanisms from the fraction of energy in the various pulses.

This philosophy has been utilised in the study of various shaped apertures placed on an acoustically hard wide barrier. Some preliminary results will be discussed shortly, however, a short summary of the measurement technique will be presented first.

Experimental arrangement

Like the set-up used in previous barrier studies involving an impulse source[2], an undeviated direct pulse waveform is compared with that of a pulse which has travelled an equivalent distance around the barrier, Fig.1. By always using the direct pulse to trigger the waveform capture electronics, a Data 6000 waveform analyser, any slight change in the flight time of the pulse which has passed around the barrier can be detected as well as the timing of any delayed pulse components. The sampling interval used in these experiments was 5μs.

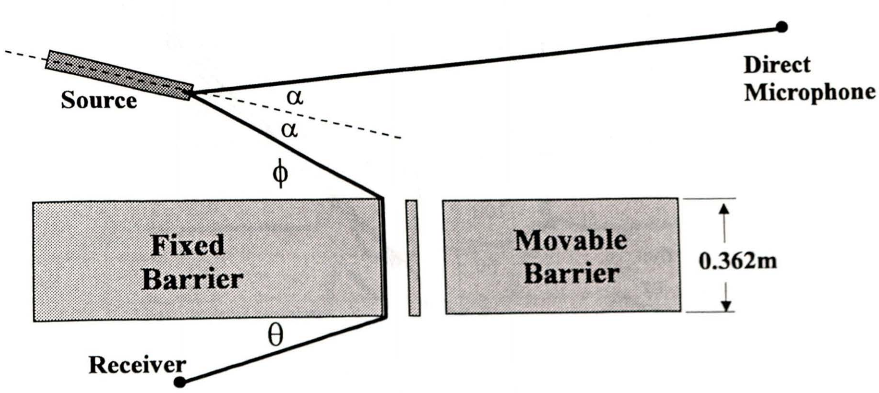

Fig.1: Plan view of experimental layout showing the case of a double slit in an wide barrier.

As described elsewhere[2], the barrier was mounted with the diffracting edge vertical to maximise the delay from unwanted reflections from the floor or ceiling of the laboratory. Measurements were taken on a plane 1.4m above the floor, such reflections were easily removed before analysis of the data. The fixed barrier used in these studies was solidly constructed from wood and was 0.362m across. When required, a movable second barrier of the same thickness could be positioned at various distances from the fixed barrier, forming a parallel sided slit of known width or a wedge-shaped slit. Alternatively, a board, 0.362m across and 12mm thick, could be mounted at different distances from the fixed barrier forming a cap or, when combined with the second barrier, a double slit system. The slit was formed along the full 2.5m vertical dimension of the barrier. All the measurements discussed in this work were obtained when both the source and receiver were 1m from the nearest diffracting edge of the fixed barrier and at a ϕ of 60°

Results

Barrier with single slit formed by cap or second barrier.

The pulse waveform obtained with a single 50mm wide slit and $\theta = 60^{\circ}$ is presented as the solid line in Fig.2 (a) along with the simple diffracted pulse measured when only the fixed barrier was present, Fig 2(b). In these, and all further waveforms, the vertical axis is pressure. Also shown by the dotted line in Fig.2(a) is the direct pulse from the source, scaled down in amplitude by a factor of 6.7. Inspection of the curves indicates that the signal received through the slit is composed of a simple diffracted pulse with a series of additional pulses superimposed at various intervals. At a given θ value, only about 3% of the energy incident on the slit passed through to the receiver with just over half of this energy being in the bare barrier diffraction component. Measurements using the same geometry but with a cap spaced 50mm away from the barrier were essentially identical and so will not be considered separately.

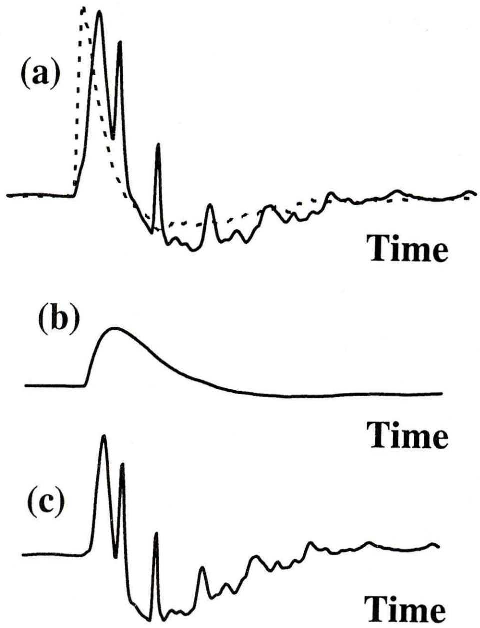

Fig.2: Pulse waveforms associated with a 50mm wide, parallel sided, slit in a barrier.

Careful observation of the onset of the direct pulse and that diffracted through the slit showed that they arrived within the sampling time of the system when the path lengths were equal and that they did not shift when different thickness slits were created, indicating that the diffracted pulse always travels across the face of the barrier at the free-space speed of sound. The shape of the diffracted pulse produced by a wide bare barrier can be calculated reasonably accurately using a number of theoretical models[3].

The nature of the superimposed pulse train becomes apparent when the simple diffracted waveform is subtracted from Fig. 2(a) to give the curve of Fig. 2(c). This is typical of the results for other slit widths, where a strong first peak, containing 56% of the received energy, is followed by a very narrow one (16%), the later peaks getting progressively broader and weaker. Eight or more peaks can be observed from a 20mm wide slit while beyond 150mm it is difficult to distinguish more than the first two peaks. The delay of a particular peak is best made by reference to the onset of the diffracted peak, however, the onset of other peaks is often difficult to judge, especially in the case of the small broad peaks which occur at greater delays. Thus, time differences were obtained using the peak position.

The explanation of these peaks would appear to be the formation of various modes within the slit[4], or waveguide, as suggested by the ray-type diagram of Fig.3. The path difference between that taken by the diffracted signal which just grazes the top of the fixed barrier and the length travelled in the different modes can be calculated from the known geometry and plotted against the measured delay, as in Fig.4. The trend would

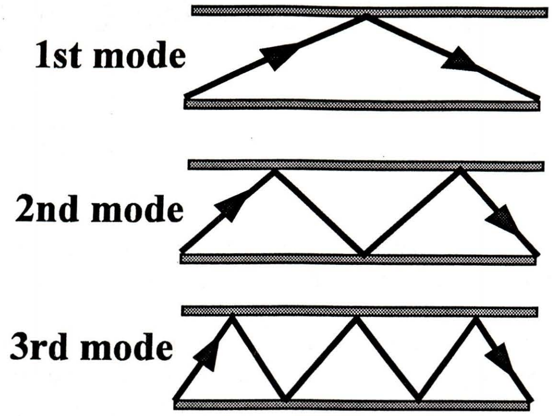

Fig.3: Possible modes in guide.

appear to verify the mode model, however, the gradient of this graph indicates that the wave-speed of the sound through the waveguide is 355ms^{-1}. It might be noted that the first, and strongest peak, rarely fits the trend of later peaks. For comparison, the dotted line corresponds to a speed of 380ms^{-1} while the dashed line is 340ms^{-1}. Similar data can be gathered for other slit widths; the estimates of the speed of sound being given in Table 1. While increasing consistently, many of the speeds appear to be unacceptably high. Two independent experiments have since been carried out to determine if the speed of sound in such a parallel sided waveguide is greater than in free space. These confirmed that the speed is greater, however, they do not suggest such a large increase. One experiment indicated that the wave speed does not return to the free-space value until the separation between the walls of the guide exceeds 0.5m.

For slit widths exceeding 0.1m, another possible ray path exists - direct reflection of the sound from the surface of the guide (i.e. barrier) facing the source and receiver. When this path difference is calculated it is only

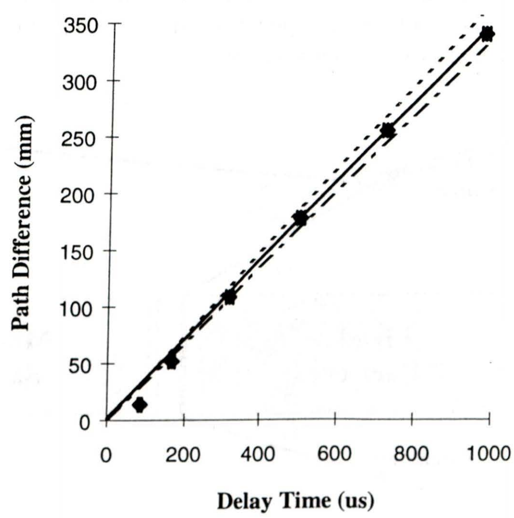

Fig.4: Estimation of speed in guide.

Table 1: Apparent sound speeds for different slit widths.

Slit width (mm)	20	30	50	70	100	150
Sound speed (ms^{-1})	320±5	340±5	355±8	365±10	370±10	400±20

Table 2: Apparent speed as function of angle θ.

Angle	20°	40°	60°	80°
Sound speed (ms^{-1})	335±5	345±8	355±8	390±20

slightly shorter than the difference for the first mode. Indeed, at such distances the "first" peak after the diffraction pulse does split into two distinct components at 110mm, the first having a higher amplitude, consistent with this concept.

The mode model of Fig.3 predicts path differences which are independent of the angles ϕ and θ. As indicated in Table 2, measurements made at different θ values for a 50mm wide slit, do show an angle dependence.

At small θ values, over 10 well defined peaks are apparent, however, beyond 70° they become small and broaden making it difficult to be certain of the position of more than 3 peaks. Various attempts have been made, unsuccessfully thus far, to modify the mode model to make it more realistic. For example, by assuming the source of radiation is, say, 1 cm outside the physical end of the guide (analogous to the end correction for an organ tube), it is possible to obtain an angular dependence, although not of the correct magnitude. However, extending the effective path length in this way also increases the effective speed in the guide.

Angled slit

In these experiments, the mobile barrier was rotated about the corner closest to the source to form a wedge shaped slit. The opening of the slit on the incidence side was maintained at 100mm and θ was 60°. The received pulses display an overall behavior which is similar to that discussed above in that they consist of at least two parts, a diffraction waveform and an additional disturbance with high peaks.

Figure 5 shows the residual pulse after subtracting the diffracted pulse over a bare barrier from the received pulse. Results (a) to (e), which correspond to moving from a contracting, to parallel and then diverging slit, indicate that the peaks

- move to later times
- for the same order, evolve from wider smaller disturbances, through sharper peaks in the near parallel slits, to return to wider and smaller maxima for the most divergent case.

The first peaks in (d) and (e) show significant changes in their trailing edge, possibly indicating that they could be made up of several unresolved peaks. Unlike the parallel case, the wedge geometry has a limited number of modes which obey a pure Snellian reflection law. In general, the results displayed more pulses than the number predicted by the simple mode model. As a result, it is difficult to determine a sound speed, as in Fig. 4, although overall value of 350 ±10 ms^{-1} appears to fit the data.

Two parallel slits

This situation will be discussed in terms of two 100mm wide slits, with θ and ϕ both 60°. The simplest model is to assume that the two slits are independent, which is supported by the general appearance of the received pulse waveform shown as Fig.6(a).

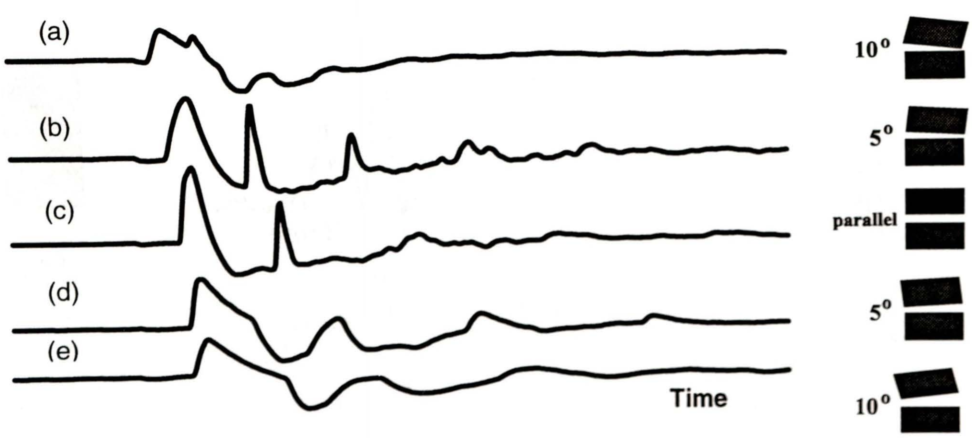

Fig. 5: Residual Waveforms for a Variety of Wedge Shaped Gaps

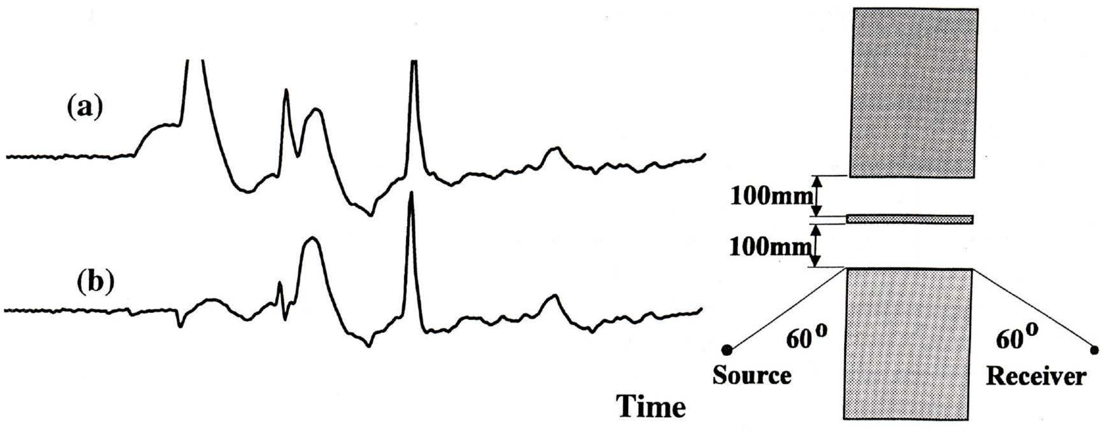

Fig.6: Waveforms resulting from two 100mm wide slits separated by a 12mm wide opaque region.

Because of the wider slit, more of the diffraction peak is evident, before the arrival of the first strong peak, than in Fig.2. When the waveform obtained from a single 100mm slit geometry is subtracted from that of Fig.6(a), the result is as shown in Fig.6(b). There is almost complete cancellation of the first peaks, leaving what appears to be the first two modes for sound moving through the second slit. Calculations based on a 355ms^{-1} sound speed and the geometric path differences expected for the second slit were in reasonable agreement with the timing of the residual peaks.

Two parellel slits with foam lining

The effect of placing different types of foam on one surface of a double slit system is shown in Fig.7. Waveform (a) is the full received signal which, as above, has initial peaks characteristic of sound passing through the first slit. When such a signal is subtracted, the residual waveforms are those in Fig. 7(b) and (c) for a "yellow" foam and a "black" foam respectively. After the small residue caused by imperfect subtraction, the dominant peak is due to sound passing directly through the slit combined with sound transmitted through the foam itself. The yellow foam (flow resistivity 1800Pa s m^{-2}) is relatively soft compared to the hard black foam (25000Pa s m^{-2}). It appears that the yellow foam allows more leakage[4], resulting in a slightly broader first peak, i.e. earlier arriving sound, with an increased intensity. Somewhat curiously, the yellow foam slit exhibits a marked later peak which is not present in the grey foam case. One might have expected the black foam to have acted as an effectively hard surface, producing stronger mode reflections - however, this is not the case.

Conclusion

This work has demonstrated the ability of pulse techniques to provide useful information about the mechanics behind the passage of sound through a slit system in a wide barrier. Further, while the mode model has limitations, it appears to be a useful way of calculating the effects of more complex structures.

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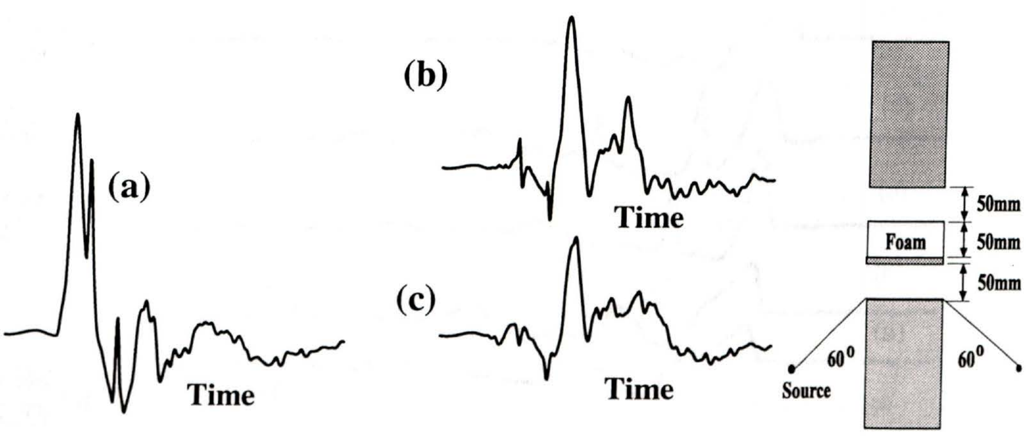

Fig.7: Waveforms recorded behind double slit system when one inside surface has foam lining.

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Noise and Vibration Analysis

Investigation of Transient Orthogonal Shaft Motion of Large Turbogenerators

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Abstract

Journal bearings are used extensively on large and small turbogenerator units in the process and power generation industries. The ability to predict the condition of the white metal bearing surface using non-invasive means such as measurement of the shaft motion in two orthogonal planes could lead to a reduction in maintenance costs due to unnecessary overhauls and could also improve the operational safety of the high speed rotating machines. The failure of the journal bearing surface can quickly lead to an increase in bearing temperature, shaft rubs, increased friction, degradation of the hydrodynamic lubrication regime and subsequently catastrophic failure. This paper presents an analysis of actual 120 MW shaft displacement probe data indicating the typical transient shaft motion which can occur, and an overview of a simplified dynamic model of a turbogenerator shaft predicting the shaft response to a transient forcing function.

Introduction

Orbit analysis is a vibration monitoring technique which has been in use for the last few decades based on the use of two proximity probes mounted in journal bearings to monitor low frequency displacement of the shaft with respect to the bearing housing [1-9]. The output signals from the probes are typically used to produce orbit plots representing machine shaft centre line motion which can be used as an indicator of general machine condition and various machine malfunctions [10-16]. For monitoring shaft motion in turbogenerators it is normal for two probes to be used orthogonal to each other to measure x-y or horizontal-vertical motion.

