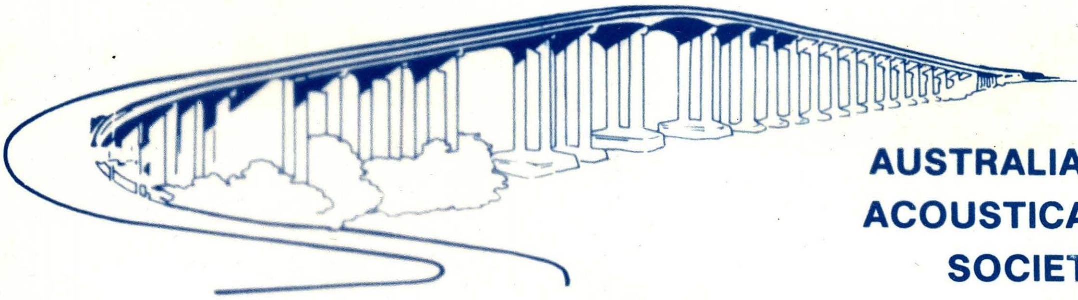

AUSTRALIAN
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ACOUSTICS IN THE EIGHTIES

ANNUAL CONFERENCE
HOBART 12-13 NOVEMBER 1987

Proceedings

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Acoustics in the Eighties:
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Session 4C: Transportation Noise

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Acoustics in the Eighties

Keynote Address

Acoustics in the Eighties

COMMUNITY NOISE AND PLANNING

T.E. Brown

COMMUNITY NOISE AND PLANNING

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ABSTRACT

The effects of noise on the community is an ever increasing intrusion on the living environment, and in addressing this problem scientific and engineering professionals must consider noise reduction strategies which will enable the increasing development of our planet's resources with a minimum of disturbance to the lifestyle of the population. The application of land use planning as a means of controlling noise must be considered along with other strategies, as more effective noise reduction can be achieved by this means than by all other strategies in relation to minimizing community annoyance.

1. INTRODUCTION

Man's capacity to create noise has increased dramatically over the last 200 years with increase in the size of communities and the rapid development of industry and transport systems. These 'improvements' to our living style, which are an integral part of progress, continue to add noise to the environment in which we live. In a 1972 USEPA report to the President and Congress it was estimated that in some urban areas noise was increasing at the rate of 1 dB per year or 10 dB per decade.

Surveys carried out in 1948 showed that 23% of those interviewed claimed to have been disturbed in their homes by external noises; in 1961 this figure had risen to 50%; and in 1986 a survey commissioned by the Australian Environment Council concluded that noise was the most serious form of environmental pollution perceived by residents in their homes.

Noise can be defined as 'sound which is undesired by the recipient'. This subjective definition is quite alien to a scientist whose description of noise is couched in terms of frequency and intensity. While the power, pitch and timing of sounds can be precisely measured and described, their 'noisiness' is a subjective quality.

Noise is an invisible but insidious form of pollution.

Sound sources today are increasingly powerful and ubiquitous, and the challenge to the engineer or scientist is to find ways of reducing the sound level or its information content to reduce the annoyance to the recipients.

The goal of a quieter community environment can only be achieved through the adoption of a co-ordinated noise abatement strategy. Measures will include:

(a) reduction of noise at the source;

(b) improvement in design of noise sources;

(c) introduction or enhancement of desirable sounds;

(d) comprehensive land use planning to achieve compatibility between noise sources and receptors.

The subject of noise and the community is complex, controversial and challenging and the introduction to the issues in this paper should set the scene for the many specific presentations which follow on the program.

Sound is mechanical energy from a vibrating surface, transmitted by cycling series of compressions and rarefactions of molecules of the material through which it passes. (Sound can be transmitted through gas liquid or solid media). A vibrating source producing sound has a total power output which results in sound pressure that alternately rises to a maximum pressure of compression and drops to a minimum pressure of rarefaction. The number of compressions and rarefactions of the molecules in a unit time is defined as frequency (expressed as Hertz (Hz) which is the same as cycles per second.)

Sound power (or pressure) do not provide practical units for sound or noise measurement for two reasons:

(a) the range of sound power and sound pressure is extremely large;

(b) the human ear does not respond linearly to increases in sound pressure.

Noise measurements are therefore expressed by the term 'sound pressure level' (SPL) which is the logarithmic ratio of the sound pressure to a reference pressure and is expressed as a dimensionless unit of power, the decibel (dB). To further utilize the sound pressure level results of measurement, in a manner which more closely represents the response of the human ear to changes, there have been various weightings and statistical factors applied to the decibel readings. The use of A-weighted sound levels, energy-equivalent noise levels, day-night loudness, and time above (TA) intensities, are all attempts to interpret results in a manner which approaches the subjective human response to unwanted noise.

The individual perception is variable until the level of organic damage is reached. Extremely loud sounds or prolonged exposure to high levels can cause deterioration of the auditory nerve, with a loss of sensitivity to sounds in the 2000 to 8000 cycles range. The damage begins for most individuals at continued exposures above 105 dB and rarely occurs below 85 dB. Below 85 dB different individuals have varying actuities for different frequencies.

(The human ear can distinguish frequencies ranging from 20 to 20,000 cycles per second or 50 to 10,000 at low noise levels. Ordinary street noise ranges from 40 to 8,000 cycles, while human speech is transmitted in the 100 to 3000 cycle range).

2. NOISE LEGISLATION

Noise legislation and legal aspects must be considered when approaching noise assessment and prediction problems. The legal aspects of community noise control range from public nuisance to specific Acts and Regulations which limit noise at the source or at the receptor location. The Australian Environment Council (AEC) Report No. 6 provides a summary of State legislation related to noise control, and the AEC Guide to Environmental Legislation and Administrative Procedures also details the various Acts and Regulations as they apply in each State. Local authorities also have powers to act under various By-laws, and Local Government Acts or Regulations in cases of public nuisance, or in approval processes associated with land use zoning or building codes, etc.

Consideration must therefore be given to legislative requirements of all levels of government which are applicable to noise control associated with any development proposal.

3. NOISE ASSESSMENT

Noise assessment may take the form of comparisons of existing and anticipated noise levels with the criteria established by the regulatory agencies, or in the absence of such criteria of existing community acceptance standards. The basic outline of the steps required for assessment of the noise regime associated with a specified area or development are:

(a) Determine current noise regulations;

(b) Describe existing noise levels;

(c) Determine the potential noise sources, mobile and stationary;

(d) Describe the noise characteristics of the project by measurement, comparison with a similar project, and/or modelling;

(e) Determine the noise sensitivity of the surrounding land uses;

(f) Assess the impact of the development on surrounding land uses;

(g) Develop mitigation measures for potential impacts;

(h) Discuss the noise considerations of alternatives to the project.