The residual unbalance of a turbogenerator rotor coupled with the differing stiffness and damping properties of the fluid filled bearing will generally give rise to an elliptical motion of the shaft centre in a plane normal to the rotation axis. A typical elliptical orbit motion in a rectangular coordinate or real-imaginary complex plane can be represented in the frequency domain by positive and negative frequency vectors [17, 18, 19]. If the Fast Fourier Transform (FFT) is used on the periodic form of the complex time motion over exactly one shaft revolution then the results from the FFT will provide the average amplitude and phase of the forwards and backwards whirl components which make up the elliptical shaft orbit motion. A lot of information about the shaft rotation can be determined from an orbit plot including unbalance, misalignment, resonance, fluid induced instabilities and preload, to name but a few [9-13].

The trending of orbit plots over time can provide a clear indication of changes in the running condition of the rotor system. A number of avenues exist for trending the results from orbit analysis and include trending the a) amplitudes of the positive and negative frequency components, b) orbit centreline position (eccentricity), c) maximum orbit amplitude from centre, d) phase of the maximum orbit amplitude and e) combined amplitudes of the high frequency shaft orders. These trend parameters will all be based on the averaged orbit plot obtained over a number of shaft revolutions to indicate the stable part of the shaft motion at the current period in time. The ability to obtain the average shaft orbit motion over a number of shaft revolutions is a particularly powerful technique which has been used extensively in gearbox vibration analysis [18], to reduce extraneous signal components which are not periodic with the shaft motion. The positive and negative frequency spectrum of the averaged orbit motion contains the same information as the orbit and also lends itself to automatic fault classification [20].

The FFT while providing the average forward and backward whirl components is unable to represent the transient nature of the orthogonal shaft motion within each revolution. The transient motion of the shaft will be the result of a number of forcing inputs such as steam pressure fluctuations and hydrodynamic pressure variations due to journal fluid film thicknesses and surface damage. To further analyse the transient shaft motion, a range of time-frequency analysis techniques can be used such as the Short Time Fourier Transform, Wavelet Transform or various forms of the Wigner-Ville distribution [21, 22]. It should also be possible to analyse the instantaneous orbit motion of the shaft rather than the average orbit to detect the presence of transient type behaviour of the rotor system. The focus of this paper is to outline current research methods aimed at investigating the link between the transient shaft motion and the presence of localised white metal damage. An analysis of transient shaft motion measured from a 120 MW turbogenerator with large fluid filled bearings is presented along with a simplified theoretical model of a turbogen-

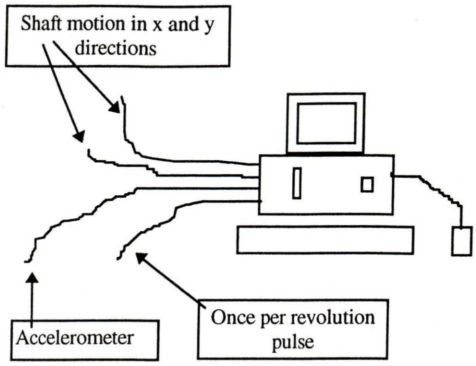

Figure 1: Schematic of the PC based data acquisition and signal processing system.

erator having a transient forcing function representing localised white metal damage.

Measurement and analysis of transient shaft motion

As part of an ongoing research program, a PC based data acquisition and signal processing system was developed using commercial hardware and software allowing on-line measurements of four channels of transient data to be recorded from a number of 120 MW turbogenerator bearings under typical load and speed operating conditions. A schematic diagram of the hardware is given in Figure 1 showing the four channels being digitised after signal conditioning and stored on the PC hard disc for later analysis. The data acquisition card chosen for the project was a National Instruments AT-A2150 providing 4 channels, simultaneously sampled on a single A/D card with automatic onboard anti-aliasing filtering protection.

A desktop 486 PC with the A/D card was taken to a number of 120 MW turbogenerators and four channels of data were digitised for each bearing, at 24kHz per channel, with direct memory access being used to write the continuous 16Mbyte data file to hard disc. The LABVIEW software from National Instruments was used to control the data acquisition, with all post-processing accomplished using the MATLAB software [18].

Figure 2 shows an orbit plot obtained over 1 single shaft revolution from a bearing of a 120 MW turbogenerator unit at Western Power. The resulting +ve and -ve frequency amplitude spectrum is shown in Figure 3.

The orbit shaft motion shown in Figure 2 illustrates the transient nature of the shaft motion over each revolution of the shaft and a number of high frequency transient events appear to be present. A large number of signal processing techniques can be used to analyse the transient shaft motion. One of the simplest ways of analysing the transient shaft motion is to plot the amplitude and phase components of the orbit as separate quantities

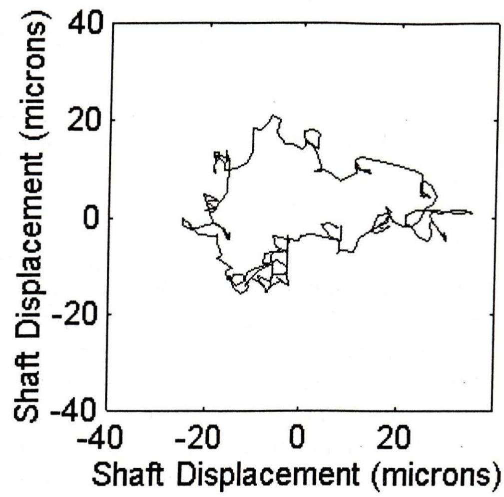

Figure 2: Orbit plot over 1 revolution with 256 points per revolution.

about the position (0, 0) of the orbit plot. Figure 4 shows the amplitude of shaft motion over 5 consecutive shaft revolutions and Figure 5 shows the corresponding phase. The shaft amplitude motion as seen in Figure 4 contains a significant amount of high frequency detail which is almost exactly repeated over consecutive revolutions.

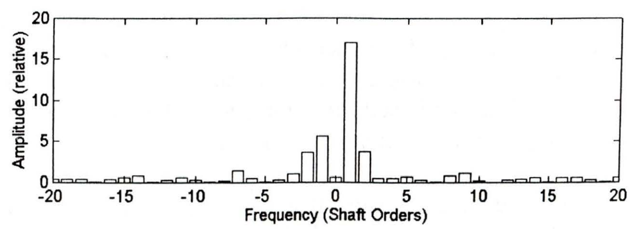

Figure 3: Amplitude spectrum of the complex orbit motion showing the +ve and -ve whirl frequency components

The high frequency shaft motion would be expected to contain the information regarding the quality of the white metal bearing surface as any resulting pressure fluctuations must affect shaft motion. Other factors such as non-homogeneity of the shaft material and shaft surface, non-uniform shaft loading, etc, will also be detected with the proximity probes. From Figure 5 it can also be noted that periodic rotational forcing of the rotor occurs once per revolution, possibly due to steam pressure fluctuations or due to the complex nature of the coupling of the bearing stiffness and damping in the x and y directions.

The major lesson obtained from the measured high frequency orthogonal proximity probe data is that a lot of transient shaft motion occurs, predominantly periodic every revolution, along with significant differences between revolutions. The shaft motion shown in Figures 2 - 5, were known to come from a journal bearing with significant white metal damage [23]. The difficult task of relating the transient shaft motion with damage in the white metal journal surface remains and is the subject of on-going data analysis and dynamic modelling research.

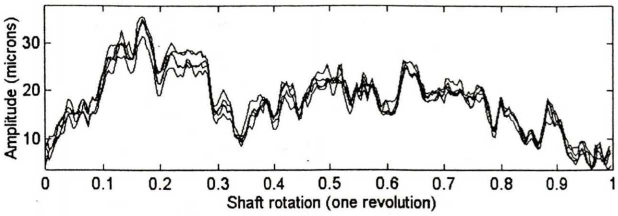

Figure 4: Shaft amplitude motion over 5 consecutive revolutions.

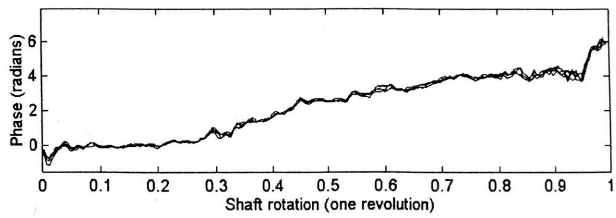

Figure 5: Shaft phase motion over 5 consecutive revolutions.

Simplified turbogenerator model

The lumped mass approach was chosen to create a simplified dynamic model of the continuous shaft and inertia of the rotor system. Coupled linear and cross journal bearing stiffness and damping were included and the major forcing function for the model was assumed to be some slight residual unbalance of the attached rotor. A transient pressure fluctuation induced forcing function was superimposed onto the unbalance forcing function to simulate the effect of journal pressure variations caused by localised white metal damage. When localised white metal damage occurs it would be expected that the pressure profile within the hydrodynamic lubrication regime of the journal bearing would be modified as the shaft moved closer to the damage region [23].

Figure 6 shows a simplified model which is currently being used to investigate the effect of the pressure fluctuations upon shaft motion where the rotor flexibility has been taken into consideration by including the shaft critical frequency of interest as an additional degree of freedom in both orthogonal directions [23].

The present model has included the effects of quadrature journal bearing stiffness and damping, transient pressure fluctuations, residual imbalance and rotor mass. The three lowest critical whirl frequencies of the actual rotor (21.7, 39.3 and 55.8 Hz) have been modelled separately by adjusting the natural frequency of the system to be equal to the shaft critical frequency of interest. The equations of motion derived from the model using Newtons second law,

$$\sum Forces = mass * acceleration$$

give for the x direction,

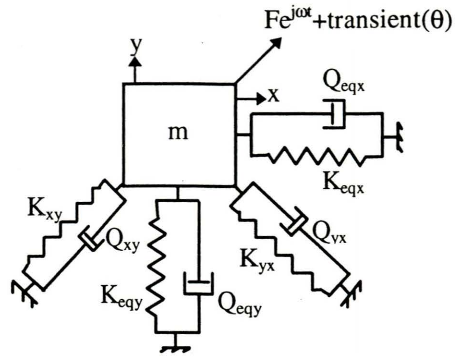

Figure 6: Simplified bearing model.

$$m\ddot{x} + Qeqx\,\dot{x} + Keqx\,x + Qxy\,\dot{y} + kxy\,y = Fx\,, \qquad (1)$$

and for the y direction,

$$m\ddot{y} + Qeqy\,\dot{y} + Keqy\,y + Qyx\,\dot{x} + kyx\,x = Fy\,, \qquad (2)$$

where F_x and F_y are the resulting unbalance and impulsive forces resolved into the x and y directions respectively.

The equations of motion can be rewritten in state space form so as to use standard numerical integration techniques for their solution. Introducing state variables, $v1 = x, v2 = y, v3 = \dot{v}1 = \dot{x}, v4 = \dot{v}2 = \dot{y}$,

gives, $\dot{v}3 = \ddot{x}$, and $\dot{v}4 = \ddot{y}$, which can be rewritten from equations (1) and (2) as,

$$\dot{v}3 = -v3(Qeqx/m) - v1(Keqx/m) - v4(Qxy/m) - v2(Kxy/m) + Fx \qquad (3)$$

and

$$\dot{v}4 = -v4(Qeqy/m) - v2(Keqy/m) - v3(Qyx/m) - v1(Kyx/m) + Fy \qquad (4)$$

The parameters K_{eqx} and K_{eqy} are regarded in the simplified model as being that equivalent rotor stiffness required to give the critical frequencies of the actual rotor system. The mass m was 2700kg, being the equivalent mass of the turbine/bearing stage. The direct equivalent stiffness and damping and quadrature stiffness and damping parameters can be developed from the bearing geometry and are functions of the eccentricity ratio, bearing geometry, oil viscosity, temperature, bearing radial clearance, etc [23]. To further simplify the model, values of quadrature stiffness and damping were chosen to give a converging shaft solution. To include the first three critical frequencies in the model, since the rotor mass remains the same, the equivalent rotor stiffness was changed so as to give the required natural frequency.

The forcing function used in the model was the superposition of the once per revolution unbalance and the impulse due to the transient pressure fluctuations as the shaft moved closer to the white metal damage. The force function was developed assuming an unbalance force of 2500N and a peak transient pressure pulse of 1000N due to the white metal damage. The force expressed analytically is given by

$$f(t) = F_{unbalance}\, e^{j\omega t} + F_{imp} \quad (5)$$

where the impulse described in stepwise form becomes

$$F_{imp} = \begin{cases} 0, & 0 \le \omega t < 5.06 \\ f_{imp} \sin(30(\omega t - 5.06)), & 5.06 \le \omega t \le 5.165 \\ 0, & 5.165 < \omega t < 2\pi \end{cases}$$

where f_{imp} is 1000N for this simulation.

Figure 7 shows a representation of the force function f(t) with the unbalance and pressure impulse combined.

The force impulse is positive because the pressure reduces in the presence of the white metal damage as the shaft moves closer to the damage [23]. The simulation of the simplified rotor-bearing system was accomplished with each critical frequency in turn using the MATLAB ode23 numerical solution of differential equations. The equivalent bearing stiffness in each of the x and y directions was initially chosen so as to correctly represent the critical frequency of interest with the cross stiffness values chosen so as to give a stable converging solution for shaft motion. Table 1 contains the resulting stiffness and damping values for the initial simulation of the shaft motion for the first critical frequency with an eccentricity of 0.7.

The shaft motion resulting from the unbalance force alone of 2500N gives a stable and repeatable shaft orbit as shown in Figure 8.

When the transient force impulse of 1000 N is applied in addition to the unbalance force, the shaft orbit motion becomes more erratic as shown in Figure 9. The pres-

	Direct stiffness (N/m)	Quadrature stiffness (N/m)	Direct damping (Ns/m)	Quadrature Damping (Ns/m)
x dir'n	5×10^7	5×10^6	7.35×10^3	7.35×10^2
y dir'n	5×10^7	5×10^6	7.35×10^3	7.35×10^2

Table 1. Direct and quadrature stiffness and damping values for the first critical shaft motion simulation.

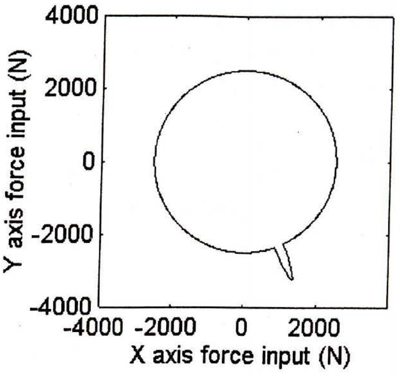

Figure 7: Forcing function f(t) due to the effects of unbalance and the transient pressure fluctuation.

ence of the once per revolution pressure induced transient force excites the resonance at 21.7 Hz which then decays with the damping until the next transient force occurs.

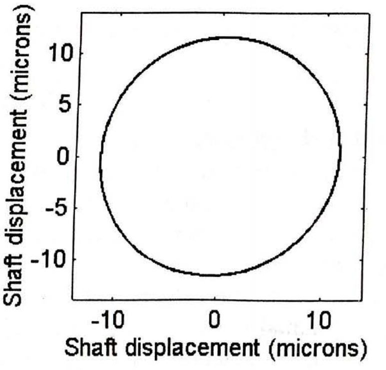

Figure 8: Stable shaft orbit motion over 50 revolutions due to an imbalance force of 2500 N.

Frequency analysis of the x and y shaft motion data confirms the presence of the 50 Hz forcing function and the first critical resonance at 21 Hz as shown in Figure 10.

Further analysis of the proximity probe shaft displacement data recorded from the turbogenerator bearing with significant white metal damage [23], is shown in Figure 11 where the power spectrum is shown upto 1000 Hz.

The power spectrum shows a broad frequency peak near 28 Hz. However, similar results were obtained from other journal bearings having no significant white metal damage. Rather than being an indication of resonance, the peak near half shaft speed is most likely to be from the presence of oil whirl. The interesting point to note from the above power spectrum is the large number of shaft related harmonics. This would appear to indicate a very complex forcing function, periodic with the shaft rotation.

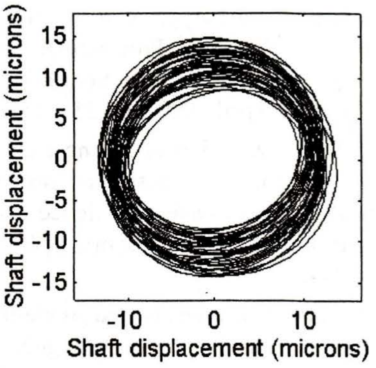

Figure 9: Shaft motion over 50 revolutions with unbalance and impulse forces applied.

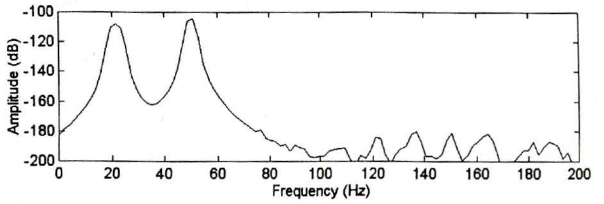

Figure 10: Power spectrum resulting from the simulation of shaft motion including the unbalance and transient pressure induced forcing function.

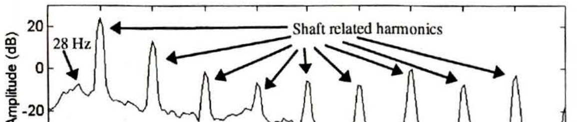

Figure 11: Power spectrum obtained from proximity probe data measured from a bearing with significant white metal damage.

Discussion and conclusions

The vibration signal processing and simplified dynamic modelling presented in this paper should be seen as an initial attempt at understanding the complex area of rotor dynamics. The signal processing of the measured proximity probe data from damaged white metal bearings has shown considerable amount of high frequency transient shaft motion. It is recognised that this motion can be caused by a number of non-bearing related phenomenon such as the non-homogeneity of the shaft material and shaft surface, non-uniform shaft loading effects, steam pressure fluctuations, etc. The high frequency transient motion has been detected in both damaged and undamaged journal bearings and the standard time and frequency analysis techniques have been unable to detect appreciable differences between the different states of damage.