These steps are directed toward determining the noise impacts of proposed action or developments associated with individual noise sources. The issue becomes much more complex when multiple developments and associated noise sources, or when more than one receptor is impacted. The balance must then be met between expected abatement of noise to satisfy the various receptors, and available and/or economically acceptable noise attenuation methods which must be applied to the development to bring the noise down to an acceptable level.

3.1 IDENTIFICATION OF EXISTING AND POTENTIAL NOISE SOURCES

(a) Measurement of 'background' noise levels prior to any proposed development must include identification of existing noise sources. These may include highways, rail systems, existing equipment and industrial noise, natural sources such as rivers or streams, vegetation movement, animals etc.

(b) Development sources of noise to be identified must include both construction and operation phases:

(i) construction activities generally generate noise levels in excess of those from operating activities. Noise from construction varies relative to the site and the development proposed, but all phases including

- ground clearing,
- demolition and removal of existing structures, trees, etc.,
- excavation for foundations,
- erection of structures,
- finishing, including filling, paving and cleanup

contribute noise of varying intensity. The ground clearing, excavation and finishing tend to produce considerable noise annoyance. Foundation and erection phases tend to be somewhat quieter.

[3.1 (b) cont.]

(ii) commissioning and operation of a development have different sources of noise which tend to be associated with:

- transport based noise - roads, rail, aircraft, etc.
- industrial machinery,
- raw materials and product handling,
- transformer and other electrical sources,
- general steady continuous noise from operations.

3.2 DETERMINATION OF NOISE LEVELS OF VARIOUS SOURCES

This involves the collation of information on existing and expected noise levels associated with the area and the development.

(a) The existing noise levels in an area can be obtained by either measurement, or sources of data which are available associated with similar land use patterns.

(b) Expected noise levels associated with plant and machinery etc. can be obtained from manufacturing specifications or available noise levels charts for industrial machinery, equipment, and vehicles, if measurement of the real situation is not possible.

3.3 DEFINITION OF ACCEPTABLE STANDARDS AND CRITERIA FOR THE NOISE SOURCES AND RECEPTORS

The determination of acceptable standards for noise in a particular area is to reduce the undesirable effects of:

- interference with speech,
- causing hearing damage,
- causing disruption to sleep and rest,
- reducing work performance,
- causing annoyance to humans or animals.

(a) Interference with speech can impede activities and human relationships. If two people are three metres apart:

- normal speech can occur at less than 55 dB(A),
- the voice will have to be raised between 55 and 75 dB(A),
- shouting is necessary between 75 and 92 dB(A),
- speech communication is impossible above 92 dB(A).

(b) Hearing changes, damage, or loss can be caused by noise exposure:

- Temporary threshold shift (TTS) is a lessened ability to hear weak auditory signals. TTS increases linearly with the average noise level from about 80 to 130 dB(A) and is proportional to the length of exposure, thus steady noise is the major offender. Recovery can occur and may take a few hours or up to 4 weeks if exposure to the noise is removed.

[3.3 (b) cont.]

- Noise induced permanent threshold shift (NIPTS) is hearing loss from which there is no recovery. It is therefore a form of deafness. Unprotected exposures of 8 hours per day to noise above 105 dB(A) will produce NIPTS in almost all individuals; exposure to 95 dB(A) will cause effects in approximately 50% of individuals; and it is rare for NIPTS to be found at levels below 80 dB(A).

(c) Disruption to sleep - sleep interference is not quantifiable in tested individuals when a continuous indoor level of non-fluctuating noise at 35 dB(A) is maintained.

35 dB(A) with a fluctuation of 5 dB will produce response from about 10% of individuals.

40 dB(A) continuous equivalent indoor level has been shown to affect approximately 20% of individuals, and 50 dB(A) continuous equivalent indoor level will affect approximately 50% of subjects.

A sudden noise level of 80 dB(A) will awaken 90% of individuals.

An increase of 10 dB or more occurring in 0.5 seconds imposed on the continuous background level may lead to awakening of sensitive individuals.

(d) Reducing work performance - a steady noise, without special significance does not appear to interfere with most human activities that do not require acoustic information in order to be carried out.

Intermittent or impulsive noise has a marked disturbing effect. High frequency noise components (above 200 Hz) usually cause worse interference with performance than do low frequency components. Noise tends to affect the quality of work, more than the quantity, and complicated tasks demanding considerable concentration are more easily influenced by noise than simple tasks.

(e) Causing annoyance - in average town living conditions outside noise which emanates from transportation and industrial sources should be maintained within the folowing maxima to ensure the level of annoyance inside residences is kept to a tolerable level:

- Daytime L_{dn} of 45 dBA indoors (which implies an outdoor criterion of 60-65 dBA)
- Nighttime L_{dn} of 35-40 dBA indoors (which implies an outdoor criterion of 55 dBA Leq maximum).

(These noise ratings were developed by USEPA). A comprehensive set of recommendations are found in Australian Standard 2107 for internal noise levels. Attenuation of noise by a typical house can range from 10-20 dBA depending on whether windows and doors are open or closed.

4. COMMUNITY REACTION TO NOISE

The effect of noise on people is highly variable for individuals dependent on their sensitivity to noise generally or particular types of noise. Their responses may be triggered by:

(a) the activity they are engaged in at the time of the noise episode,

(b) interruption to the activity being undertaken (e.g. sleep, speech commnication, etc.),

(c) the duration or suddeness of noise may elicit variable response,

(d) the presence of tonal components from a particular source.

While it is possible to study the effects on people on an average or statistical basis, the response to a certain noise source by an individual will be highly variable and unpredictable. Community response can be considered using the various weightings and measurement formula as a means of providing an indication of expected reaction to particular noise episodes. Careful interpretation of this information can provide the basis for preparation of noise abatement strategies associated with each particular source, and the probable community acceptance of the resulting noise levels. Individual reaction to the noise and acceptance of the decisions made using the 'average' response criteria will however always provide the exception to the rules.

In general, community reaction increases in magnitude and intensity as the noise levels increase:

Below 45 dBA	No reaction.
45-55 dBA	No reaction, although noise is generally noticeable.
55-60 dBA	Sporadic complaints.
60-70 dBA	Widespread complaints.
70-80 dBA	Complaints with threats of legal action or strong appeals to stop the noise.
Above 80 dBA	Vigorous community action.

The sensitivity of the community noise also changes with -

(a) lifestyle alteration - examples of this are found in areas where 5 acre subdivisions occur and the country style of life is expected even when the area maybe adjacent to a rural activity - e.g. timber based industry, primary produce based processors (milk factories with truck movements, driers, etc.), abattoirs, etc.