The simplified dynamic model of the turbogenerator system is known to have several deficiencies. The direct and quadrature stiffness and damping values in both directions have been assumed to be the same which is clearly incorrect, particularly as the actual bearings were known to be elliptical in shape. The transient pressure induced force pulse has been assumed to occur as a reduction of pressure when the shaft moves closer to the journal damage. This hypothesis has as yet not been experimentally confirmed [23]. The stable unbalance force is known to be an oversimplification as there are clearly many higher harmonics of shaft motion observed in any measured orbit or spectrum analysis. These could be included in any more complex simulation if required.

The model has shown that system resonances can appear in the shaft orbit motion in the presence of transient forcing functions. The resulting motion will appear to be more erratic than the regular orbit motion with no transient force applied. The irregular once per revolution shaft motion does actually occur in practice and can easily cover 5-10μm as the simulation shows.

This research program is continuing experimentally and theoretically to try and identify the effects of white metal damage in the journal surface. A flexible rotor system with journal bearings is being designed which should allow the study of journal pressure profiles in the presence of white metal damage and the measurement of x and y shaft motion under controlled laboratory conditions. A more sophisticated dynamic model of a flexible rotor is also being developed and will include the model of the continuous shaft-rotor-shaft and journal bearing direct and quadrature stiffness and damping.

Acknowledgments

The financial and engineering support from Western Power, which made this project possible, is greatly appreciated.

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A Study Of Impact Response Of Ship Hull Using The Finite Element Method And Scaled Experimental Model

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Introduction

The finite element method is now increasingly used in the analysis and design of complex ship structures[1]. A ship has to be analysed not only as a non-uniform rigid body but also as a flexible structure taking hydroelasticity into consideration[2]. The analysis of rigid body motion will determine the stability of the ship, while the study of resonant response of the flexible ship structure may lead to the prediction of fatigue cracking followed by ultimate failure. Undesired vibration and noise radiation are also important issues associated with the dynamic response of the ship structure[3].

Most of the previous research was concentrating on the global dynamics of the ship and little work was found on the local distortions of the hull. If a ship is to survive side impacts or local slamming, the local response of ship hull needs to be analysed in detail[4]. Often the local response of a ship can be described in terms of a sub-structure subject to an external loading. Any interaction between the sub-structure and other part of ship structure will be included as boundary conditions. The characteristics of the sub-structure is described by the natural frequencies and mode shapes. Frequency response and transient response of the structure to an impact are often used as measures of the structural dynamics.

The prediction of structural response of a ship hull to point impact forces is the concern of this work. Because of the complexity of the sub-structures and of the impact force, structural engineers usually resort to the finite element packages seeking a solution to the problem. However, uncertainty in the predicted results may rise when finite element programs are used for the dynamics of complex structures. This is due to the following:

1) The assessment of the accuracy of the FE results is difficult because no other theoretical method can be used as a benchmark test of the complicated structures;
2) No quantitative guidelines are available to describe the structural uncertainty and to show the error in prediction when the structure and structural boundary conditions are idealised in the pre-process of the modelling ;
3) The exact damping information of the structure is not available;
4) Practical hydrodynamic loading is often provided in a statistical sense.

This work is dedicated to show the possible uncertainties in the prediction of structural response when FE models are used. Firstly, quantities of interest, such as resonance frequencies, frequency response and impulse response of a sub-structure model of a ship hull, are obtained using various FE models. Then experimental model is used to provide measured values for comparison. The analysis of the difference in the results between the FE model predictions and experimental measurements will provide an understanding of the possible uncertainties. This work suggests that a good understanding of the underlying physics and a feeling of the expected results will always be useful before the FE analysis is applied to a new complex structure. Also illustrated in this work are the role of the structural components in the transmission/attenuation of vibrational energy and the effect of the impact positions on the response of the structure.

Experimental test rig and procedure

To provide an experimental comparison, a scaled middle section of a bulk carrier ship hull was constructed. Structural response to the impacts on the side of the section will be the focusing area of investigation. The test structure shown in Figure 1 is a 1:50 scale model representing an actual bulk carrier. The model is constructed from mild steel plating because steel is the most commonly used material for large ships. The model has the whole cross section because the boundary conditions for the symmetry line are difficult to produce in practice. The skin of the hull model is a 0.6 mm plate with the transverse and longitudinal ribs constructed from 0.8 mm plates. The keel plate (1.6mm thickness) is connected to the transverse ribs by riveted angle sections. All the ribs are soldered to the hull since any form of welding would buckle the plating. Soldering was also used by other researchers[4] on their test rigs and favourable results were achieved.

As freely-supported conditions are commonly used[5], the test structure is suspended from the ceiling to provide a free dry hull. The dry hull response is a good representation of the overall response of the hull when it is in a fluid environment[3]. The natural frequencies of the free rigid body motion of the test model were found to be less than 1 Hz. The hull was impacted with an impact

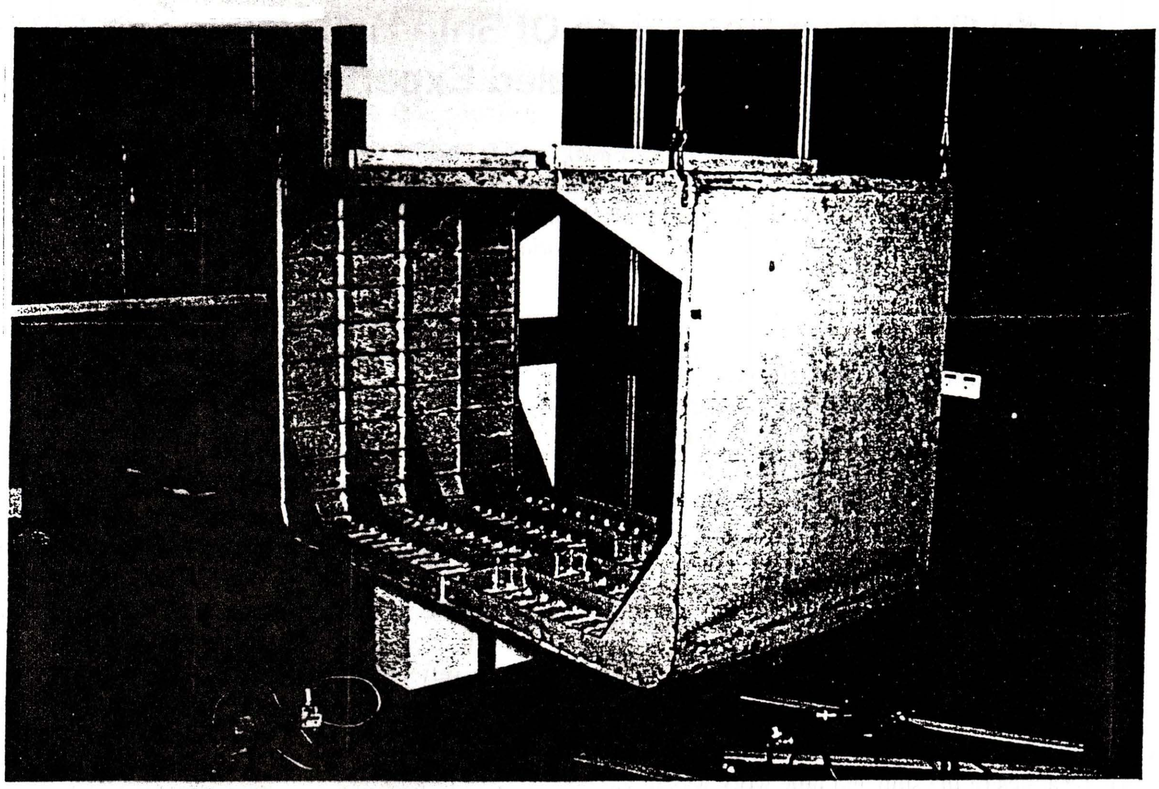

Figure 1: Experimental test model

hammer fitted with a force transducer. The hammer fitted with a plastic tip weighs 284.1 grams and can give force in the range of 100 N to 700 N. The estimated impact time is about 1.5 to 5 ms and the frequency spectrum of the force ranges from 0 to 2000 Hz.

The response was measured by a light weight accelerometer(2.4 grams). The accelerometer was fixed to the hull plating by bees wax. Charge amplifiers were used to amplify the force and acceleration signals and to filter out all frequency components below 1 Hz and above 1000 Hz. The two signals are then fed into a two channel frequency analyser for the qualities of interest.

Finite element models

Three FE models with different degrees of details and approximations were investigated. The aim was to compare the results from the three models against each other and against the measured results of the test model, and to assess the feasibility of the FE models for the dynamic response of ship hull structures. The three FE models are representing the experimental test hull.

Shown in Figure 2, model 1 consists of a series of plate elements joined together to represent the shape of the hull. The equivalent plate thicknesses were calculated so that the bending stiffness of the element is equivalent to the ribbed hull. This equivalent thickness had to be calculated in both longitudinal and transverse directions. For the structure considered here, the equivalent thicknesses turned out to be different in these two directions

For the isotropic plate elements used in this model, the larger thickness was selected for both directions. Consequently, the bending stiffness of the plate elements is equivalent to the stiffness of the test model about the X axis, and the stiffness of the elements about the Y (or Z) axis is higher than that of the real structure. The material density of the elements was adjusted to give equal stiffness to mass ratio as the test model.

Model 2 (Figure 3) is a more detailed model where the transverse ribs are represented with equivalent beams and the skin and longitudinal stiffeners are replaced by equivalent plate elements following the same procedure as for model 1. This simplification leads to a higher bending stiffness about the X axis than that of the test model. For the model 3 shown in Figure 4, the mesh density was increased to a level that computing cost is still reasonable. It enables more realistic mode shapes and accurate transient impact response.

An averaged damping of the test structure was measured and inserted into the FE models as a constant structural damping coefficient. In this study, the action of impact hammer is described as a point impact force(100 N, for 2 ms). To investigate the effect of structural components (skin, ribs and stiffeners) in the dynamic behaviour of the hull, the frequency response of the structure was obtained for the frequency range 1:100 Hz. The impact response covered a one-second period. The impact positions selected are located at the lower and the middle parts of the side of the hull and denoted as a and b in

Figure 2. The responses were examined at 4 position shown in Figure 2. The excitation at transverse ribs and longitudinal stiffeners was also examined to investigate the effect of these structural components on the dynamic behaviour of the hull.

Results And Discussions

Structural characteristics

The structural characteristics are described by the mode shapes and corresponding resonance frequencies. The mode shapes obtained from the FE analysis are depicted in Figure 5. The predicted and measured resonance frequencies (up to 100 Hz) of these modes are included in Figure 5. The mode shapes of the test structure corresponding to the measured resonance frequencies were also checked experimentally. By comparing the resonance frequencies, the following may be summarised:

1) Although model 1 is consisted of only 384 plate elements, the resonance frequencies were predicted at a reasonable accuracy. This indicates that the simple model can be used in the early design stages while the more detailed model can be used in the fi-

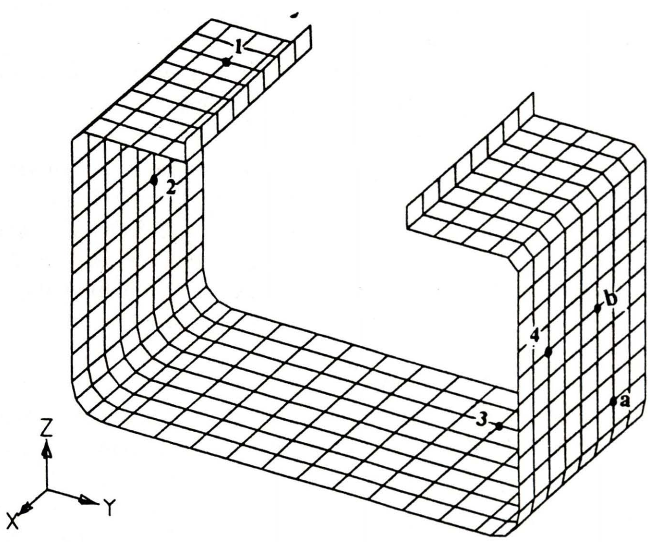

Figure 2: Finite element model 1 - 'equivalent thickness'.

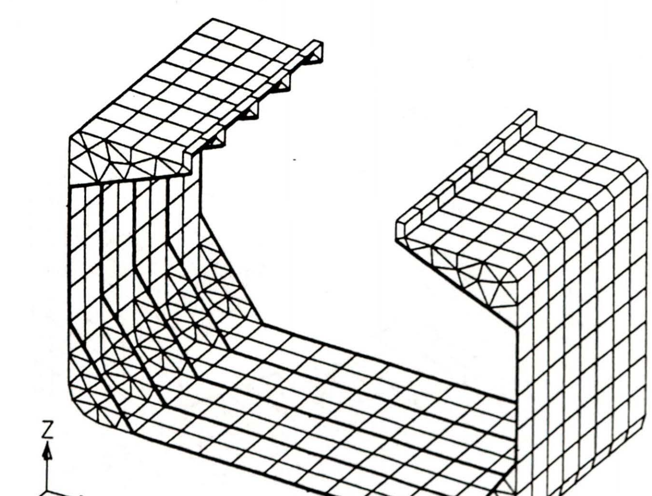

Figure 3: Finite element model 2 - 'course mesh'

nal design stages when accurate and detailed response is needed. One of the merits of the equivalent thickness model (model 1), which over weighs its reduced accuracy, is that it allows a quick look at the mode shapes and a check of mesh size for the frequency range required. The rule of thumb is that five elements in each half wave length should be considered so that the corresponding mode can contribute reasonable accuracy to the response.

2) It seems that the FE models tend to give overestimated resonance frequencies.

3) The discrepancy of the resonance frequencies by model 1 and model 2 from that of model 3 is mainly due to the fact that the bending stiffnesses about Y (or Z) axis for model 1 and about X axis for model 2 are higher than the corresponding ones in model 3 which is close to the test model. Any mode with highly curved shape in a certain direction, its resonance frequency is very sensitive to any inaccuracy of the stiffness modelling in this direction(eg. mode 4 and mode 6 for model 2 and mode 2 for model 1).

Structural response

As Shown In Figure 2, The Structural Response Were measured at 4 locations respectively(main deck, non-excited side, bottom plating and excited side). Impact force was applied either at location a or b as shown in Figure 2. The frequency response from FE calculation and measurement are shown in Figures 6 and 7. We can observe the following:

1) In general, FE models provide under-estimated frequency response;

2) The measured frequency response shows that the damping of structural modes is different from one mode to the other. For example, the mode at 66 Hz is very lightly damped resulting in a very high response as compared to the FE predictions. This result shows that even though the recommended or measured values of averaged damping are used, the predicted response will still deviate from the measured response.

The predicted maximum overshoot of the structural accelerations response to an impact is very sensitive to the mathematical representation of the time history of the impact process. For the FE prediction, a mathematical representation of the measured force from the impact hammer was used so that experimental results can be used for comparison. The maximum overshoots of the structural acceleration are presented in Table 1. The experimental values are consistently higher than the FE predictions.

The acceleration responses on the deck (position 1) and the side (position 2) to an impact at location a are plotted in Figure 8. It is observed that the lower frequency modes have greater contribution to the impact response of the deck and that the responses decay at different rates.

In summary, the FE models can predict most of the resonance frequencies and mode shapes of the test model at reasonable accuracy. However, the simplified FE model is not able to predict a few modes which exist in the test model and are predicted by the other FE models. This is largely due to the fact that the bending stiffness about the Y (or Z) axis for model 1 and about X axis for model 2 is much larger than that of the test

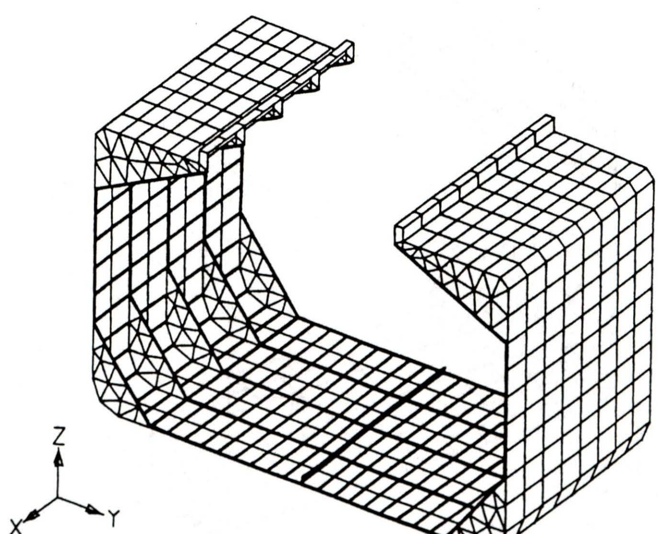

Figure 4: Finite element model 3 - 'fine mesh'.

model. In general, all the present FE models tend to overestimate the resonance frequencies of the test model. It is understood that stiffness of the three models were higher by different degree than that of the test model.

Using averaged structural damping from the test model and measured impact force, the transient response of the structural acceleration can be predicted using the FE models. In general, the predicted maximum overshoots of the response are all lower than the that from the measurement. The relative difference between predicted and measured values may vary from 20% to 90%.

Experimental investigation of the effect of the impact positions on the structural response indicates that: an impact at a transverse rib is usually associated with low vibrational levels in the local area of the impact and high vibrational level over the entire hull, while an impact between the transverse ribs produces highly localised vibrations and lower level of vibration on the rest of the hull. It is believed that this will also be the case for the impact response of full scale vessels because impacts on the highly stiffened areas will lead to minimum energy dissipation in vicinity of the impact and the energy imparted to the structure will be carried by the stiff frames to the remote areas. On the other hand, the impact response in the area between the transverse ribs tends to be localised, because the transverse ribs behave as rigid boundaries so that the impact energy is confined and attenuated locally.

Conclusions

Accurate prediction of the dynamic response of ship structures using FE models can be a difficult task. The accuracy of the results are usually governed by the

1) accuracy of structural idealisation;
2) accurate finite element representation of real structures;
3) refinement of the FE mesh;
4) values of damping;
5) accuracy of the impact force model.

Although a good accuracy may be attained in assessing the resonance frequencies and mode shapes, the levels of accelerations and consequently stresses and strains have to be interpreted in view of the above mentioned factors.

Experimental investigations revealed that the transverse ribs play the main role in transmitting the vibrations due to impacts through the hull structure. Impacts on the positions of transverse and longitudinal ribs produce higher vibrational levels as compared to impacts on the plate panels between stiffeners. Finally, as the present study is conducted on a dry hull, further research including the hydrodynamic loading is recommended.