(b) age variation - increase in age was shown in the AEC 1986 noise survey to be associated with a decrease in the reaction to noise,

(c) education level - AEC survey showed that an increased reaction to noise was consistent with individuals having higher educational training,

(d) change to noise levels or information can result in reaction from the community even if the level has been reduced overall,

(e) intolerance to existing noise, when individuals move into an area, is often experienced with the 'newcomers' expecting retrospective change to meet their expectations of an area,

(f) the zoning of land which allows for certain development to take place at some future time, does not mean that the existing community will accept development when it is proposed. This is a consistent source of basis for appeal against development in 'their' area.

5. NOISE CONTROL PRACTICES

There are two basic ways of solving the contentious issue of noise -

(a) by eliminating the noise at its source,

(b) by separating the noise source from its receptors.

To implement the above options three basic principles may be applied -

(a) reduce vibration (and thus the noise) at the source,

(b) enclose the noise source,

(c) attenuate the noise by absorption.

The reduction of noise at the source can be applied most effectively and economically by the substitution of quieter equipment, processes, or improved item design at the time of project planning. Alteration of equipment once a development is in operational phase is not only costly in terms of changing items of equipment, but also in loss of production during change over.

The enclosure of noise source(s) can be effectively considered at the site planning stage by locating noisy items within a development so that the construction of the project as a whole provides significant attenuation for the noise source. Where locational noise control is not possible or was not included in the site planning, insulation of buildings and isolation of noise sources within the buildings has to be achieved. This latter option can be used where unexpected noise problems occur and noise abatement can usually be achieved at a cost.

Attenuation of noise by absorption can be achieved by use of berms, barriers of various kinds, or by providing sufficient distance between the source and the receptors through land use planning principles.

6. LAND USE PLANNING

Considering the aforegoing discussion, it is obvious that there is a need to place greater emphasis on land use planning as an instrument in noise attenuation. Comprehensive planning must include standards for community noise in order to assist the planning and locating of noise sources and receptors. Standards need to be explicit, quantitative, and <u>used</u> by authorities and the community if the objective is to be achieved.

The implementation of a noise abatement strategy, through control of sources and land use changes involves a balancing of interests between those who wish to engage in activities that will result in noise, and those who wish to maintain the quality of their environment. This balancing of interests must take place in a changing economic, social, political and technological environment.

Noise reduction strategies in existing builtup areas can be prohibitively expensive. The least expensive approach is to avoid incompatible land uses through forward planning of developments, ensuring adequate buffer zones around various proposals.

Examples of planning options which can provide noise reduction from developments are:

- decentralization of central business districts to reduce traffic congestion with its associated noise and air pollution;
- establishment of new regional growth centres;
- promotion of inner city renewal with well planned residential areas to enlarge 'quiet zones';
- a balanced approach to planning of airports and roads through residential and institutional areas;
- separation of industrial areas from residential areas with noise corridors maintained to ensure that intrusion of either development does not occur;
- establishment of adequate buffer zones around developments which are proposed in rural zones;
- co-ordination of land use and transport planning and integrated urban planning is vital to ensure that high noise levels which cannot be obviated occur only at sites that can accommodate it without causing inconvenience to the community.

When transport developments are proposed a number of factors should be taken into account to ensure that noise is kept to a minimum:

- land use adjacent to prospective highway alignment should be carefully assessed;
- traffic and adjacent land uses should be separated as much as is economically feasible;

- use of service roads parallel to freeways serve the dual function of providing access and also increasing the distance from the noise source and the receiver;

- elevated or inclined roads should be avoided in urban areas due to the increased noise from this design;

- rough asphalt and grooved concrete pavement should be avoided as these may add 10 dB(A) to noise emission;

- bypasses and ring roads help divert traffic and reduce congestion, with a resultant reduction in noise.

Industrial developments when proposed must take into consideration the numerous components of noise generating items if acceptance of the project is to be gained with minimum impact, and delay:

- ensure that planning scheme zoning provides adequate cover for the community and the project to exist together;

- identify all noise components and assess realistically in relation to community response;

- consider transport aspects of the development - these can have large areal impacts beyond the project boundary;

- use best practicable technology to ensure that noise impact is kept to a minimum. (Costs on quieter equipment may be far more cost-effective than noise attenuation costs);

- consider inclusion of a buffer zone around the project to ensure that future expansion can be achieved and that residential development cannot encroach into the noise impacted area around the plant.

7. CONCLUSION

Effective application of land use planning principles can yield greater benefits than any other means of noise control. The principal barrier to effective planning has been the fragmentation of responsibilities between various authorities in the various States. Many States now have a close relationship between planning and environment which will help overcome some of the problems associated with fragmentation of responsibilities and expertise and the preparation of planning schemes which include consideration of environmental planning will further help set out a 'sound' basis for development which is acceptable to the community. The use of discretionary approvals for proposals, within planning schemes, which are often not adequately assessed before the development is approved, result in incompatible land use situations in spite of satisfactory planning principles being applied to the overall plan for an area.

Noise levels suffered by a community can be reduced at a price. That price includes -

- noise emission constraints on sources;
- cost of shielding transport systems from urban development;
- reduction in the density of urban and industrial groupings to accommodate buffer zones;
- restrictions on use of excessively noisy equipment.

Acoustics in the Eighties

Session 1A: Community Noise

Acoustics in the Eighties

COMMUNITY NOISE SURVEYS — WHAT DO THEY TELL US?

W.D. Renew

COMMUNITY NOISE SURVEYS - WHAT DO THEY TELL US?

W.D. RENEW, M.A.A.S.

ABSTRACT

This paper describes the major community noise surveys carried out in Australia since 1974. Comparisons are made of survey techniques and of the values of the major noise descriptors in noise area categories specified in AS 1055-1984. Descriptions are given of the more usual methods of data presentation including noise probability distributions and noise spectra. Some discussion is given of accuracy and reliability of measurement.

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1. INTRODUCTION

A community (or urban) noise survey is generally carried out to evaluate environmental noise levels in a city. The information obtained may be required to act as a basis for land-use planning, to enable comparisons to be made between current and regulated noise levels, and to estimate the noise exposure of community groups.

In Australia, community noise surveys have been conducted in several major cities since 1974, in the main by State environmental control agencies (Renew 1986a). The objectives of the surveys are shown in Table I. A classification of these surveys is given in Table II, based on the work of Brown (1987). It is apparent that the most popular objective has been Objective A, the evaluation of current environmental noise levels.

TABLE I

OBJECTIVES OF THE NOISE SURVEYS

Class	Objective
A	To evaluate current environmental noise levels
D	To provide a basis for measuring future changes
E	To provide noise level data against which to measure community response
F	To provide data to assist in setting levels for legislation and planning
H	To determine changes in noise level

2. SURVEY METHOD

2.1 TYPES OF SURVEY

Brown (1987) identifies the following types of survey based on the method of spatial sampling of noise levels in the urban field:

(a) random sampling
(b) sampling by land-use categories
(c) receptor-oriented sampling
(d) source-oriented sampling.