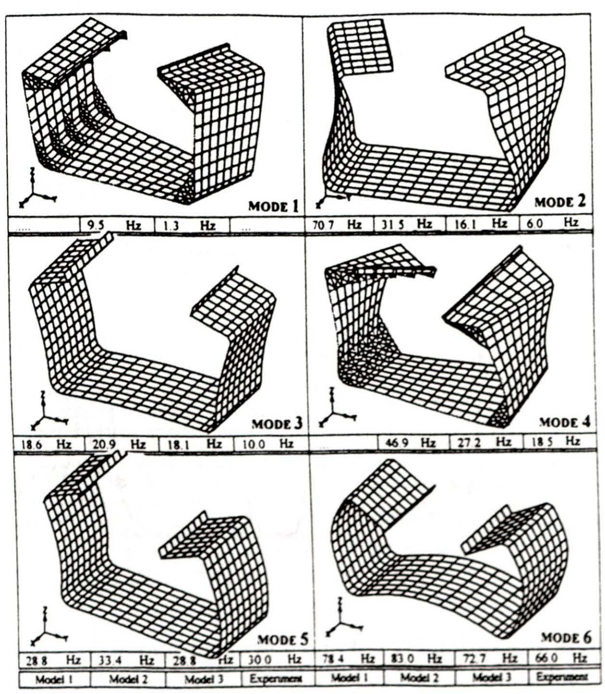

Figure 5: Mode shapes and resonance frequencies from F.E. models and test model.

Table 1. Maximum accelerations (m/s/s) predicted by F.E.models and measured experimentally.

	Impact on position (a)				Impact on Position (b)			
Response position	**1**	**2**	**3**	**4**	**1**	**2**	**3**	**4**
F.E. Model 1	12	20	11	22	28	44	25	55
F.E. Model 2	13	11	15	12	38	36	32	33
F.E. Model 3	15	38	24	29	42	64	42	42
Experimental	19	105	142	107	23	143	150	93

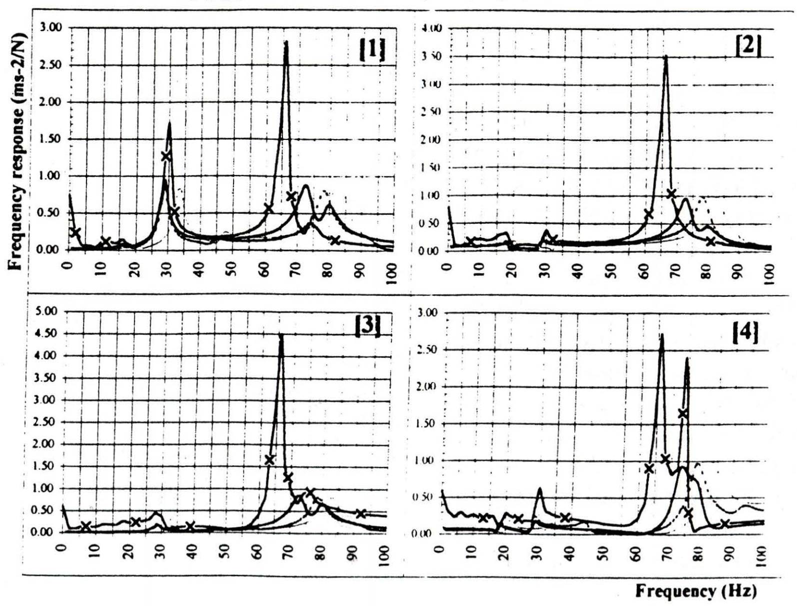

Figure 6: Frequency response at positions 1, 2, 3 and 4 to excitation at 'a'. dotted line: model 1; dashed line: model 2; solid line: model 3; solid line with crosses: experiment.

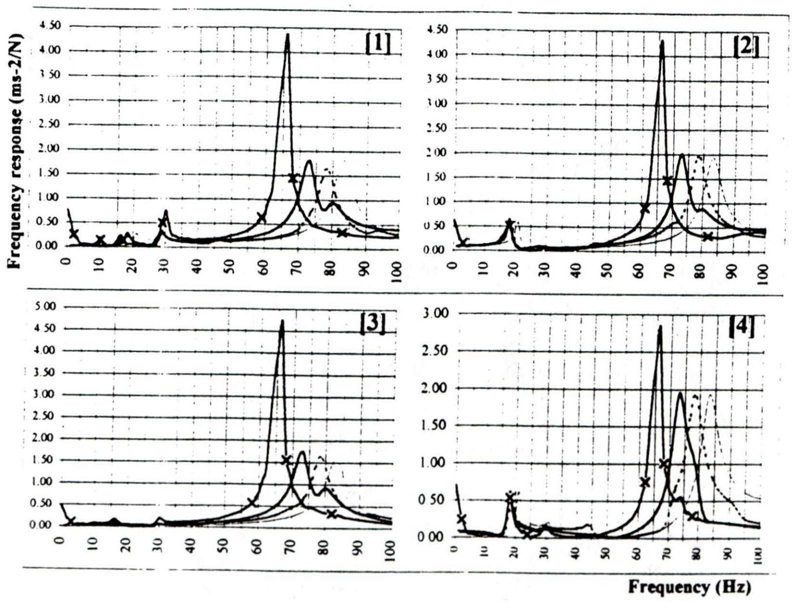

Figure Figure 7: Frequency response at positions 1, 2, 3 and 4 to excitation at 'b'. dotted line: model 1; dashed line: model 2; solid line: model 3; solid line with crosses: experiment.

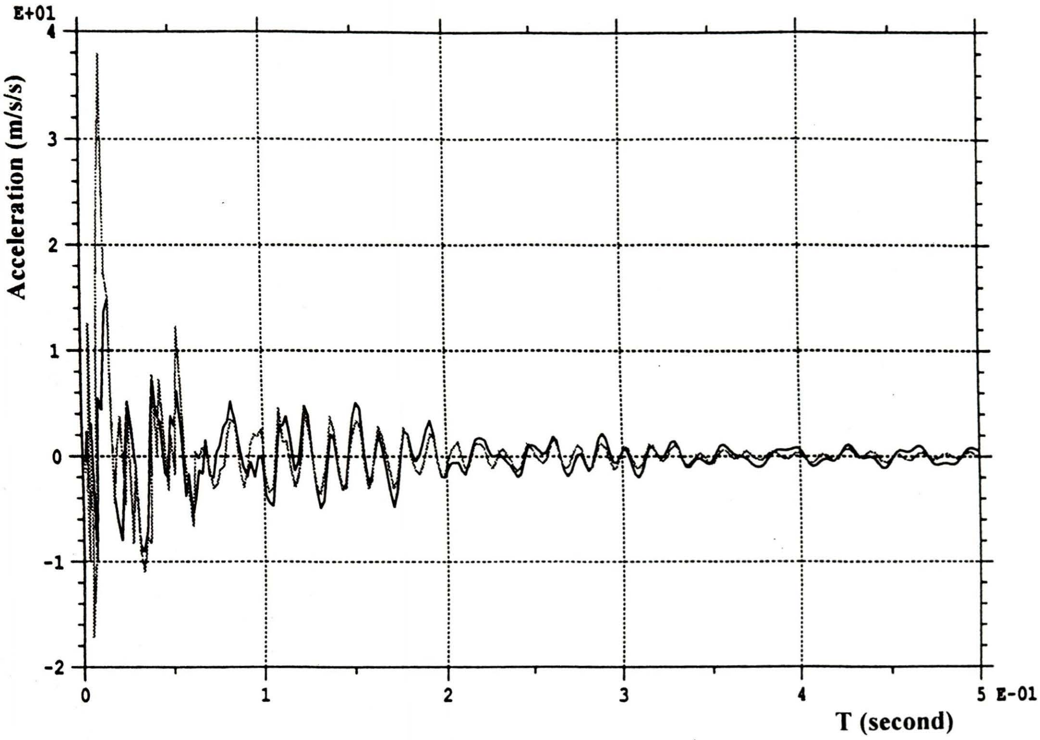

Figure 8. Response at positions 1,-------- 2, ... and impact at 'a'

Acknowledgment

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Session 7B

Environmental Noise

Bells - Sound Or Noise?

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Abstract

Conflicting views became apparent in the quiet Perth suburb of Peppermint Grove, when St Hilda's Anglican School for Girls proposed to install a set of eight bells in the new Chapel. Ringing the bells would clearly exceed noise regulations, and the community was divided on the issue. There were also at least four other proposals to install sets of bells in other locations around the State. The Department of Environmental Protection became involved in the issue and decided to conduct a community survey to identify a bellringing schedule which would satisfy the great majority of residents around the school. As part of this process, detailed measurements of sound levels of other bells were carried out and predictions made of the likely noise emissions from the Chapel bell-tower. This paper outlines the sound level measurements and predictions and the survey methodology and results. The development of a Ministerial exemption to allow limited bellringing is also discussed.

Introduction

Sound versus noise

Sound may be described as everything we can hear, while *noise* refers only to those sounds which are unwanted.

In the case of *Haddon v. Lynch* (1911), the plaintiff complained of the "unmelodious sound of the church bell of St Paul's Church of England, Malvern at seven-thirty and eight on a Sunday morning", while a neighbour for the defendant claimed that "The bell ought to ring. Whoever opposes it is no man."

While the sound of bells may be pleasant to many people ("sound"), it may be regarded as a nuisance by others ("noise"). The sound levels from bells, when measured in a residential area, are likely to exceed statutory requirements. A set of sound level measurements however, cannot tell us whether the ringing of bells will be perceived as sound or noise. Thus it is important that the perceptions of local residents be considered in determining appropriate times for bellringing.

Bell Ringing

Bell ringing is an ancient practice and bells of cast bronze have been associated with English churches since at least the seventh century[1]. Bells were used as a call to worship, to signify the time, to mark celebrations and sometimes in emergencies. The practice of "change ringing", in which a team of six or eight people ring a set of tuned bells in sequence, has been common in the United Kingdom for centuries but has not generally been adopted in Europe or elsewhere. Bells are normally rung in "touches" of 10 to 15 minutes' duration, although on occasion they may be rung in a "peal" of up to three hours' duration.

In Western Australia, the earliest set of bells was the St George's Cathedral bells, installed in 1902. The only other significant installations were at Holy Trinity Church, York and Christ Church, Claremont, both in the late 1980s[2]. The Australian and New Zealand Association of Bellringers have advised of several other proposals to install sets of bells in churches or civic buildings in this State[3].

St Hilda's Project

Approval for construction of a chapel at St Hilda's School was granted by the Town of Mosman Park in January 1993. Although the chapel incorporated a bell-tower, the approval specifically excluded the installation of bells. In August 1994, St Hilda's applied for approval to install a set of eight bells in the tower, the basis of the application being that the tower was designed to be fully enclosed for practice sessions and would only be opened for ringing on certain defined occasions.

Council referred the application to the Environmental Protection Authority (EPA) for assessment under Part IV of the Environmental Protection Act. Because of the unusual nature of the issue, it was decided that the Department of Environmental Protection should attempt to achieve a negotiated solution. As a result, the author prepared the detailed assessment report on which this paper in based[4].

Assessment of sound levels

Sound generation

Bells are normally cast in bronze, consisting of 77% copper and 23% tin. The sound produced by a bell is a complex series of harmonics radiated from the various parts of the bell. The bell is tuned by varying its thickness, a thinner bell having a deeper note than a thicker bell of the same size. A set of bells is normally tuned so that their strike notes form a major scale, with the "tenor" bell as the tonic (lowest) note and the "treble" as the highest note.

A set of bells used for change ringing is supported in the belltower with the open end facing upwards and the clapper resting against the side of the bell. When the rope is pulled, the bell swings down and then on the reverse upwards swing, the clapper strikes the rim ("soundbow") of the bell, producing a tone. This technique produces a consistent sound level each time the bell is rung.

Sound Measurements

Sound level measurements were conducted at the Holy Trinity Church in York and at St George's Cathedral in Perth to provide a basis for sound level predictions for the St Hilda's bells. Tests were conducted both inside the towers adjacent to the bells and outside at distances up to typically 100 metres. The measurements were conducted by the author and Mr Richard Langford of the Department of Environmental Protection, accompanied by acoustical consultants Herring Storer Acoustics in the case of the York tests. All results are quoted as $L_{Aeq,T}$ levels, in units of decibels, or dB(A). $L_{Aeq,T}$ is the "average" sound level, taken over time T; in this case the averages were taken over periods of typically 1 to 3 minutes, while bells were being rung continuously.

The results of the measurements inside the two belltowers indicated typical sound levels of 109 dB(A) at Holy Trinity Church, York and 116 dB(A) at St George's Cathedral. This difference is in part explained by the measurement position being closer to the bells at St George's and in part by the greater weight of the St George's bells. As the weights of the St Hilda's bells are generally in between those of St George's and Holy Trinity, it was considered relevant to use these as the basis for sound level predictions for St Hilda's.

The measurements outside the two belltowers illustrate the effect of openings in the tower structure. At Holy Trinity, the tower had openings in the wall in the reverberation chamber above the bells and folding doors in the roof, which were open at the time of the tests. The small windows adjacent to the bells had been blocked in with perspex. The sound level at 50 metres away was 68 dB(A) and at 100 metres was 60 dB(A).

The St George's tower has an open roof with a canopy surrounded by a parapet wall. The openings in the eastern wall were open at the time of testing, while those on the western side were closed. The sound level 40 metres away on the eastern (open) side was 74 dB(A) and on the western (closed) side was 67 dB(A). These results indicate that the majority of sound at 40 metres on the eastern side is from the open wall, the reduction in sound level from inside to outside being 42 dB(A). This result is comparable with Holy Trinity, where the sound level reduction from inside to 50 metres outside was 41 dB(A). With the enclosed wall, the reduction from inside to outside at St George's was 49 dB(A) and the sound from the open roof tended to predominate.

These results were used as the basis for a series of sound level predictions for the St Hilda's Chapel, as described below.

Sound Level Predictions

Sound level predictions were carried out for the St Hilda's Chapel by acoustical consultants Herring Storer Acoustics (HSA), using the sound level data outlined above. They used the well-accepted Environmental Noise Model (ENM) to predict sound levels emanating from the Chapel, taking into account the structure of the tower, with its solid walls, sealed windows and open roof incorporating acoustic doors and canopy.

The model predicts sound levels over the surrounding area, taking into account distance from the source, topography and other factors which affect sound propagation. The sound level predictions were only carried out for calm conditions. The results were plotted as a series of sound level contours, as shown in Figure 1.

These contours show the nearest residences about 50 metres away receiving sound levels of approximately 60 dB(A). As expected, this result is consistent with the result for the western (enclosed) side of the tower at St George's Cathedral. The predicted levels drop to 40 dB(A) at a distance of about 600 metres from the Chapel, except on the southern side where the rising ground provides a screening effect and the contour is closer to the Chapel.

Community survey

A survey of community attitudes towards bell-ringing was conducted in December 1994, by means of a questionnaire delivered to all properties in the study area. This was not intended as a rigorous social survey, but rather as a series of specific questions to gauge community views.

Study area

The study area was defined as all those properties within or touching the 40 dB(A) contour as predicted by Herring Storer Acoustics. This level was selected as representing those residences where the sound of the bells would be generally audible. A total of 566 properties was identified as within or touching this contour level and this was defined as the study area.

Questionnaire

A one-page questionnaire was delivered to each property in the study area on the weekend of 10 to 11 December 1994, with a response requested by 20 December. The sound level contour map (Figure 1) was incorporated on the back of the question sheet to assist respondents in understanding that the sound of the bells would be audible at their premises and to allow them to indicate what range of sound levels they were within. It was accepted that many residents might not understand the meaning of the map.

TABLE 1: "Yes/No" Responses by Sound Level Range

Sound Level Range dB(A)	"Yes"	"No"	TOTAL
40 - 44	98 (82%)	22 (18%)	120
45 - 49	41 (82%)	9 (18%)	50
50-54	19 (70%)	8 (30%)	27
55 - 59	11 (55%)	9 (45%)	20
60 - 64	3 (75%)	1 (25%)	4
No indication	12	4	16
TOTAL	184 (78%)	53 (22%)	237

Response Rate

A total of 238 responses was received up until 31 January 1995 and all but one were included in the analysis. The exclusion was the response from the school itself, which would possibly have skewed the results for the very small group of responses in the range 60 to 64 dB(A). The 238 responses represent 42% of possible responses, which indicates a high level of interest in this issue. Analysis of the response rate against sound level shows the response rate rising with sound level (34% response in the range 40-44 dB(A) rising to 100% in the range 60-64 dB(A)), indicating those closest are more concerned about the issue.

Yes/No Responses (Question 1)

Question 1 simply asked whether or not people agreed with "open" ringing of the St Hilda's bells and requested them to tick either: "Yes, at reasonable times" or "No, not at any time". Overall, 184 (78%) said "Yes" and 53 (22%) said "No". Table 1 gives a breakdown of these figures by sound level.

Table 1 shows a consistent response across the 40 to 49 dB(A) range. The "No" response increases sharply from 18% to 30% then 45% over the range 45 to 59 dB(A). It then drops to 25% in the highest range. This may be explained either by the small sample size or by the fact that these four respondents are the School's closest neighbours and therefore have a different relationship or expectations regarding the School.

However one interprets Table 1, the result indicates significant opposition to open ringing, especially from the residents in the ranges above 50 dB(A). The 18 "No" responses in the ranges above 50 dB(A) represent 22% of the 81 possible responses from this area. With this result in mind, it is relevant to now consider the general comments in Question 6 of the questionnaire.

General Comments/Reasons for Answers Given (Question 6)

Question 6 asked respondents to "Please explain your reasons for the answers you have given." All respondents were asked to complete this question, regardless of whether they had answered "Yes" or "No" in Question 1. The vast majority of respondents took the opportunity to make comment and, as expected, their answers embraced a wide range of views. To facilitate analysis, the comments were grouped under the following headings:

"No objection"	No objection to bell ringing, in some cases provided it was done at reasonable times.
"Like bells"	Includes expressions such as "bells are beautiful", "we like bells".
"Heritage"	Includes comments to the effect that bells are part of a tradition, or English/Christian heritage.
"Enhance community"	Includes responses to the effect that the school and its Chapel are part of the community, or the sound of bells is preferable to other noises such as traffic.
"Intrusive"	Includes responses to the effect that the sound of bells will intrude on the peace and quiet of the area, the area is already too noisy, ringing outside of restricted times will be intrusive.
"Need for rest"	Bells would disturb rest for children/elderly/sick people.
"Dislike bells"	Includes expressions such as "we don't like bells", "we don't want to hear them".
"Unfair to neighbours"	Some respondents commented that, while they liked bells,

they lived far enough away for it not to be a problem, and it may be unfair to others living closer.

"Not needed" — Some noted that the bells were not needed, as other schools did not have bells and most of the occasions would be for St Hilda's only.

"Enforcement" — Responses such as "Give an inch, take a mile" were classed under the issue of whether ringing restrictions would be adhered to.

"Trial" — Includes calls for a trial period or trial ringing.

"Other institutions" — Raised the issue of whether the bells would set a precedent whereby other religions or institutions would also be allowed to make excessive noise.

"Query noise" — Some responses queried the meaning of the sound levels provided with the questionnaire, some noting that they found it difficult to comment without understanding the sound levels.

"Too loud/close" — Includes comment to the effect that the noise would be too loud and/or the respondents lived too close.

The results are presented in summary form in Table 2, broken down into two sound level ranges: above 50 dB(A) and below 50 dB(A).

The most common response was "Like bells", which appeared 74 times (31% of all responses). This view strongly outweighted the "Dislike bells" comment (3% of all responses) and was reflected in the "Enhance community" response which appeared 35 times (15% of all responses) . This positive comment is significant in that a decision not to allow any open ringing would disappoint a substantial proportion of the population which would like to hear the bells and feel they would add character to the area.

The second most common response was "Intrusive", which appeared on many of the "No" responses and also on a number of the "Yes" responses. This was supported by the number of comments on "Need for rest", "Unfair to close neighbours" and "Too loud/close" (42 altogether).

Finally, "Query noise" and "Trial" were mentioned 31

TABLE 2: Summary of General Comments

Comment	No. of Responses			
	Sound Level 50 - 64 dB(A)	Sound Level 40 - 49 dB(A)	Sound Level Not Indicated	TOTAL
"No objection"	6	23	1	30
"Like bells"	8	61	5	74
"Heritage"	2	9	2	13
"Enhance community"	7	26	2	35
"Intrusive"	15	35	5	55
"Need for rest"	3	7	1	11
"Dislike bells"	3	5	0	8
"Unfair to neighbours"	4	11	0	15
"Not needed"	3	3	0	6
"Enforcement"	1	2	1	4
"Trial"	11	2	0	13
"Other institutions"	1	5	1	7
"Query noise"	10	7	1	18
"Too loud/close"	7	8	1	16

times, indicating the need for a trial period should open ringing be allowed. Assuming such a trial went ahead, the answers to Questions 2 through 5, (which those who answered "Yes" to Question 1 were asked to complete) provided a basis for developing a possible set of operating conditions.

Preferred Occasions for Open Ringing During the Day (Question 2)

Question 2 asked those who had answered "Yes" to Question 1: "What would be the best number of occasions for open ringing of the bells?" The notes to the question specified that the question related to ringing between 8.00 am and 7.00 pm; that each "occasion" meant one peal or "touch" of 10 to 15 minutes' duration; and the maximum duration of open ringing in one day would be 30 minutes. Respondents were asked to nominate a preferred number of occasions in any month by circling one of the numbers 5, 10, 15, 20, 25 or 30, or indicating a number of their choice. Most respondents circled one of the numbers, however a few (7) indicated numbers below 5 and a larger group indicated numbers above 30 or "No limit". The results are presented in Table 3.

The results are presented in two sound level ranges, above 50 dB(A) and below 50 dB(A), the results within these ranges having been found to be fairly consistent.

For the respondents in the sound level range above 50 dB(A), the strongest indication was for 5 occasions per month or less, supported by one third of respondents in this group. Almost half of this group opted for 10 occasions or less. Eight responses (24%) opted for 30 or more occasions. The responses from the group in the range below 50 dB(A) were similarly spread, with 35% opting for 10 occasions or less and 40% opting for 30 occasions or more. Once again, a range of views is apparent, from those who would happily hear the bells daily to those who would be concerned at more than a few occasions per month.

Other restrictions on open ringing during the day (Question 3)

Question 3 was an open question related to Question 2, asking: "Are there any other restrictions which should be applied?" Of the 184 "Yes" responses to Question 1, 107 (58%) responded to Question 3. Of these, 73 (68%) indicated that no further restriction was needed beyond that in Question 2. Of those who called for further restrictions, 16 mentioned ringing times, 10 mentioned types of occasions, 10 mentioned ringing duration and 2 mentioned sound level.

Open ringing during the evening (Question 4)

Question 4 asked the 184 respondents who had answered "Yes" to Question 1 to nominate times bells should be rung with the tower open outside the hours 8 am to 7 pm by nominating one or more of four options. The response forms contained a total of 236 answers to Question 4, some having ticked more than one box (which is a valid response).

There was considerable support for some ringing in the evening, especially for "Christmas/New Year" (114 responses) and "4 other evenings" (72 responses). In the open part of Question 4 ("Other"), 33 responses were received, and of these 17 said "No restriction". A further 17 responses ticked "Never" to open ringing during the evening.

Ringing one bell before and after school (Question 5)

In Question 5 those who had answered "Yes" to Question 1 were asked: "Would you also agree to one bell being rung for less than one minute at the start and end of each school day?"

The "Yes" response outweighed the "No" response with 76% of the responses to Question 5. As expected, the "Yes" response was weaker in the 50 - 64 dB(A) range (67%) than in the 40 - 49 dB(A) range (77%).

Development of ringing schedule

Assessment Criteria

The criterion used to identify an appropriate schedule for open ringing was that any open ringing activity should not go against the wishes of more than 10% of the householders in the study area, as expressed in the responses to the questionnaire. In using this criterion it is assumed that those who did not respond to the questionnaire are not strongly opposed to some open ringing.

The choice of "10%" as a criterion level was made in recognition of both the expressed wish of a large group of people to hear the bells and also the need to provide protection of the peace and quiet of those who do not wish to be disturbed.

Assessment

There were 53 "No" responses to Question 1 out of the 237 responses, representing 9% of a total of 566 householders in the study area. It may therefore be assumed that allowing some open ringing would go against the wishes of 9% of the householders in the study area. The 10% criterion suggested above would therefore allow for a small amount of open ringing.

Analysis of the results to Questions 2 and 3 indicated that open ringing would meet the 10% criterion if allowed on 6 occasions per month between the hours 8.00 am - 7.00 pm Monday to Saturday and 9.00 am - 7.00 pm Sunday, for a period of not more than 10 minutes on each occasion, with a total of not more than 20 minutes on any day. There may be some scope to vary the numbers of occasions and their durations slightly, within the confines of a total of 60 minutes' ringing per month.

TABLE 3: Preferred Number of Occasions for Open Ringing

No. of Occasions per month	No. of Responses			
	Sound Level 50 - 64 dB(A)	Sound Level 40 - 49 dB(A)	Sound Level Not Indicated	TOTAL
0 - 5	11	31	3	45
6 - 10	5	17	3	25
11 - 15	5	18	0	23
16 - 20	3	11	0	14
21 - 25	0	6	1	7
26 - 30	4	37	2	43
Above 30	4	18	1	23
Did not answer	1	1	2	4
TOTAL	33	139	12	184

Ringing during the evening would not meet the 10% criterion. While ringing one bell before and after school would not meet the 10% criterion, this activity may possibly be brought within the noise legislation by careful selection of the bell and the ringing duration.

It was therefore considered appropriate to allow the bells to be heard by members of the community so they could make their own assessment. This is now in the process of being done done by means of a Ministerial exemption under the Environmental Protection Act 1986, with conditions reflecting the findings of the study[4].

Conclusions

While bell-ringing may be expected to invoke a positive response from many hearers, it must also be recognised that the same sound may cause a negative reaction from others who do not wish their peace and quiet to be disturbed. The assessment of noise from the proposed St Hilda's School Chapel Bells showed that, to the great majority, bells are "sound" and not "noise", provided the bells are rung at reasonable times. The study also showed that it is possible to balance the desire of the great majority of the community to hear bells at reasonable times with the wish of others for undisturbed peace and quiet, through a combination of objective sound level predictions with a simple subjective community survey.

Acknowledgements:

The author wishes to acknowledge the assistance of Richard Langford of the Department of Environmental Protection throughout the project, and the contribution of Mr Timothy Reynolds of Herring Storer Acoustics in preparation of the sound level contour map.

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1. "Towers and Bells", A J Frost (ed), The Central Council of Bellringers, Guildford, Surrey GU1 1BL, UK, 1990.
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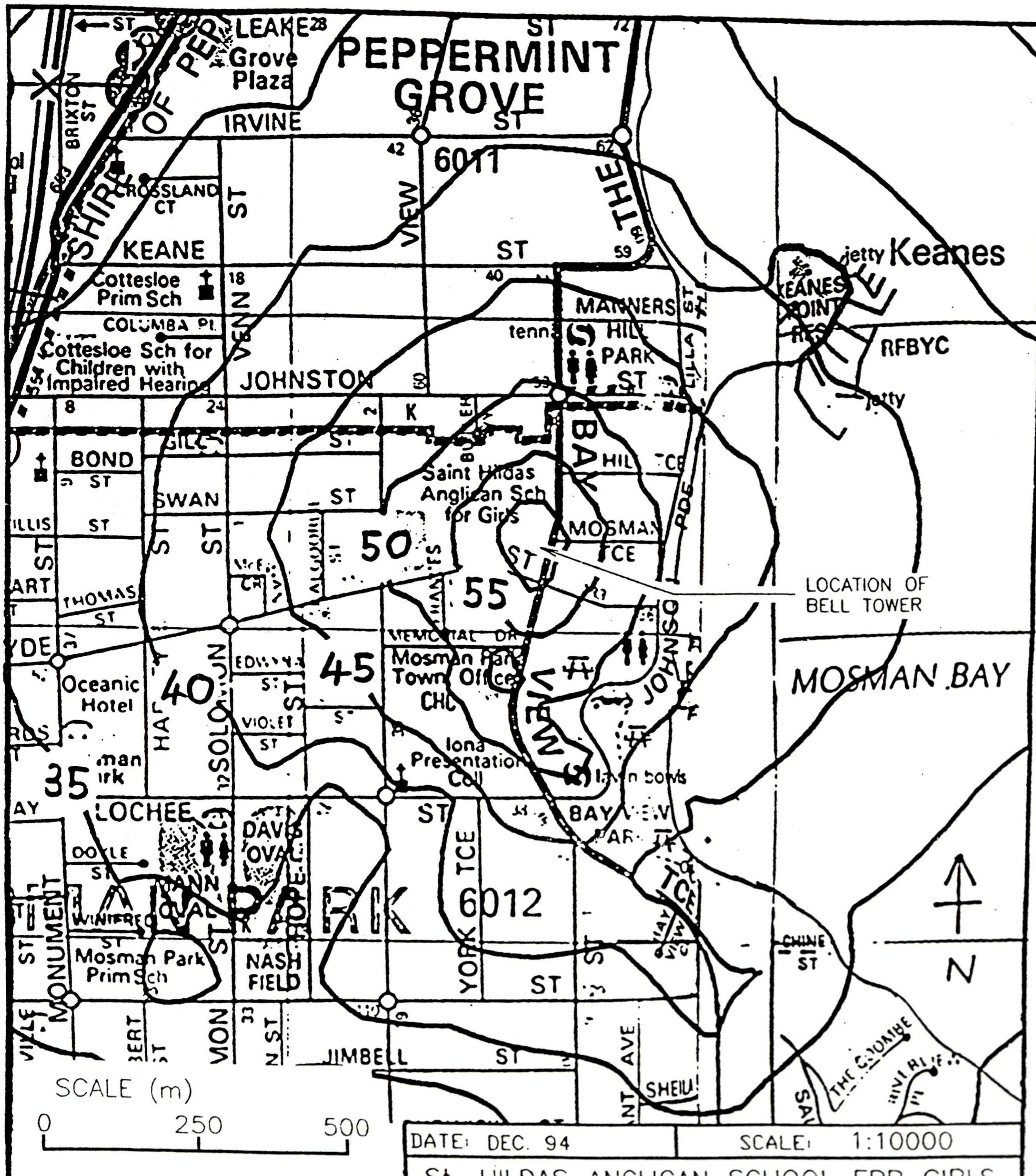
FIGURE 1 - SOUND LEVEL MAP



Source: Herring Storer Acoustics

Session 8A

Applications

Feasibility Study Of An Acoustic Projectile Location System

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Abstract

An acoustic projectile location system which is capable of detecting both subsonic and supersonic projectiles without the use of a special box target has been developed. A theoretical uncertainty analysis for the configuration proposed indicates that, for a target area 600 mm by 700 mm, the maximum uncertainty is less than 8 mm. Field tests of this system reveal other sources which can degrade the accuracy achieved. Various techniques for improving this accuracy are discussed.

Introduction

There are several commercially available projectile location systems. Generally, systems such as the Lomah/Superdart[1,2] which rely on detecting the leading edge of the shock wave generated by supersonic bullets, do not work for subsonic rounds and will not detect oblique shots. On the other hand, systems which work for both supersonic and subsonic rounds[3] require a box target made of special material which has to be replaced on a regular basis. Australian Patent No. 523897[4] provides good background information on the various means of detecting ballistic projectiles.

The objectives of this study on projectile location system were to explore

(i) the feasibility of detecting both subsonic and supersonic projectiles using a microphone-based system, without the use of a special box target;

(ii) to improve on the accuracy of detection by using cross correlation technique to determine the time delays between microphone signals instead of using the leading edge of the microphone signals; and

(iii) to compare the accuracies achieved from field tests with theoretical uncertainty estimates.

Apparatus & instrumentation

As shown in Figure 1, the proposed acoustic projectile location system consists of 3 electret microphones separated by 350 mm from each other. The microphones were of electret type with impedance 600 Ω, sensitivity -64 dB at 1k Hz and frequency response 50 Hz - 16 kHz, and were mounted directly behind the target. In the feasibility study, a triggering microphone was used to initiate the data acquisition. The microphone signals were

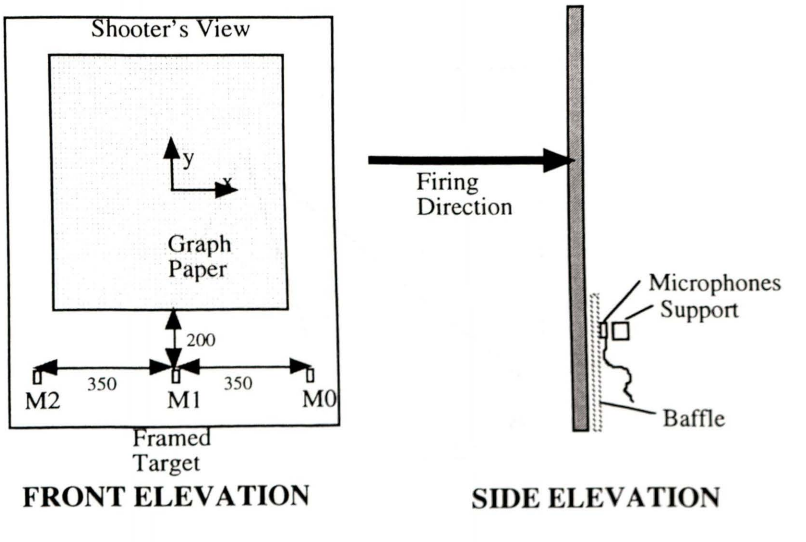

Figure 1 Set-up of microphones and target.

conditioned using Tektronix AM502 differential amplifiers. The three microphone signals were then digitised using a WIN-30 DS4 1 MHz data acquisition card with simultaneous sample and hold. The sampling rate was 250 kHz per channel. The data acquisition card was installed in a Toshiba 3200SXC laptop computer. The acquired microphone signals were then processed off-line to determine the location (x,y) coordinates based on cross-correlation analysis. It is envisaged that in the prototype system, all the electronic hardware involved (such as signal conditioners, A/D converters including data processor) can be assembled into one card and the coordinates of the target location can be calculated in real-time. With the set-up as shown in Figure 1, no special target box is required and the microphones can be further shielded from stray bullets.

Uncertainty analysis

The three microphone signals were used to determine the delays in path lengths δ_1 and δ_2 between the two end microphones and the central microphone (Figure 3). If the microphones are separated from each other by "a" and if the distance of the 'hit' location (x,y) is at a distance l from the central microphone, then it follows from geometry that

$$l^2 = x^2 + y^2 \tag{1}$$

$$\delta_1 = \sqrt{(a - x)^2 + y^2} - \sqrt{x^2 + y^2} \tag{2}$$

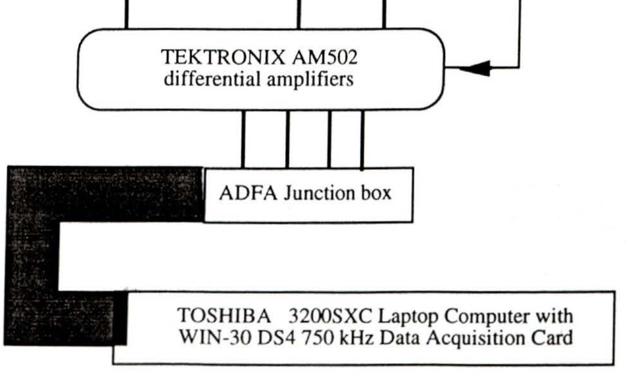

Figure 2 Schematic diagram of data acquisition system.

$$\delta_2 = \sqrt{(a + x)^2 + y^2} - \sqrt{x^2 + y^2} \tag{3}$$

It can be shown that the uncertainty in x, l and y can be estimated from equations (4),(5) and (6) respectively:

$$\Delta x = \frac{\Delta\delta}{2aD}\left\{[-2a^2\delta_2 - 2\delta_1^2\delta_2 - 4\delta_1\delta_2^2 + \delta_2^3]^2 + [2a^2\delta_1 + 2\delta_1\delta_2^2 + 4\delta_1^2\delta_2 - 2\delta_1^3]^2\right\}^{1/2} \tag{4}$$

$$\Delta l = \frac{\Delta\delta}{D}\left\{[\delta_2^2 - \delta_1^2 - 2\delta_1\delta_2 - 2a^2]^2 + [\delta_1^2 - \delta_2^2 - 2\delta_1\delta_2 - 2a^2]^2\right\}^{1/2} \tag{5}$$

$$\Delta y = \frac{1}{y}\sqrt{(l\,\Delta l)^2 + (x\Delta x)^2} \tag{6}$$

where $D = 2(\delta_1 + \delta_2)^2$

The total error e can, therefore, be quantified by $\sqrt{\Delta x^2 + \Delta y^2}$. The minimum theoretical uncertainty in estimating Δl may be attributed to the resolution of the digital sampling rate. For a sampling rate of 250 kHz and the speed of sound of 340 m/s, an uncertainty of 1.36 mm in Δl can be expected.