TABLE II

SUMMARY OF COMMUNITY NOISE SURVEYS

Survey Year	*Study Location*	*Objectives* A	B	C	D	E	F	G	H	*No. of Sites*	*Spatial Sampling*		*Temporal Sampling*
<u>Surveys using sampling by land-use categories</u>													
1974	Brisbane	A					F			25	4 town planning zones	arbitrary - avoided transport sources	24h at 20 min/h
1979	Sydney	A			D					6	R2 areas	arbitrary	24h x 3 days
1982	Adelaide	A					F			24	typical areas	arbitrary - avoided main roads and rural areas	24h
1983	Brisbane	A							H	25	4 town planning zones	similar to 1974 sites	24h x 3 days
1984	Sydney	A							H	4	R2 areas	similar to 1979 sites	24h x 3 days
1986	Brisbane	A				E				27	4 town planning zones	similar to 1983 sites	24h
<u>Surveys using random sampling</u>													
1976-7	Melbourne	A				E				40	selected from grid		24h
1985	Toowoomba	A								496	selected from grid		2 x 10 min during day

It may be seen in Table II that the first two sampling methods have been employed in Australian surveys and that sampling by land-use categories has been the more popular method. This is a result of the inclusion of six noise area categories (R1 to R6) in a table of estimated mean background sound levels in issues of AS 1055, Acoustics - Description and Measurement of Environmental Noise (Standards Association of Australia 1984). These categories are based upon the density of transportation and the extent of commerce and industry in the areas under consideration. They are often regarded as land-use categories. Stratification of the urban noise field has been accomplished by arbitrarily choosing a number of measurement sites in each noise area category considered to be typical of the neighbourhood.

In the 1985 Toowoomba survey (Eddington 1986) the 496 survey sites were randomly selected on a grid with a spacing of 364 m while a 1000m grid was used in the 1977 Melbourne survey.

2.2 SITE SELECTION

The number of sites selected in the land-use categories for each survey are shown in Table III. Intermediate ratings of land-use were employed in the Melbourne and 1974 Brisbane surveys in an attempt to improve the method of classification. It is apparent that there is considerable variation in the number of sites chosen and that in general these numbers are too low to guarantee a high level of accuracy.

TABLE III

NUMBER OF SITES IN SURVEYS

AREA RATING R	NUMBER OF SURVEY SITES						
	BNE 1974	MEL 1977	SYD 1979	ADE 1982	BNE 1983	SYD 1984	BNE 1986
1	2	-	-	1	-	-	4
2	4	11	6	5	9	5	5
2 1/2	-	4	-	-	-	-	-
3	8	1	-	2	7	-	5
3 1/2	-	2	-	-	-	-	-
4	4	9	-	5	6	-	5
4 1/2	2	7	-	-	-	-	-
5	3	6	-	5	2	-	4
6	2	-	-	6	1	-	4
TOTAL	25	40	6	24	25	5	27

It was found difficult in certain surveys to select very quiet (ie. R1) areas. In addition, it was no easy task to find sites in areas which could safely be termed 'predominantly industrial' and hence substantiate an R6 rating. In the 1986 Brisbane survey (Duhs) special emphasis was placed upon distance to major traffic routes during the selection of R6 sites.

With regard to accuracy in site rating, Renew (1986b) reported the results of a test carried out by six officers during the 1983 Brisbane survey using the rating system employed in the Mebourne survey. The maximum standard deviation of a mean area rating at a site was 0.8 while the maximum standard deviation for a category mean was 0.3. These results indicate that consistent area ratings can be obtained.

2.3 SCOPE OF SURVEYS

The scope of a survey depends upon several factors, amongst the most important of which are the time allowed for its completion, the number of staff available to carry it out and the objectives of the survey. A comprehensive survey would be expected to entail the evaluation at all sites of:

(a) statistical noise descriptors such as L_1, L_{10}, L_{50}, L_{90}
(b) energy-based noise descriptors such as L_{eq} and L_{dn}
(c) octave band noise spectra.

This information would normally be obtained for specific time periods so that comparisons could be readily made with data from similar surveys. In Australia the time periods employed in community noise surveys have generally been those specified in AS 1055.

Temporal sampling has usually been carried out on a continuous basis at hourly intervals over at least a 24 hour period at each site. As may be seen in Table II, smaller sampling periods were used in the 1974 Brisbane and Toowoomba surveys. Although the statistical and energy-based descriptors mentioned above were obtained in the majority of Australian surveys, noise spectra were produced in only the 1983 Brisbane and 1984 Sydney surveys.

3. ANALYSIS OF DATA

3.1 APPROPRIATENESS OF SURVEY METHOD

Brown (1987) has questioned the suitability of current survey methods for achieving the objectives of the surveys. With regard to surveys in which sampling is carried out by using land-use categories, he concludes that the categories should be based on proximity to transportation. Results of this type of survey will be acceptable if it can be shown that the variability of noise levels at sites within land-use categories is significantly less than the variability between the mean levels of noise in the categories.

One-way analysis of variance was carried out to test the appropriateness of the sampling method used in five of the surveys. Significant results were established for the 1974 and 1986 Brisbane surveys and the Melbourne survey for all noise descriptors tested. In the Adelaide survey, results were significant ($p < 0.01$) for L_{50} and L_{90} during the morning and night periods. The high levels of significance obtained for some of the surveys would be the result of the assessor's perception of the noise climate at the sites, as well as his assessment of the land use and the density of transportation in the neighbourhood.

3.2 MEAN NOISE LEVELS

Values of mean noise level in the standard time periods were calculated from the hourly data. It is apparent from the L_{90} values for R3 areas in Table IV that the values of estimated background levels in AS 1055 have on occasions been exceeded. The surveys do not indicate any increasing trend in noise levels.

TABLE IV

L_{90} NOISE LEVELS FOR R3 AREAS

AS1055 1984	BNE 1974	MEL 1977	ADE 1982	BNE 1983	BNE 1986
45 (Mng)	40.8	44.1	42.6	44.8	41.6
50 (Day)	42.0	54.7	41.9	45.6	41.1
45 (Evg)	40.7	50.4	41.2	43.1	38.3
40 (Ngt)	35.1	34.1	37.0	38.8	32.7

Values of standard deviation calculated for each time period and for each area rating showed considerable variation among the surveys. The range in standard deviation for daytime L_{90} values was 0.2 to 8.2 dB(A) with a mean value of 2.9 dB(A). There was a tendency for the maximum variation in values of L_{90} to occur during the evening period and the minimum during the day. The mean value of standard deviation for L_{dn} was 4.1 dB(A). According to Sutherland (1986), readings would be required over six and nine days respectively in order to obtain mean values of L_{90} and L_{dn} with a 95 percent confidence limit of $\pm$ 3 dB(A).

Renew (1986a) reported a relatively low variation in the mean daily L_{90} value in the standard time periods over three days. The standard deviation for all area ratings during the 1983 Brisbane survey ranged from 0.3 to 2.9 dB(A). During the 1986 Brisbane survey noise levels were monitored at an R3 site over a period of seven consecutive days. The value of the standard deviation of L_{90} was found to range from 1.0 to 3.4 dB(A) and of L_{eq} from 1.4 to 6.6 dB(A).

3.3 REGRESSION ANALYSIS

Renew (1986b) has described the regression analysis carried out to determine how closely survey data corresponded with estimated background noise levels for area categories specified in AS 1055. Values of the correlation coefficient were found to be generally significant at the 95 percent confidence level for mean L_{90} and L_{eq} values. Lower levels of significance were determined for mean values of other noise descriptors. The calculated standard error values indicated that, for certain of the surveys, the data were rather dispersed and accordingly not highly reliable. It is apparent that more sites are required in order to achieve more reliable results.

3.4 CUMULATIVE NOISE DISTRIBUTIONS

The cumulative noise distribution is useful in illustrating the statistical makeup of the noise climate at a location in a given period. It is particularly of use in comparing past and current noise climates. In the U.S.A. the Department of Housing and Urban Development has adopted daytime outdoor noise criteria in terms of the cumulative noise distribution. Noises are classified as 'clearly acceptable', 'normally acceptable', 'normally unacceptable' and 'clearly unacceptable'. Challis (1982) has prepared a similar set of criteria for night-time noise levels. Renew (1986a) reported that, in the 1974 and 1983 Brisbane surveys, the cumulative noise distributions were located at worst in the 'normally acceptable' region during the day and night. A comparison of survey distributions is given in Fig. 1.

3.5 NOISE SPECTRA

According to Sutherland (1986), the noise frequency spectrum gives a more complete description of the temporal pattern of noise level fluctuations than is provided by statistical descriptors. In addition, the analysis of noise into its octave (or one-third octave) band components gives valuable information on the major noise sources contributing to the overall noise climate. Again, the noise spectra from proposed developments can be examined before construction to ascertain whether they will unduly affect current ambient noise spectra.

In the U.S.A., some local noise ordinances specify limiting values of noise spectrum components. Sutherland (1986) describes daytime median (L_{50}) noise spectra averaged over 350 sites in the U.S.A. The spectra are similar in shape to those obtained in the 1983 Brisbane (Renew 1986) and 1984 Sydney (Kotulski 1986) noise surveys. Typical noise spectra are shown in Fig. 2.

4. CONCLUSIONS

A description has been given of the results of community noise surveys carried out in Australia. The major objective, the evaluation of environmental noise levels, has been achieved by the determination of statistical noise descriptors. The appropriateness of the survey method using the site selection categories of AS 1055 has been validated. A description has been given of the usefulness of cumulative noise distributions and noise spectra in increasing knowledge of the urban noise field.

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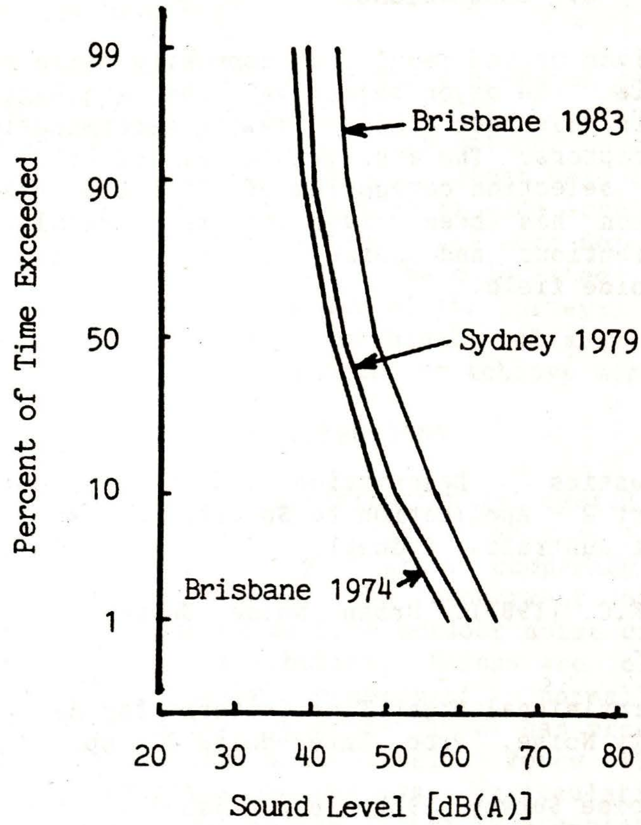

Fig. 1. Cumulative Noise Distributions
R2 Areas Daytime.

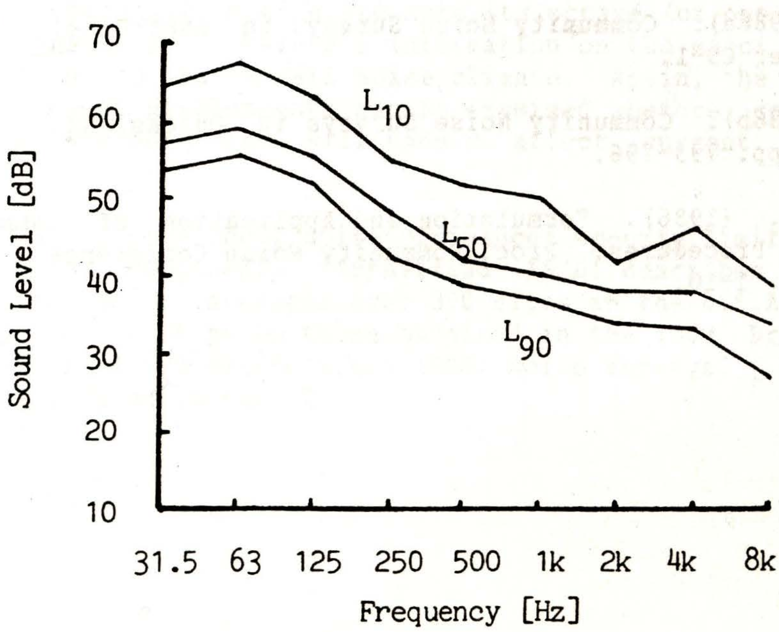

Fig. 2. Noise Spectra - 1983 Brisbane Survey

Acoustics in the Eighties

BACKGROUND NOISE LEVELS FOR COMMUNITY NOISE ASSESSMENT

M. Eisner, E. Taylor and M. Burgess

BACKGROUND NOISE LEVELS FOR COMMUNITY NOISE ASSESSMENT

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ABSTRACT

The method used for the assessment of offensive noise in most parts of Australia is based on a comparison between the background noise levels, ie when the potentially offending noise is not present, and the levels with the noise present. When measurements are possible, the background levels are considered as the L_{A90} *and the noise levels as the* L_{A10}. *This paper discusses situations where the use of* L_{A90} *does not give a true indication of the type of background noise in the area. Consideration of the noise climate based on the* L_{A10} *and the* L_{A90} *for both the noise present and noise absent situations is discussed as a fairer method for noise assessment.*