The theoretical uncertainties in the determination of the x and y coordinates for microphone separations of 350 mm are displayed in Figures 4(a) and (b) respectively. It can be seen that for most of the target area, the uncertainty in the x-coordinate due to the sampling rate is less than 2 mm while that in the y-coordinate is considerably higher. The total error e corresponding to the uncertainties in Δx and Δy shown in Figure 4 is depicted in Figure 5. It is evident that the total error is dominated by the error in Δy. For a target area 600 mm by 700 mm, the theoretical total error is within 8 mm. An obvious method to improve on the accuracy is to have an additional 3 microphones lined up on one side of the target.

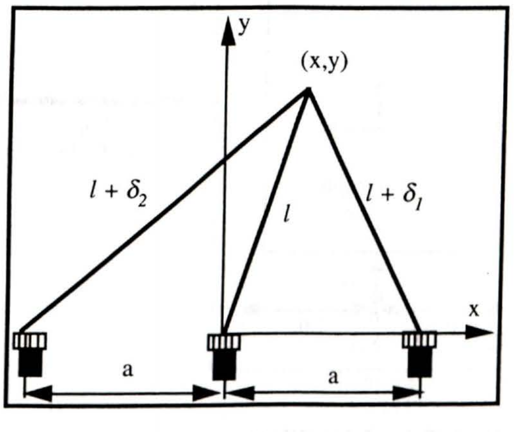

Figure 3 Coordinate system used determining the target location (x,y).

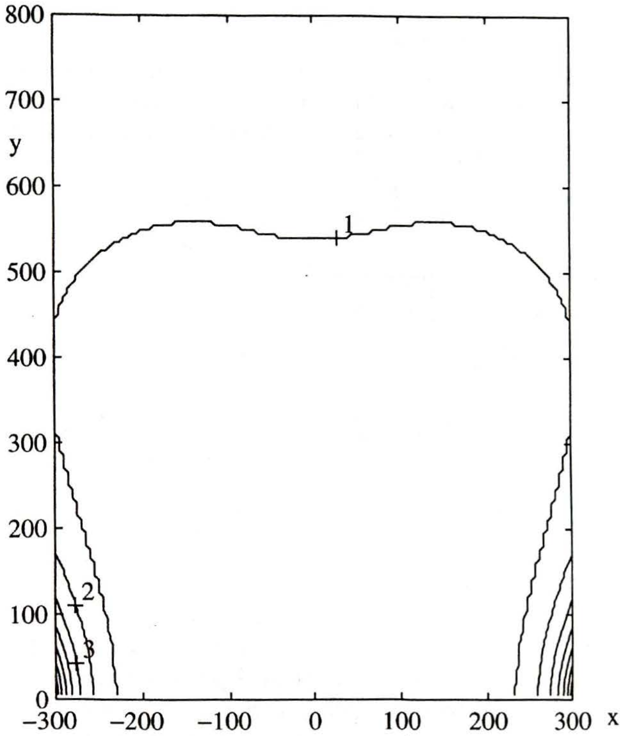

Figure 4(a) Contours of uncertainty in x.

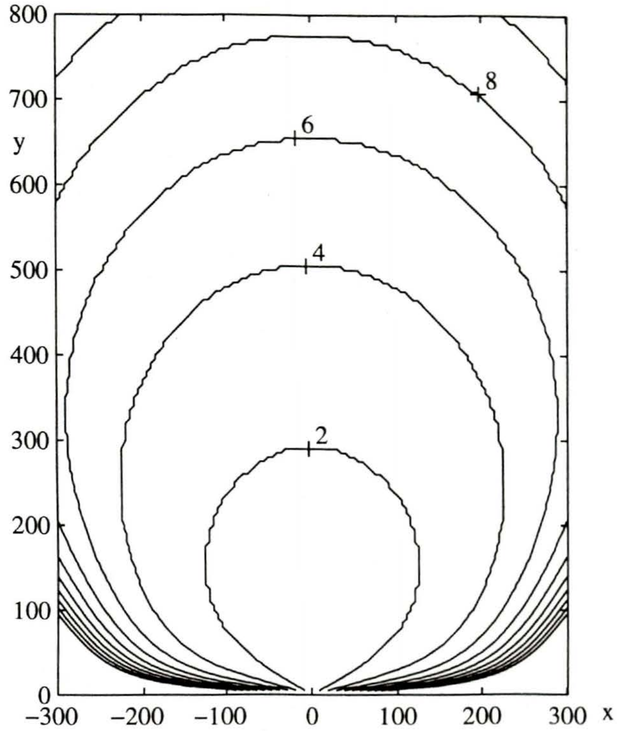

Figure 4(b) Contours of uncertainty in y.

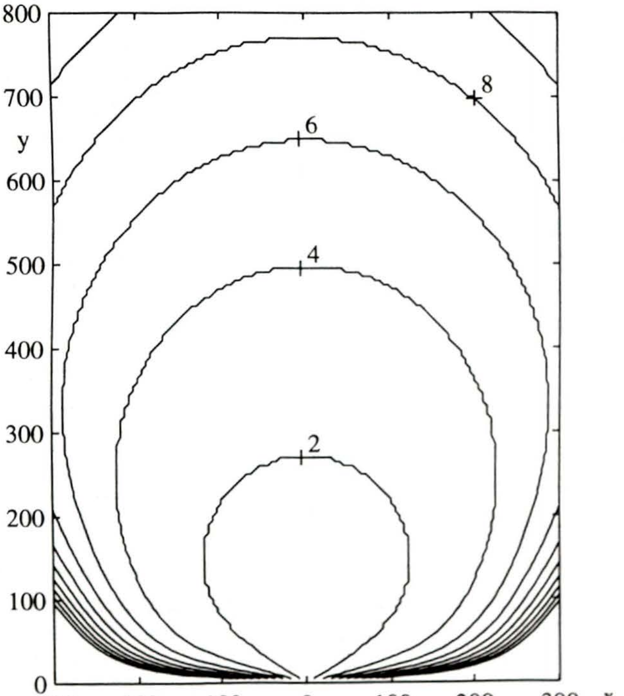

Figure 5 Contours of total uncertainty.

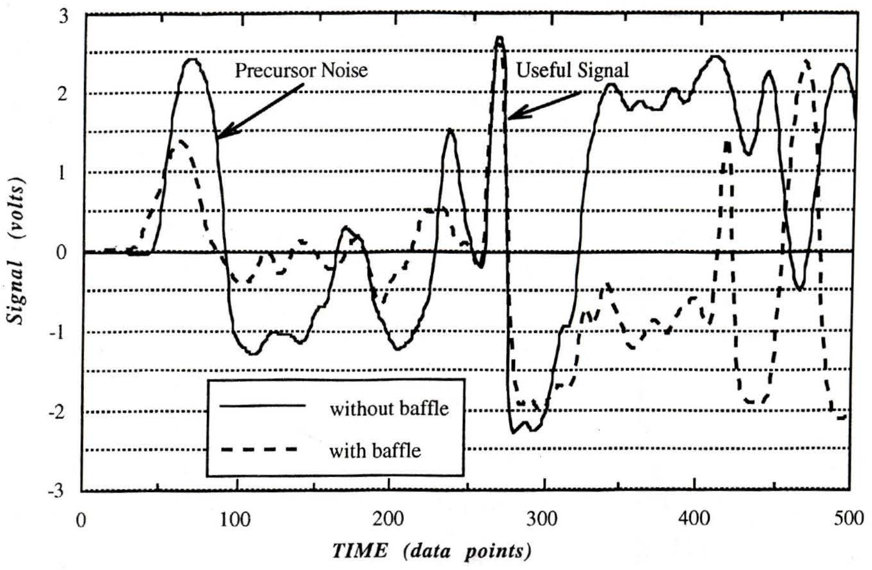

Figure 6 Comparisons of microphone signals with and without installation of a baffle.

This option, however, is not desirable because these microphones could be damaged by stray bullets. Other means of improving the accuracy is currently being investigated.

It must be pointed out here that this uncertainty analysis only takes into account the uncertainty due to the resolution in the digital sampling rate. Errors due to the nature and shape of the signal which could be influenced by other means have not been considered.

Field tests and discussions

The proposed system was tested in both indoor and outdoor range for a range of weapons with projectile velocities ranging from subsonic to supersonic, namely, .22 Ruger, 5.56 mm M16A2, 9mm Heckkler and Koch SD and 9 mm Heckler and Koch A3 weapons.

During the tests, it was noted that the signal from the 9 mm projectile was masked by a precursor noise and could not be recovered for processing. It appeared that the sound from the bullet penetrating the target was masked by a noise conducted along the target at a higher speed than the speed of sound in air. As soon as a baffle was installed between the target and the microphones as shown in Figure 1, this noise was significantly reduced and processing of the signal was possible. The effect of the installation of a baffle on a typical signal acquired is illustrated in Figure 6. It must be pointed out that the precursor noise appeared to occur only in the signal acquired from a large diameter projectile. This is perhaps the precursor noise due to a small projectile is too low in level to interfere with the signal.

It was also noticed that the spurious noise occurred at a much lower frequency than that of the signal of interest.

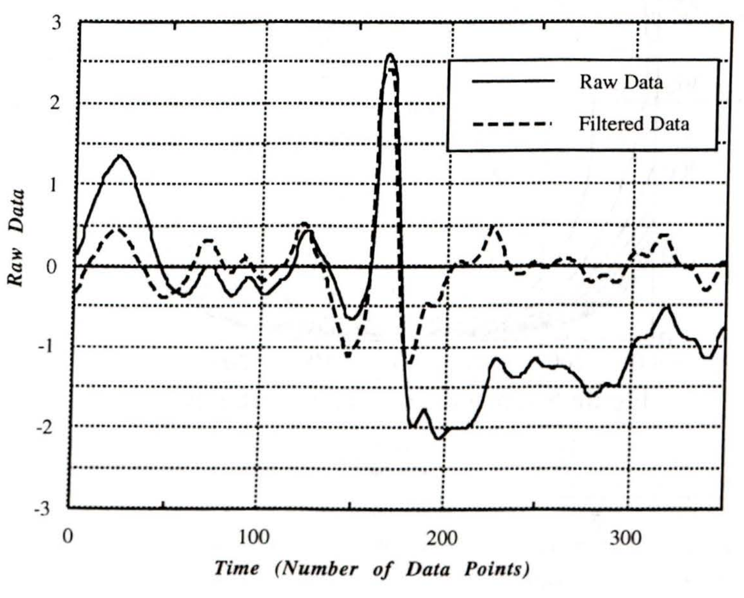

Figure 7 Comparison of unfiltered and filtered signals.

Table 1 Averaged errors in determining the 'hit' locations.

Ref	Weapon	Velocity (m/s)	Del x (mm)	Del y (mm)	Error (mm)
OUTDOOR RANGE					
a	.22 Ruger	317	1.7	6.6	7.0
b	5.56	821	2.2	14.4	14.7
INDOOR RANGE					
c	9mm SD	317	3.6	10.2	11.0
d	9mm A3	410	3.5	14.9	16.2

An attempt was, therefore, made in extracting the signal embedded in the spurious noise using digital filtering technique. It has been found that a 256 point digital filter with Hamming window reduces the spurious noise by a factor greater than 2 while reducing the peak amplitude of the signal of interest by less than 10%, as shown in Figure 7. It is envisaged that in the prototype system, both a physical barrier for the microphones and digital filtering will be used to enhance the extraction of the signal of interest.

The results for the tested weapons in both outdoor and indoor range are listed in Table 1. The averaged error in mm in the x direction obtained from a number of shots is tabulated as 'Del x' while that in the y direction is tabulated as 'Del y'. The distance in mm from the actual position in the target to the position determined by the microphone system has been determined for each shot and the average is tabulated as 'Error' in Table 1. It can be seen that for the .22 Ruger projectiles, the averaged error is reasonably close to the theoretical values predicted by the uncertainty analysis which only accounts for the digital sampling rate. On the other hand, the averaged error for other weapons are about twice as much as what would have been anticipated from the uncertainty analysis. This is because the accuracy in determining the 'hit' position is highly dependent on the nature of the signal, especially the signal-to-precursor noise ratio. It is expected with shielding of the microphones, the signal of interest will be greatly enhanced and coupled with digital filtering, the accuracy of the proposed target location system can approach that predicted by the uncertainty analysis presented here.

Conclusions

An acoustic projectile location system based on 3 low cost electret microphones has been proposed. A theoretical uncertainty analysis which only takes into account the resolution of the digital sampling rate shows that for a target area of 600 mm by 700 mm, the uncertainty in determining the 'hit' position is dependent on its position within the target area but is generally within 8 mm. Field tests have shown that for smaller projectiles, this accuracy is being approached. For larger projectiles, because of the presence of a precursor noise, the accuracy achieved is double that of what is expected from the uncertainty analysis. It has been demonstrated that the precursor noise can be significantly reduced by proper shielding of microphones and applying appropriate digital filtering to the signals. By optimising the shielding of the microphones and the choice of digital filters, an accuracy of about 8 mm using the proposed system for both subsonic and supersonic projectiles can be expected to be achievable. A system based on 5 microphones is currently being considered which has the potential to improve on this accuracy.

Acknowledgment

This study has been supported by the Defence Centre, Perth. The encouragement and support provided by Maj. B. Hughes and Mr. J. Tick are gratefully acknowledged.

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2. Lomah Brochure. Polytronic Int., Switzerland.
3. Polytronic TG 3000 (box target) brochure.
4. Australian Patent No. 523897 Projectile locating apparatus for use in target shooting.

Acoustics Applied - putting the science and technology to work

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Perth, Western Australia 15-17 November 1995

Session 8B

Standards and Regulations

Appropriate Standards

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Abstract

Australian Standard AS2012.1-1990 Acoustics - Measurement of airborne noise emitted by earth-moving machinery and agricultural tractors - Part 1: Determination of compliance with limits for exterior noise. This Standard describes a method for determining the exterior noise emitted by earth-moving machinery and agricultural tractors in terms of the A-weighted sound power level while the machine is stationary. At six positions on a hemispherical surface, the equivalent continuous A-weighted sound pressure levels are measured. The A-weighted sound power level of the machinery is calculated from the measured values. This Standard applies to the following types of machinery: a) agricultural tractors; b) crawler and wheeled excavators; c) crawler and wheeled loaders; d) crawler and wheeled dozers; e) backhoe loaders; f) graders; g) compactor/loaders; h) four-wheel drive skid-steer machines. In the absence of an Australian Standard for earth-moving machines larger than an agricultural tractor and the items listed above, suppliers and users of *large* earth-moving machines, such as dump trucks, shovels and draglines, expect AS2012.1 to be suitable and satisfactory for their purposes. This is not the case, as our paper and seminar presentation will show. To overcome the difficulties caused by this limited situation, I want to present a new Australian Standard to fill the vacancy for the sound level measurement of *large* earth-moving machinery.

Introduction

As the name implies, Standards are designed to provide a set of steps to be taken in a process of construction or testing, repeated from time to time. The purpose is to gain uniformity in the process so that the task is carried out the same way each time. By definition then, it must not leave room for ambiguity where a different interpretation can be applied by different persons.

By the same token, the Standard is not a set of regulations bound by law unless the Standard is included for application within a Lawful Regulation. Normally, to my knowledge, this is not the case. Therefore, if in the process of applying a Standard, some procedure cannot be carried out, for whatever reason, no crime is committed. The results obtained may or may not be the same as previous measurements, or the same as the next set of measurements, or the same as those obtained by others who were able to apply the Standard. Nevertheless, if the Standard can be followed it should be. If it cannot be followed then some notation should be made in the Report Document to this fact.

At my last count there were 45 Standards on Acoustical Subjects listed in the latest Catalogue of Australian Standards. I cannot claim to have used all of them or even most of them, but I do use some of them repeatedly. One such Standard is AS2012 - 1990 - Acoustics - Measurement of airborne noise emitted by earth-moving machinery and agricultural tractors - Stationery test condition. There are two parts to this Standard. Part 1: covers the Determination of compliance with limits for exterior noise. Part 2: applies to the measurement of noise levels in the 'Operator's position.'

AS2012 commenced its life of service in 1977 following the issue of a Draft and the receipt and consideration of comments. It was given the Title "Method for the measurement of AIRBORNE NOISE from agricultural tractors and earth moving machinery."
I believe this original Standard came from work carried out by Mr Horrie Weston of the NSW Dept of Health in an attempt to passively introduce the need for hearing protection for those working on road construction and on farms. All the requirements of this Standard for operator noise exposure and for the bystander position point to this fact.

The Preface to the Standard described the method for the measurement of noise emission as perceived at the operator's position and at specified bystanders positions. It then stated: The information in this standard may be used for noise control by engineers, for comparison of machines by purchasers, or for legislative control.

The text is expressed in a simple, logical and straightforward manner and includes measurement of noise generated by the machine during its operation and operation of any attached implements. The measurements for a backhoe for example would include operation of that implement. Measurements were to be taken at 7 metres from the surface of the machine, on each side, at the front and at the rear. For self-propelled earth moving machinery, sound levels were to be taken at the bystanders position at 7 metres, on each side of the machine.

Noise measurements were then taken at the operator position to allow a noise dose to be determined for the operator. Further details were given in Appendix A for the calculation of the dose covering various time frames or cycles of operation typical for such a machine. This section of the Standard included Table A1, listing typical work cycles for the machines which may be measured. These machines included Loaders, Dozers, Scrapers, Tractors, Graders, Rippers, Rollers and Off-highway Trucks.

While measurements were taken at 7 metres for the bystander position rather than for determination of an acoustic power level for the machine, the Standard was soon used for this purpose. It also became apparent that certain aspects of this Standard could not be applied, such as measuring in front of a bulldozer while it was operated. The term bystander became an operative word meaning just what it said. We then had to adjust the distance of 7 metres if this situation was hazardous. So, the bystander position was applied by us to mean a safe place nearby, where operation of the machine could be observed and the sound level measured.

Observance of the Standard for some machines started to come adrift at this stage. Some machines operate by movement of implements while remaining stationary while others operate by moving along.

Then came difficulties due to the size of the machine.

At this stage, the Standards Committee AV/6 produced a draft standard revising the 1977 issue. It applied a completely different methodology for measuring the noise of the machine. It also had a new purpose, that is, to determine the sound power level of machines.

To achieve this purpose we now had to measure the machine at 6 locations, on a hemispherical surface having a radius related to the size of the machine. When the machine to be measured had a length greater than 4 metres, the radius for measurement is 16 metres. Four of the locations were 1.5 metres above the ground and two locations at 11.36 metres above the ground. Further study of the Standard revealed that it applied to the same types of machinery as applicable in the 1977 issue, viz:

(a) Agricultural tractors;
(b) Crawler and wheeled excavators;
(c) Crawler and wheeled loaders;
(d) Crawler and wheeled dozers;
(e) Backhoe loaders;
(f) Graders;
(g) Compactor/rollers;
(h) Four-wheel drive Skid-steer machines.

No reference is made to larger machinery such as Scrapers and Off-highway Trucks etc.

To quote again from the revised Standard, " Ideally, the site for the measurement of these machines should be asphalt or concrete, with no sound reflecting obstacles within a distance from the source, equal to three times the greatest distance from the source centre to the lower measuring positions." This equates to a radius of 48 metres, almost the size of a football field.

I decided the Standards Committee personnel were not people who undertook the measurements required by the Standard they were designing. Further, Caleb Smith Consulting could become fairly unique in being able to undertake the measurement of machines at the specified locations using a telescopic mast or boom, assuming we could find a football field made of concrete.

Because this new version of AS2012 was so different from the 1977 issue in its requirements and methods of measuring Airborne Noise from Agricultural Tractors and Earth Moving Machinery, I wrote to the SAA and suggested that they leave the 1977 issue as it was and give the new version a new number. In that way I could continue to measure Dozers, Scrapers, Off-highway Trucks and large mining machines under the auspices of the 1977 Standard and the new Standard could be used to measure sound power levels from the specified machines, ie, Tractors, Graders, Backhoes, Front End Loaders, etc.

While this draft standard was being created I was using the current 1977 issue to measure the sound from all sorts of machinery regardless of whether it was nominated in the text or not. The only departure I made from the quoted methodology was the measuring distance of 7 metres from the surface of the machine. This I took upon myself to decide, relative to safety and accessibility.

There had been no suggestion of a laboratory type situation or environment in the 1977 Standard. Rather, the sound level measurements were to be taken while the machine carried out its normal function. Certainly there was the necessity to ensure that the background sound level was more than 10 dB(A) below that of the machine, that the weather was suitable and apart from the ground, there were to be no reflecting surfaces within 50 metres of the machine being measured.

So as you can see, there was no reason to limit the type of machines that could be measured under this 1977 issue of Standard AS2012.

Because there was a Standard available, albeit a little inappropriate in the type of equipment originally intended, I could see no reason not to use it for the measurement of mining machinery, regardless of its size, that is, until the new version came along specifying a measuring distance of 16 metres, which I could ignore, and the insidious measuring position 11.36 metres above the ground around the machine. I can carry a sound level meter and tripod on to the mine site and measure any mining machine, believing I had the right to choose a safe distance away from the machine to carry out the measurements. Now I ask you, how was I supposed to get myself, or the microphone, 11.36 metres above the ground. Let us face the fact that the authors of the new version of the Standard tried to have the measurements carried out in a laboratory type environment.

My letter of concern to the Standards Association of Australia indicating the inappropriateness of the new Standard went unheeded. The 1977 Standard was withdrawn and in its place came the new Standard AS2012-1990 - Acoustics - Measurement of airborne noise emitted by earth-moving machinery and agricultural tractors-Stationary test condition.

The world would not stand still while we considered our position. The new Standard claimed to be appropriate for the measurement of many machines scheduled in the 1977 issue. While I applied the Standard for the measurement of any and all earth moving machines, and I suppose I am as guilty as anyone in applying the Standard to those machines for which it did not apply, the simple fact is, the 1977 issue did not apply to Shovels, Draglines, Scrapers and Off-road Trucks. There was no Standard in the catalogue to cover the measurement of these machines. So I could ignore it. Not so fast.

By the time I had measured a thousand machines between 1977 and 1990 under the 1977 Standard and issued a thousand reports quoting same, mining personnel had become habituated to the Standard and its title. Within this time-frame also, EPA type persons required to know how much noise was being propagated into the environment and I was being asked to provide the answer.

Now, in order to be informed in their decision making, some mining people bought the Standard and quoted it in their purchase specifications as "Operator noise exposure levels and machine sound power levels shall be measured in accordance with AS2012 - 1990."

New and proposed mines were being investigated by Commissions of Enquiry and in the course of events, the Authoritative Powers gave permission for the mine to be constructed and operated, provided the sound power level of equipment did not exceed such and such dB(A). Where did this magic number come from? Well it so happens, one mine being investigated had to purchase all the farm lands on which it would operate as well as all the farm land where the sound level due to mining would exceed the background sound level plus 5 dB(A).

What was the background sound level? We could and should have a seminar on that subject alone, the controversy that question caused. When was the sound measured? Where was it measured? For how long was it measured? What was the weather at the time? Were there insect noises? Was there a temperature inversion applying at the time?

After much deliberation the Authorities investigating this particular mine decided the background sound level was 32 dB(A) and the mine could create an L10 sound level of 37 dB(A), when measured at the nearest exposed residence under neutral atmospheric conditions. When the nearest residence was identified and the distance to the mine established, the authorities then decided to limit the sound power level of all mining equipment on the site to 118 dB(A)Lw.

Now, such is the power of the EPA in NSW that before you could say hold on, all and sundry persons having any association with a mine, be he operator, manager, or supplier of equipment, *knew* that all mining equipment operating in the Hunter Valley of NSW *must* be measured under AS2012 - 1990 and *must* have a sound power level not exceeding 118 dB(A)Lw.

The Standard also requires the measurement of sound absorption of the ground on which the item of equipment is to be measured. The result of this measurement is then applied to the calculated average sound pressure level of the machine and the calculated sound power level adjusted to compensate for sound absorption due to the ground condition. The adjusted sound power level is then quoted in the report.

So, when the adjusted sound power level of the machine came to 119 dB(A) or any other number greater than 118 dB(A), the machinery supplier and the mine management decided we had obviously measured the machine incorrectly. The mine management and/or their suppliers then went through the Standard and through our measuring methodology with a fine tooth comb to find why their machine had exceeded the Authority's permissible power level of 118 dB(A)Lw.

This is why we need a standard for the measurement of LARGE earth moving machines and what better time to create one than here today with the best acoustical brains in the country gathered in one place.

If manufacturers of mining machinery are going to make larger and larger machinery to suit the demands of the mining industry, we who monitor the acoustical amenity of the residential neighbours need to come to grips with the necessity to measure the emission of noise from these kings of the lode.

Agenda For The Construction Of A New Standard

Introduction

Let us assume that those attending this seminar have read the Paper and now come to its presentation, to take part in the design of a new standard for the measurement of sound emanating from LARGE earth moving machines. Mr Chairman, I invite all those present to take part in the design of this Standard.

1. What machines are to be included in the Standard?
   a) Shall we include any earth moving machine not included under AS2012.1?
   b) All those having a length greater than 8 metres?
2. What is expected of the Standard?
   a) Sound pressure level measurements of all earth moving machinery, both stationary and mobile types, while operating in a stationary position on high-idle and the determination of the Sound Power Level.
   b) Sound pressure level measurements of stationary machinery at a safe distance while the machine is operated on a normal work cycle, such as Shovel, Excavator and Dragline.
   c) Drive-by tests for those machines that work by mobile operation such as Trucks, Graders, Scrapers, Front End Loaders and Dozers.
   d) Measure within the Drivers Cabin during a normal work cycle.
   e) Measure adjacent to the Driving Engine or within the Machinery House, if applicable.
3. In what direction and at what distance shall we measure the stationary machines?
   a) The decisions for these two question must be related to the type of machine being measured. We need to measure at a safe distance from the surface of the machine while it works for a few minutes or a few cycles, which will give a clear indication of its Sound Power Level for its working operation. I believe we should measure the sound level of the machine on each of the clear sides while in a stationary position with the engines on high idle, when applicable?
   b) If we plan to measure at a distance from the surface of the machine equal to one major dimension of the machine, then 1.5 times the dimension then twice and if possible at two and a half times the major dimension, I believe a sensible overall Sound Power Level of the machine can be determined.
4. Do we want any measurements over the machine as described in AS2012.1-1990?

   If so, how will those measurements be undertaken?
5. What about sound absorption of ground conditions where the machine is measured as required by AS2012.1-1990?
   a) Can this sound absorption factor be obtained by measurements taken at the three or four distances described in 3 above?
   b) At the present time, we determine the additional sound propagation loss due to the condition of the ground using a sound power source before the machine has arrived on site or after it has been taken away. However, this may not be possible with very large machines, particularly those designed to work from a stationary position such as Shovels and Draglines. We can, of course, return at another time to obtain this information after the machine has been moved, or, we can measure a similar piece of ground remote from the machine construction site.
   c) Shall we say, "the propagation loss shall be determined by an appropriate means, and suggest each of the above methods. The engineer will then make the decision on site at the time and note the method in the Report."

We need to include the measurement of sound at the operator position.

a) In regard to this aspect, I believe we should adopt a method similar to that described in AS2012-1977, or AS1269-1989 as it will provide an operator noise exposure level for a cycle of operating conditions rather than for a static situation.

"THE LING THING"

1995 Conference
Australian Acoustical Society,
Perth, W.A.

Paula Schneider, Curtin University
Jill Duncan, University of Western Australia

ABSTRACT

In the field of education of the deaf and hearing impaired, the "LING THING" has been widely utilised. The Ling Thing has a foundation in specific acoustical properties of sounds and their representation of the entire speech spectrum. This presentation will review the acoustical properties of the the Ling Sound Test and its applications in the areas of audiology and education of children with hearing loss, generally known as the Ling Thing.

INTRODUCTION

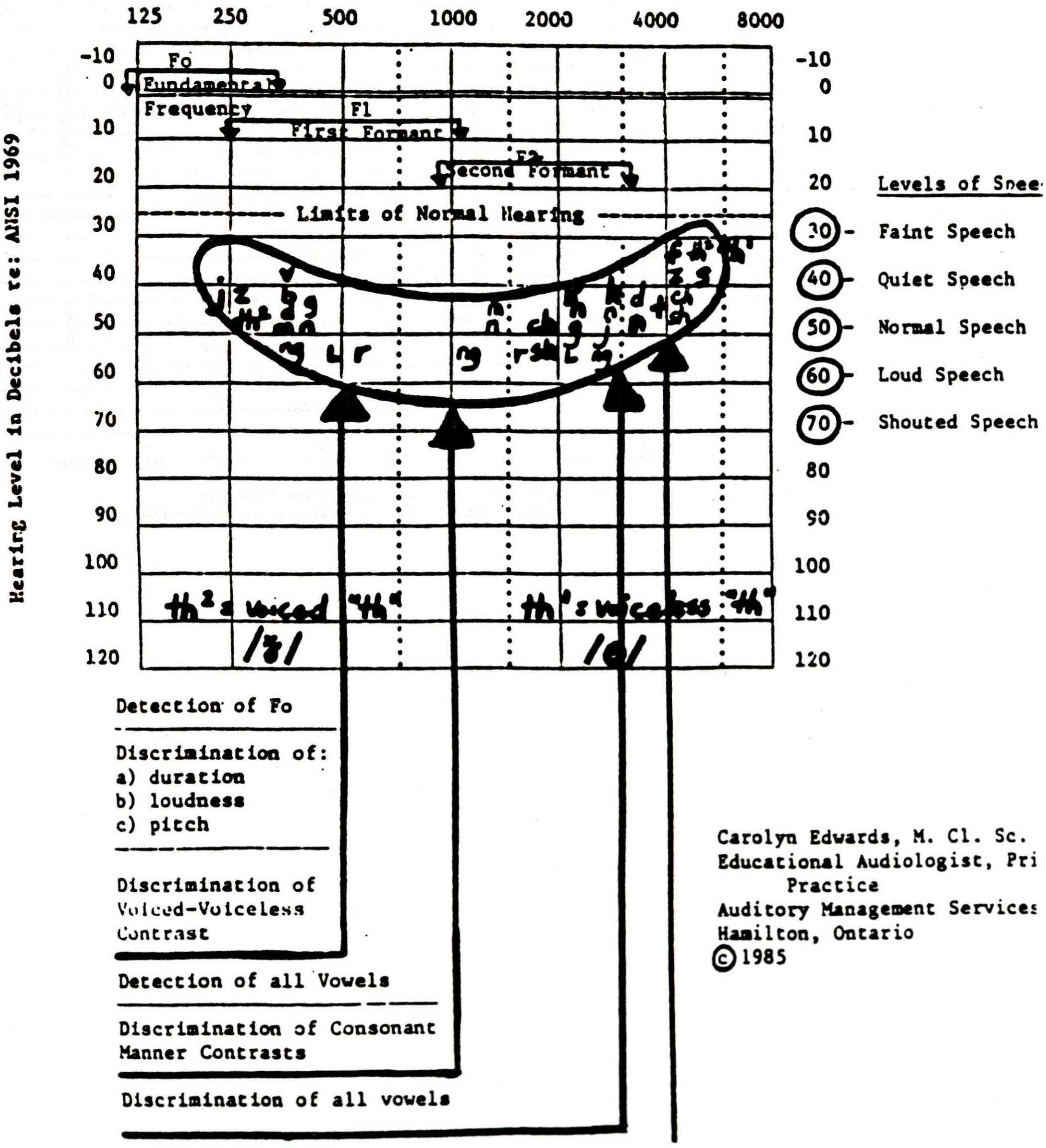
Acoustics may be defined as the science of sound. Sound may be described in terms of a physical phenomena such as frequency, intensity, and the spectrum of vibrations. Sound may also be defined in terms of psychological phenomena, or human (subjective) reactions such as pitch, loudness, sound quality, and direction. This paper will review the "Ling Thing" in a combination of physical and psychological phenomena. The practical applications of the Ling Thing in audiology and in the education of children with hearing loss will be identified.

HISTORY

In 1976 the book entitled *Speech & the Hearing Impaired Child: Theory & Practice* was published. The emphasis of the author, Dr. Ling, was that most children with hearing loss could learn to speak and understand spoken language given adequate opportunity to do so. This systematic approach was adopted by many educational programs throughout the world. The approach with its many facets and with its foundations in audition became known as the "Ling Thing".

UNDERLYING PRINCIPLES

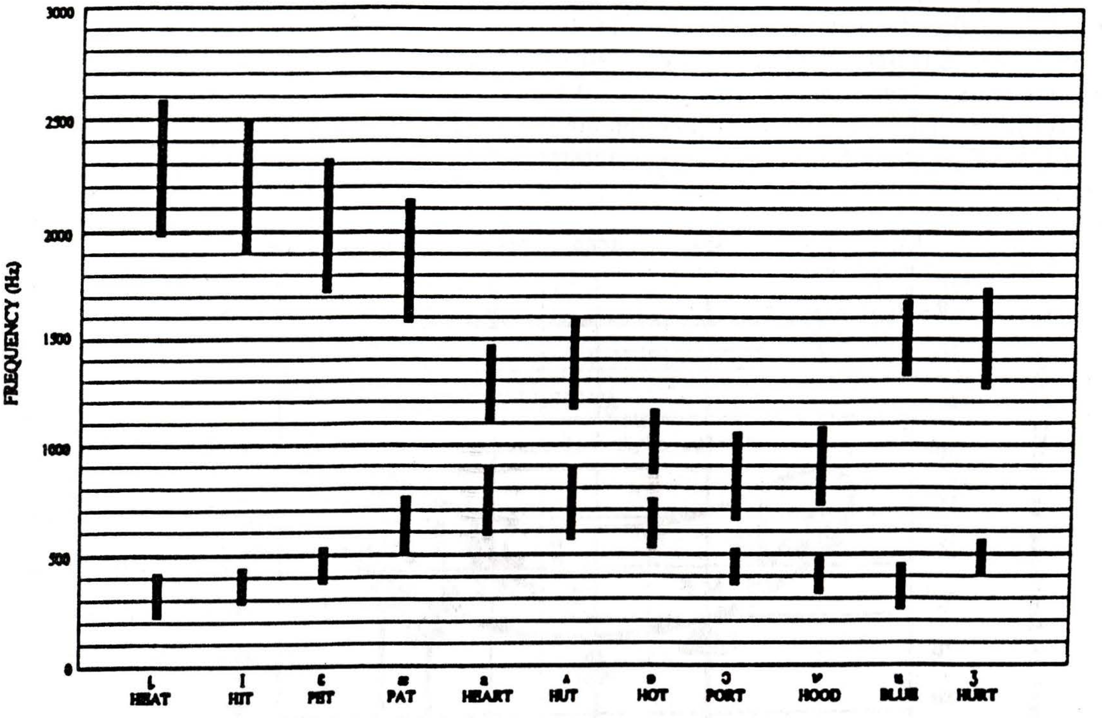
The human ear is able to detect sounds from approximately 250 Hertz (Hz) through ultra high frequencies such as 10,000 Hz or greater. The knowledge of the importance of some frequencies is minimal. What is known is that the ear detects the frequencies of 250 Hz through 8000 Hz quite well and that the majority of English speech sounds fall within this range. The contribution of the various frequencies to the discrimination of speech and speech intelligibility have been well documented. (French & Steinberg, 1947, Ling, D., 1976, Edwards, C., 1985). An interpretation of this information is presented here.



As seen on this overview, if one draws a line around the majority of speech sounds, a banana shape appears. This has been termed the speech banana and is a common tool for reference in audiology and in the field of education of children with hearing loss.