1. INTRODUCTION

The assessment of the annoyance, as opposed to the damage, likely to be caused by a noise can be determined in two main ways:

- comparison of the noise level with a value which has been specified as the maximum acceptable noise level for the area, or
- comparison of the noise level with a background noise which is either measured or calculated on the basis of the type of area and time of day

For most environmental noise problems it is usually the second method which has been applied by the appropriate Australian authorities, for example the Noise Control Guidelines produced by the State Pollution Control Commission in NSW (1985). This comparative method is also the procedure which was specified in the original AS 1055 (1975), "Noise Assessment in Residential Areas".

The concept of the background noise in the area being an all pervading noise which the residents have been exposed to and is therefore part of their environment is very appealing for noise assessment situations. However background noise is not always "well behaved" and the use of it in a comparative method can become quite complex. Some situations in which the simple comparative method, as it is currently applied by the authorities, seem to be unfair to either the complainant or the "offender" will be given in this paper with particular reference to case studies. This is essentially a discussion paper which seeks to raise suggestions for alternative methods of assessment.

2. BACKGROUND NOISE

The Standard AS1633-1985 "Acoustics Glossary of terms and related symbols" defines background noise as:

> *the level of the ambient sound indicated on a sound level meter in the absence of the sound under investigation.*

and ambient sound as:

> *all encompassing sound at a point being a composite of sounds from near and far*

This definition of background sound is not specific about the descriptor which is to be used to determine this *"level"* , ie the L_{10}, L_{50}, L_{90}, L_{eq} etc. However reference is made in the Glossary to AS 1055 (1984) which has a definition for Background A weighted sound pressure level $L_{Abg,T}$:

> *the A weighted sound pressure level obtained using the time-weighting F and arithmetically averaging the lowest levels of the ambient sound pressure levels measured in the absence of the noise under investigation, during the time interval considered.*

A Note to this definition states that an approximate equivalent to $L_{Abg,T}$ may be taken as $L_{A90,T}$.

3. ASSESSMENT METHODS

Any assessment of noise annoyance should be based on the intrusiveness of the noise. This is not a simple matter to determine and many subjective studies have been carried out around the world. A recent study by Preis (1987) proposed a rating,L, which would allow for the intrusiveness of sounds, of the form:

$$L = L_x + L_u$$

where L_x is one of the presently used noise ratings

and L_u is a correction for sound intrusiveness which needs to be based on an expression of the form:

$$L_u = C_1 \text{ x (number of components below the masking threshold)} + C_2 \text{ x (the measure of dissonance)} + C_3 \text{ x (frequency of the component with the maximum level)}$$

Preis then concluded that the values of the coefficients C_1, C_2 and C_3 require further experimental study.

Any authority has to have a practical method for noise assessment which can be used without the need of highly trained personnel and expensive/complex equipment. A comparitive method based on measurements (or calculations) of the noise levels with and without the noise is such a practical method of noise assessment, for example the NSW State Pollution Control Commission Environmental Noise Control Guidelines (1985). A typical example of the noise assessment is that stated for domestic air conditioners:

> *the broadband intrusive noise resulting from its use (L_{A10} measured for not less than 15 minutes) should not exceed the background level (L_{A90}) by more than 5dB(A) when measured at the receiver boundary.*

In the majority of potential noise problems this approach is quite reasonable and fair but there are situations where the background noise has particular characteristics or is of such a variable nature that the comparison, with the allowance of 5dB, is not as simple.

4. ALTERNATIVE METHODS FOR ASSESSMENT

We suggest that for any potential noise problem the assessment should be based on an examination of the range of levels for both the "noise on" and "noise off" situations. In this way a better appreciation of the existing noise in the area can be obtained. The effect of the noise on this environment can then based on criteria which utilise the range of noise levels in both situations.

It does not necessarily follow that such measurements would take more time and be more expensive. The measurement procedure which utilises the determination of the "average of the maximum pointer deflections" and "the average of the minimum pointer deflections" from a simple sound level meter can be used in both the "noise on" and "noise off" situations. Modern instrumentation allows for rapid determination of the percentile and equivalent energy values directly so the information can be obtained with little extra time.

There are many areas where the range of "noise off" levels are in excess of 5dB(A). In a report on part of the Syney Ambient Noise Study, Kotulski (1986) presented the long term average percentiles for sites in R2 areas (areas with low density transportation) for three time periods. His results, rounded to the nearest dB(A), are shown in the following table:

	L_{10}	L_{90}	$L_{10} - L_{90}$
daytime	52	34	18
evening	46	34	12
night	40	33	7

These values represent the averages over the four sites but the data for the individual sites showed the same trends, namely the existing L_{10} exceeded the L_{90} by more than 5dB(A) except at one site for which the difference during the night period was 4 dB(A). How can the assessment of the noise impact of any potentially noise producing facility or industry in this type of area be based on the simple comparison between the L_{10} for the "noise on" situation and the L_{90} for the "noise off" situation

A typical case of a noise assessment problem for which a more detailed examination is required occurred following complaints about noise emanating from a club when live bands were

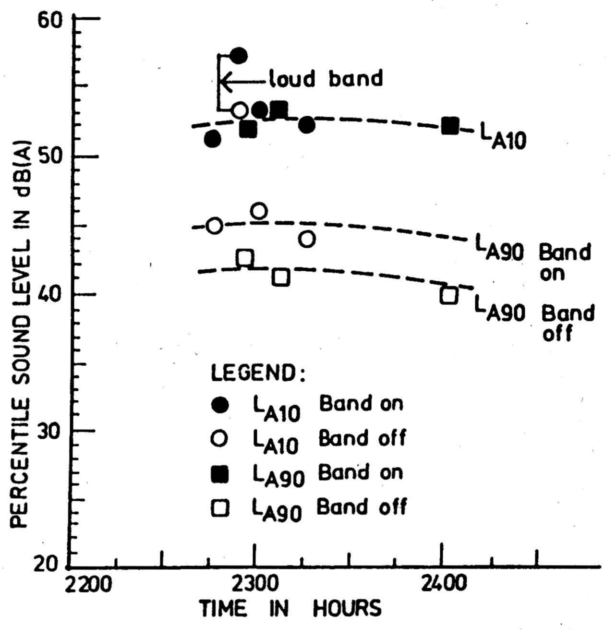

Figure 1 Percentile noise levels at the nearest residence when the band was and was not performing. Note the one occassion when a particularly loud band was performing.

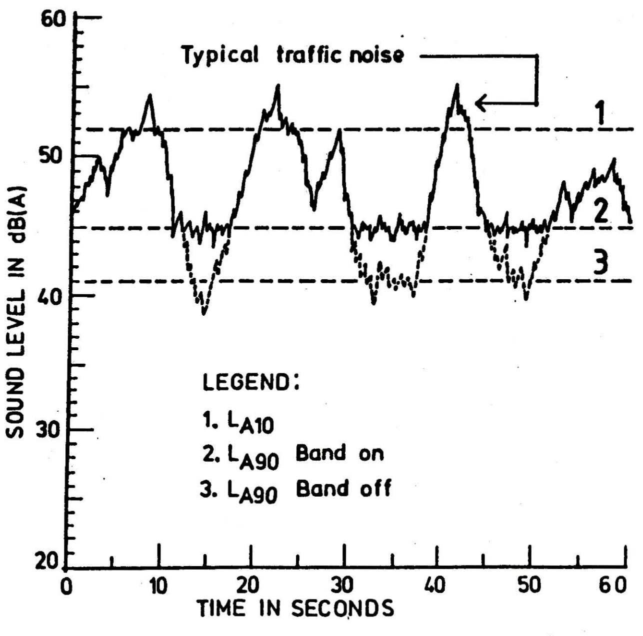

Figure 2 Typical noise levels for the area with and without the noise from the band. The higher levels were produced by traffic on the nearby road.

performing. The complainant's home was 220m from the club, on the other side of a main access road. The traffic on this road decreased throughout the evening but there were still a number of heavy vehicles which led to a range between L_{10} and L_{90} when the club did not have any live bands performing of more than 5dB(A), as shown on Fig.1. When the bands were performing the noise from the music was below the peaks of the road traffic noise and the L_{90} was only raised by 4dB(A). For a typical time period the comparisons between the noise levels with the band operating and not operating are shown in Fig.2. It can be seen from these chart records that the noise from the band simply filled in some of the time between vehicle passbys. However on the basis of the simple comparison the noise from the club could be considered unacceptable.

Another example of this situation arose following complaints about the noise from a pool motor. The site was near a minor road and the L_{10} with pool motor operating was 6dB(A) above the background L_{90}. However more complete noise data is shown in the following table:

	L_{10}	L_{90}
motor not operating	49	43
motor operating	49	44

In this case it can be seen that the motor does not contribute to then overall noise levels in terms of dB(A). Even if the characteristics of the sound are considered to be tonal so that an adjustment of 5dB(A) is applied, the real, effect on the noise environment must only be marginal while the simple comparative method of adding 5 to the L_{10} of 49 would lead to an adjusted L_{10} of 54 and an apparent excess of 11 dB(A).

Another controversial aspect of noise assessment in quiet areas is the use of the "acceptable" background level of 30dB(A) which applies under the SPCC(1985) guidelines when the actual L_{90} is less than this value. An example of an application of this approach is given by the following case study. In an investigation prior to a development in a rural area and L_{90} of 22dB(A) was measured. The estimated noise levels from the development were 33-35dB(A) with no adjustment necessary for noise character. In this case the noise falls within the acceptable range using the "acceptable" background noise level of 30dB(A) but it is 11-13 dB(A) above the existing background noise levels. The decision about whether the residents are likely to be annoyed or disturbed by the noise cannot really be made on such a simple comparison.

5. CONCLUSIONS

The assessment of noise disturbance is very complex, especially in very quiet locations. A simple comparative method based on only the L_{10} and the L_{90} does not really give adequate consideration to the range of noise existing in the area. The real question, which does not seem to have been resolved, is what is the magnitude that noise becomes disturbing to a significant number of people. Until that is resolved every attempt should be made to examine all the aspects of the noise environment which can be measured, bearing in mind the practical limitations of "in the field" measurements.

The intention of this paper was not to provide the answers but to raise some questions for future discussions.

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Acoustics in the Eighties

COMMUNITY REACTION TO OVERPRESSURE FROM BLASTING

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COMMUNITY REACTION TO OVERPRESSURE FROM BLASTING

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ABSTRACT

Overpressure from blasting has been identified for some time as causing damage to buildings and annoyance to building occupants. A survey has been carried out around a quarry in Sydney which is surrounded by residential areas to quantify the degree of annoyance caused by overpressure. Overpressure measurements were made during a series of blasts and the residents were surveyed to determine their response to the blasts. A correlation between overpressure level and the percentage of respondents highly annoyed was obtained and even at a level of 115 dBL a significant percentage of respondents was highly annoyed.

ACKNOWLEDGEMENT: The Author acknowledges the valuable contribution to the Study by Dr Gayle Avery who designed and organised the Social Survey and the State Pollution Control Commission of New South Wales who commissioned the Study and gave permission to publish the Paper.

1. INTRODUCTION

Blasting in open cut mines and quarries generates overpressure which propagates through the air as well as vibration which is transmitted through the ground. Overpressure is similar to sound excepting that it includes an unusually high amount of low frequency energy and takes the form of a shock wave.

Whilst complaints are sometimes received from residents about the effects of overpressure, little is known about the cause of the complaint and the level at which Community reaction occurs. Almost no research is being carried out, including in Australia, into the subjective reaction of residents to overpressure.

To assist in setting overpressure level limits for open cut mine and quarry blasting, the State Pollution Control Commission of New South Wales commissioned a study into the response of residents to overpressure. The Study which was carried out jointly with Dr Gayle Avery of the New South Wales Institute of Technology established overpressure levels around a quarry which was surrounded by residences, surveyed residents potentially affected and analysed the results to obtain community reaction to overpressure.

2. OVERPRESSURE LEVELS

Overpressure levels generated around the quarry during blasting were measured on a number of occasions with the aim of determining a typical maximum overpressure level for each household visited during the Social Survey.

Since blasts occurred mainly each week or fortnight, only a limited number of blasts were monitored and the accuracy of the overpressure level assigned to each household was therefore limited.

2.1 MEASURE OF OVERPRESSURE

It was decided to limit the Study to the use of unweighted decibels (dBL). This measure was at the time (and still is) being used widely for overpressure assessment throughout Australia and other countries. It appears to allow adequately for the low frequency energy, but does not take into account the frequency of occurrence of blasting.

2.2 TYPE OF BLASTING

Well controlled bench blasts were regularly carried out and these were fired electrically using millisecond delays. In addition, secondary blasting of rock on the quarry floor was also carried out involving small charges with no stemming.