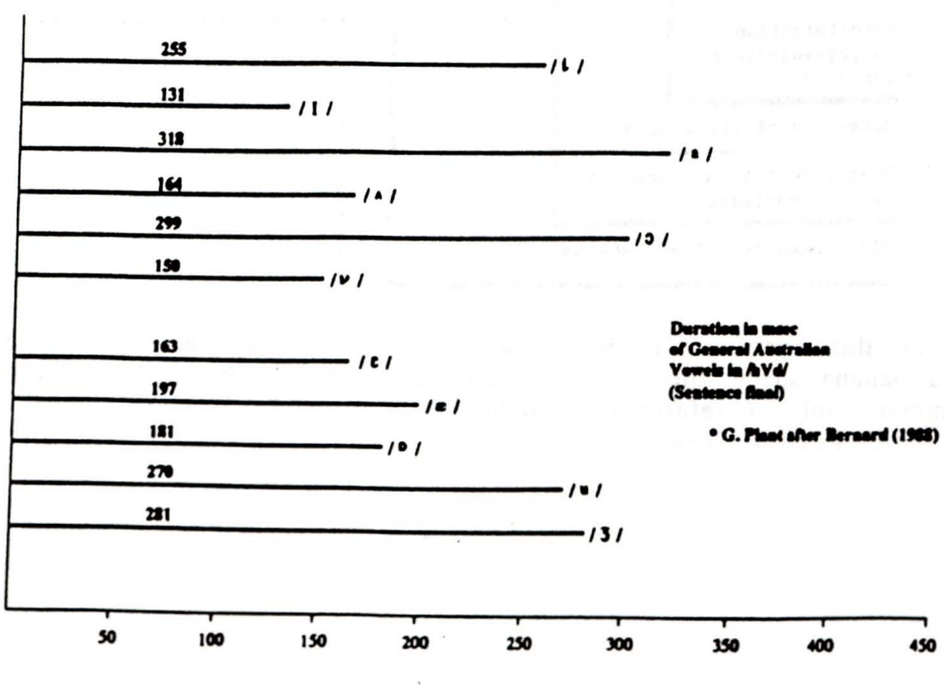
Speech sounds may be seen as quite distinct. They may be identified in terms of intensity, duration, and frequency.

Diagram 1



Australian English Vowels Robert H. Mannell (Speech, Hearing and Language Centre, Macquarie University.)
Average F_1 versus F_2 (+/- 2 standard deviations) for all male speakers. (The average of broad+general+cultured male speakers.)
Female values can be approximated by multiplying the male values by 1.2.

Diagram 2



Relative levels of speech sounds as heard by the ear.

Average (dB)	Speech Sound	
26.0	ɔ	(sought)
25.8	a	(barn)
25.8	aɪ	(sigh)
25.6	aʊ	(now)
25.3	oʊ	(soap)
24.2	æ	(sat)
23.6	ʊ	(foot)
22.0	ɛ	(set)
21.8	eɪ	(day)
21.8	ɜ	(third)
21.1	u	(food)
20.1	ɪ	(sit)
19.1	l	(low)
18.9	i	(seat)
17.4	ŋ	(sing)
17.1	ʃ	(shed)
14.5	tʃ	(child)
12.8	n	(no)
12.7	dʒ	(jump)
11.3	m	(me)
11.3	t	(ten)
10.9	g	(gate)
10.6	k	(key)
10.2	ð	(then)
9.4	d	(den)
8.8	h	(home)
7.6	z	(zero)
7.3	b	(bend)
7.0	p	(pen)
6.8	v	(vine)
6.7	f	(fine)
6.3	s	(said)
1.0	θ	(thin)

These levels are the average of two measures- the relative levels as measured with respect to the threshold of audibility and the relative levels measured with respect to that level at which the sound is just recognizable.

Working Papers of the Speech and Language Research Centre, Macquarie University.

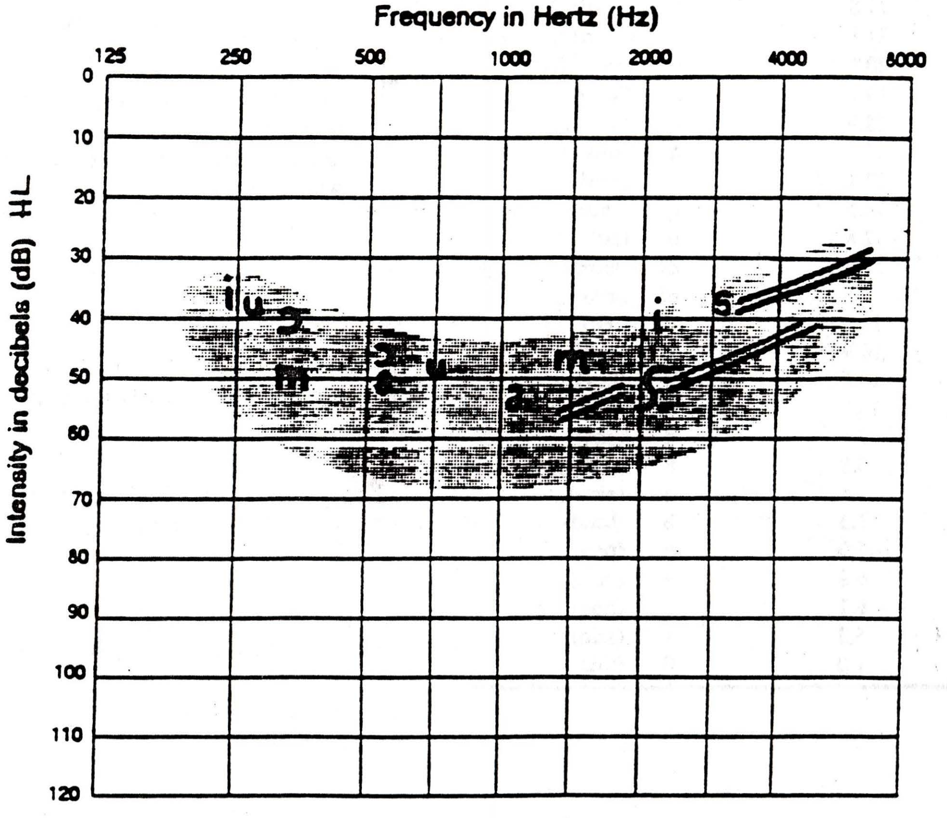
* J.R. Bernard & R.H. Mannell (1986)

Australian English Vowel and Diphthong Values for General Male Speakers

Vowels		F_1	F_2	F_3
i	(heed)	299	2263	2763
ɪ	(hid)	367	2211	2735
ɛ	(head)	459	2007	2620
æ	(had)	628	1885	2586
a	(hard)	718	1366	2426
ʌ	(Hudson)	746	1435	2500
ɒ	(hod)	613	1051	2375
ɔ	(horde)	434	810	2401
ʊ	(hood)	397	899	2416
u	(who'd)	344	1660	2370
ɜ	(heard)	472	1503	2543

Diphthongs		Target 1 F_1	Target 1 F_2	Target 2 F_1	Target 2 F_2
eɪ	(Hades)	696	1576	361	2135
aɪ	(hide)	677	1153	450	1764
ɔɪ	(hoyed)	461	925	359	1964
aʊ	(how'd)	639	1758	630	1277
oʊ	(hode)	676	1497	402	1675
ɪə	(heered)	344	2240	419	1919
ɛə	(haired)	435	2080	476	1881
ʊə	(hoored)	399	1116	434	1286

Speech sounds may also be seen to correspond to the puretones, narrowband noise (NBN), and warbled tones presented by an audiometer. Given this correspondence, a simple, informal test was devised by Dr. Ling. The distinct acoustic patterns initially chosen for this test were /u/, /a/, /i/, /ʃ/, and /s/. These sounds were representative of all speech sounds within the speech banana.



An addition of the /m/ sound occurred in regards to the detection of nasality cues in those with hearing loss. A further adaptation was made for use with the Australian population. The Ling Sound Test with its practical acoustic basis become the foundation for several applications generally known as the "Ling Thing". These applications include the following:

a. Development of speech intelligibility in children with hearing loss.

b. EARSHOT: Practical check of aided responses in real life situations.

c. Guide to the selection and verification of the fitting of amplification.

d. Listening check of amplification devices and other assistive listening devices

e. Hearing test for infants

f. Screening test for infants greater than 6 months of age.

BASIS FOR DEVELOPING SPEECH INTELLIGIBILITY

One of the most effective ways for developing speech, and intelligible speech, is through the exploitation of residual hearing. The understanding of speech information requires levels of auditory skill development. These have been defined by Erber, 1982) as the following:

Detection - the ability to determine the presence or absence of sound

Discrimination - the ability to perceive differences between sounds; that is, differences in acoustic qualities, intensities, durations, and/or pitches.

Identification - the ability to label or name what has been heard, by repeating, pointing to, or writing the word, sentence or environmental sound perceived.

Comprehension - the ability to understand the meaning of acoustic messages by reference to his/her knowledge of language. The response differs in content form the stimulus, but is closely associated in some way (i.e.: answering questions about a story, paraphrasing a story, giving the opposite of the word, etc.....

These levels of auditory skill development are the basis of auditory learning and are critical in the development of intelligible speech for children with hearing loss. The use of audition is the most effective way for developing speech. Determining the child's ability to detect all aspects of speech and the ability to perceive differences allows more effective use of strategies in the development of speech. The contribution of the various frequencies are listed below in regards to the aspects of speech development.

Speech Information Available at 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz

At 250 Hz, plus or minus 1/2 octave, the following speech information is available:
- 1st formant of vowels /u/ and /i/
- The fundamental frequency of females' and children's voices
- Nasal murmur associated with the phonemes /m/, /n/, and /ng/
- Male voice harmonics
- Voicing cues
- Suprasegmental patterns (stress, rate, inflection, intonation)

At 500 Hz, plus or minus 1/2 octave, the following speech information is available:
- 1st formants of *most* of the vowels
- Harmonics of *all* voices (male, female, child)
- Voicing cues
- Nasality cues
- Suprasegmentals
- Some plosive bursts associated with /b/ and /d/

At 1000 Hz, plus or minus 1/2 octave, the following speech information is available:
- 2nd formants of back and central vowels
- Important CV and VC transition information
- Nasality cues
- Some plosive bursts
- Voicing cues
- Suprasegmentals
- The important acoustic cues for *manner* of articulation are available at 1000 Hz.

At 2000 Hz, plus or minus 1/2 octave, the following speech information is available:

THIS IS THE KEY FREQUENCY FOR INTELLIGIBILITY OF SPEECH

- 2nd and 3rd formant information for vowels
- CV and VC transition information
- Acoustic information for the liquids /r/ and /l/
- Plosive bursts
- Affricate bursts
- Fricative turbulence
- The important acoustic cues for *PLACE* of articulation are available at 2000 Hz.

At 4000 Hz, plus or minus 1/2 octave, the following speech information is available:

This is the key frequency for /s/ and /z/ morpheme audibility. The /s/ and /z/ phonemes are critical for language learning because they signal:

plurals	idioms	possessives	auxiliaries
3rd person	questions	copulas	past perfect

Adapted from Ling, D., Foundations of Spoken Language for Hearing-Impaired Children. Washington, DC: The Alexander Graham Bell Association for the Deaf (1989).

In essence, if the child is able to hear the various characteristics of speech sounds, he/she will be able to produce that information intelligibly, with informal or formal training. If a child is unable to hear various characteristics, other strategies (visual & tactile) would be considered.

EARSHOT

EARSHOT may simply be defined as the distance over which the Ling sounds may be heard. The clarity of the message may be influenced by the distance from the speaker, the background noise present (the signal to noise ratio), and the reverberation characteristics of the environment. Further, the variables regarding the speaker (clarity of speech, voice pitch, intensity, etc....) as well as the variables involving the listener must be taken in to account. Listener variables may included the severity of the loss, the language basis of the listener, the amplification worn, etc....). EARSHOT distances may be determined in a small amount of time and may be described in terms of detection or in terms of discrimination.

LING SOUND TEST

DETECTION DISCRIMINATION

SOUND	DIRECTLY BEHIND	ONE (1) METER	TWO (2) METERS	FOUR & ½ (4.5) METERS
/m/				
/u/				
/a/				
/i/				
/ʃ/				
/s/				
		3.28 feet	6½ feet	15 feet

EARSHOT distances may be determined in any situation. EARSHOT distances will help determine what a child or an adult with a hearing loss may hear and overhear in conversation.

GUIDE TO SELECTION & VERIFICATION OF AMPLIFICATION SELECTIONS

When identifying a hearing loss, typically an audiogram is obtained for unaided thresholds. This information is utilised in a variety of fashions to determine electroacoustic settings for amplification devices. Usually these characteristics involve gain and frequency response outputs. Once a hearing aid is selected and fit to the individual, an aided audiogram is obtained to show the amount of improvement in thresholds, termed functional gain. These responses are plotted onto an audiogram and indicate a gross representation of hearing aid benefit. This information tells little about the reception of speech information speech. This point may be emphasised in the following figure.

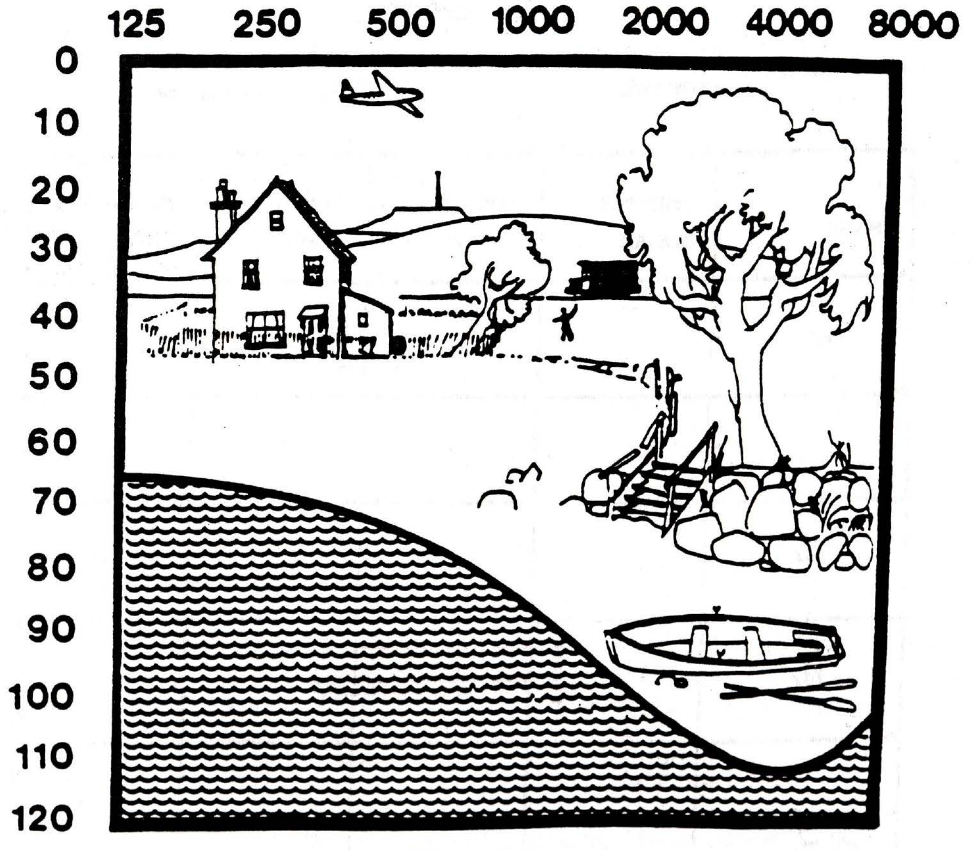


Figure 3.1

An audiogram in which the threshold is compared to a shoreline. The contour of the shoreline tells nothing about the quality of water; similarly, the threshold curve yields no information about the quality of hearing present at suprathreshold levels.

The Ling Sound Test may be used to judge the quality of the fitting. It may also be used to verify whether the aided audiogram is accurate. For infants this is an ideal method for determining the accuracy of functional gain on the audiogram. Speech signals are much more interesting than other signals (NBN & warbled tones) to the paediatric population. The benefits of earmould modifications or changes in hearing aid settings may be determined with the various sounds.

LISTENING CHECKS

An important concept in the use of amplification devices is the maintenance of the device itself. Devices are assessed using electroacoustic hearing aid analysers during clinical visits. An analysis of the hearing aid response characteristics are graphed and compared to a standard reference. The hearing aid either meets specifications or is referred for repair. Another application of the Ling sound Test has been the informal and indirect check of the frequency response of the hearing aid. This check is recommended to be completed daily to insure that the device is at its optimal performance.

For a quick listening check, the parent, teacher, or other, connects a stethoset or earmould to the hearing device via the ear hook or through the armould itself. While listening to the aid the Ling sounds are repeated. The vowel sounds and consonants should appear audible and clearly reproduced. If the sounds appear distorted, unnatural, or inaudible, the hearing aid may be malfunctioning and in need of repair. A check of acoustic feedback (turning the hearing aid on and listening for a whistling sound is an incomplete test. The same may be said for the use of 1,2,3, or /ba/, /ba/, /ba/. The entire frequency range of the aid has not really been assessed. Further the difficulties with distortion often are initially indicated in the higher frequency sound spectrum.

HEARING TEST FOR INFANTS

Hearing evaluations for the paediatric population require highly skilled testers. These evaluations may include electrophysiologic measures such as an auditory brainstem test (ABR), otoacoustic emissions (OAE's), electrocochleography (ECoG), and an immitance test battery. Behavioural tests may also used. These include those with and without reinforcement to an acoustic presentation. These evaluations will indicate whether a hearing loss is present and may define the type and degree of loss.

These tests allow specific identification of a portion of hearing. However they give little information regarding the infant or child's ability to use the most significant acoustic stimulus - speech. As another application, the Ling sound Test has been used as an informal measure to assess behavioural threshold.

It has also been used to provide initial information on the child's ability to detect sound and begin to develop higher levels of auditory language such as discrimination, identification and comprehension.

SCREENING TEST

It is a well identified scenario that parents, usually the mother, suspect a hearing loss long before the formal identification has been made. The first fear is that the child is intellectually handicapped. Eventually a hearing loss is suspected. At home the parents, inexperienced testers, try a variety of methods to determine how well the child hears. The results are usually inconclusive. At some point the child is taken to the nursing sister or GP. Eventually a referral is made to an audiologist.

Given these difficult populations to test, several tests have been devised by audiologists for use in the paediatric populations. These usually include calibrated noise makers or toys with a distracter. The distracter is used to keep the child's attention while the various noise makers are presented behind the child toward individual ears. The test are screening tests for hearing. However, the tests usually have a cost attached and also give little information about the child's response to speech stimuli. An additional application of the Ling Sound Test has been that of a screening test. At a normal conversational level, the sounds are presented behind the child and a response or non-response is noted. Failure to hear or detect any of the sounds would indicate a referral for a more thorough hearing evaluation by a paediatric audiologist

SUMMARY

As briefly described in this paper, the Ling Thing is a practical application of both the physical and psychological aspects of acoustics. The Ling Sound Test as the foundation of the Ling Thing has gained support and recognition around the world. Its ease of use and the minimal amount of time requirement is a strong advantage in assessing the paediatric population. These applied concepts of acoustics are standard elements in the training courses for teachers of the hearing impaired and deaf. The application to speech intelligibility is also well regarded in the field of education. Its many uses in informal assessments of audition and levels of hearing ability have made this a recommended method in the field of audiology.

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