2.3 TEST EQUIPMENT

Most overpressure measurements were made using a Bruel & Kjaer Impulse Precision Sound Level Meter, Type 2209 set on 2 Hz lower cut-off frequency. On some occasions at some locations, recordings of the Sound Level Meter output were made on an FM Recorder.

The use of the Type 2209 Sound Level Meter means that overpressure below a frequency of 2 Hz was not accurately measured or recorded. However, significant energy below 2 Hz was not expected from the quarry blast.

The recordings of overpressure were later analysed on a Cathode Ray Oscilloscope and a Narrow Band Analyser.

2.4 MEASUREMENT LOCATIONS

Measurements were made at four locations around the quarry at distances that ranged from 200 - 400 m from the section of the quarry being worked. The four locations were generally spread around the quarry.

2.5 OVERPRESSURE LEVEL CONTOURS

From the results of monitoring, contours of overpressure level showing the maximum levels recorded were derived. From these, maximum overpressure levels were assigned to each household in the area visited during the Social Survey.

A narrow band frequency analysis of an overpressure recording is shown in Figure 1. This spectrum shape clearly shows the high low frequency energy typical of blasts. It also shows that the energy drops off significantly below approximately 5 Hz indicating that the use of the 2209 Sound Level Meter had a negligible effect upon the results obtained. In fact, other spectra obtained indicated slightly higher dominant frequencies.

In addition to overpressure measurements, a number of groundborne vibration measurements were made and these revealed levels typically 0.04 mm/s up to 0.8 mm/s. Since these levels are below the threshhold of vibration perception, it is considered that any community reaction was related to overpressure alone.

3. SOCIAL SURVEY

The aim of the Social Survey was to determine what aspects of blast overpressure caused disturbance and what relationship existed between overpressure levels and community response.

Representing all households in the 96-110 dBL zone, 206 households surrounding the quarry were approached for interview. Effective interviews were carried out with representatives of 170 households.

The questionaire used a blind approach where questions were asked firstly about the immediate environment in general. Ultimately, specific questions were asked regarding the interviewees' response to blasting. To assist with quantifying the response, an "opinion thermometer" designed by Dr Andy Hede of the Victorian EPA was used.

4. RESULTS OF SURVEY

Although extensive analysis of the survey results was carried out, only the main results are discussed here. The survey indicated a very wide spread of reaction to overpressure level. At all levels of overpressure, overall annoyance ranged from none to high.

A significant correlation ($r = 0.89$) was found between overpressure level and the percentage of respondents highly annoyed; the percentage highly annoyed increasing significantly with level over the range of overpressure investigated. The relationship is shown in Figure 2. In this Figure, the regression line has been drawn ignoring the result obtained at an overpressure level of 111 dBL, since this result was based on only 10 respondents, a much smaller sample than at other overpressure levels.

At 115 dBL, the regression line indicates that approximately 50% of residents would be highly annoyed.

A number of aspects were identified in the survey as contributing to resident response, as follows (in words used in the Questionaire):

(a) The loud noise.

(b) The shaking and rattling of the house and things within it.

(c) The likelihood of damage to the house.

(d) Startling of the occupants.

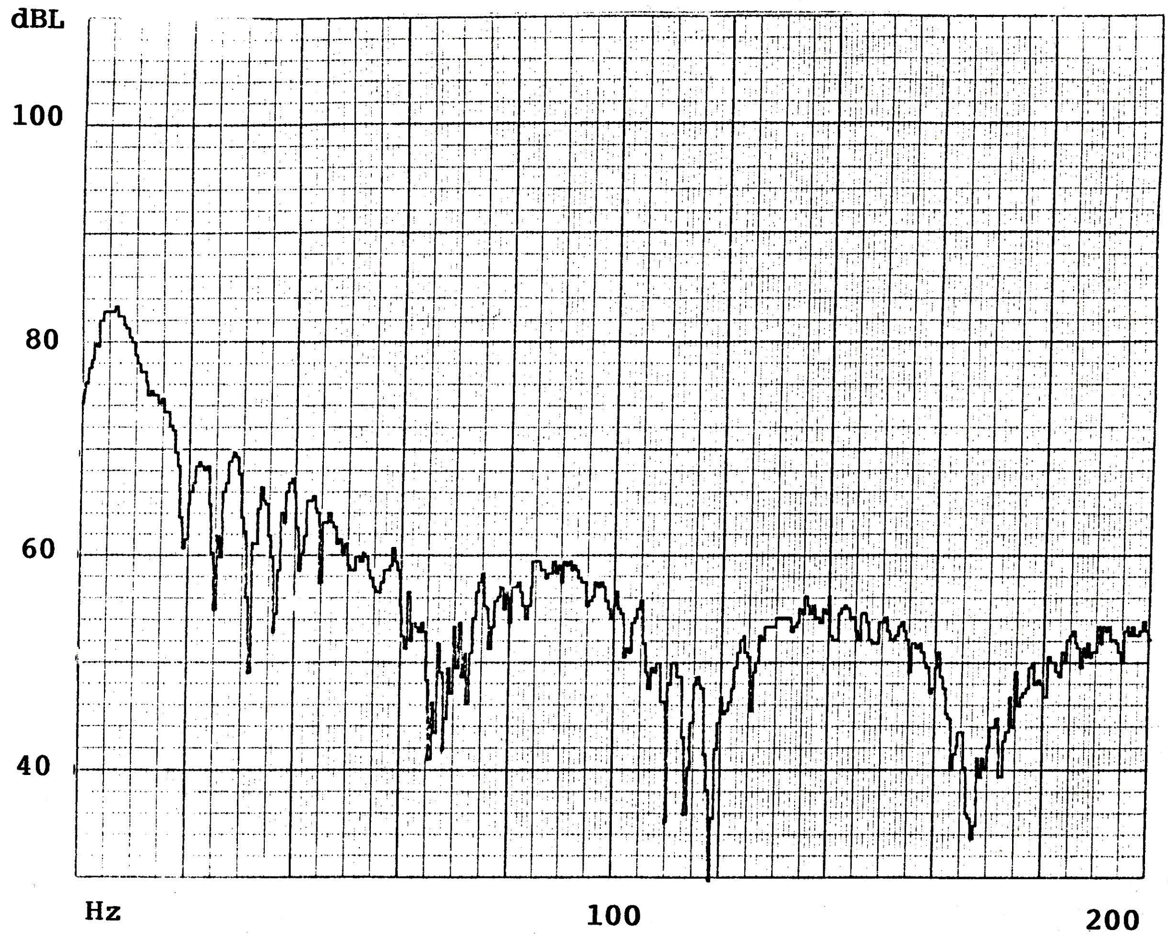

FIGURE 1 FREQUENCY SPECTRUM OF OVERPRESSURE

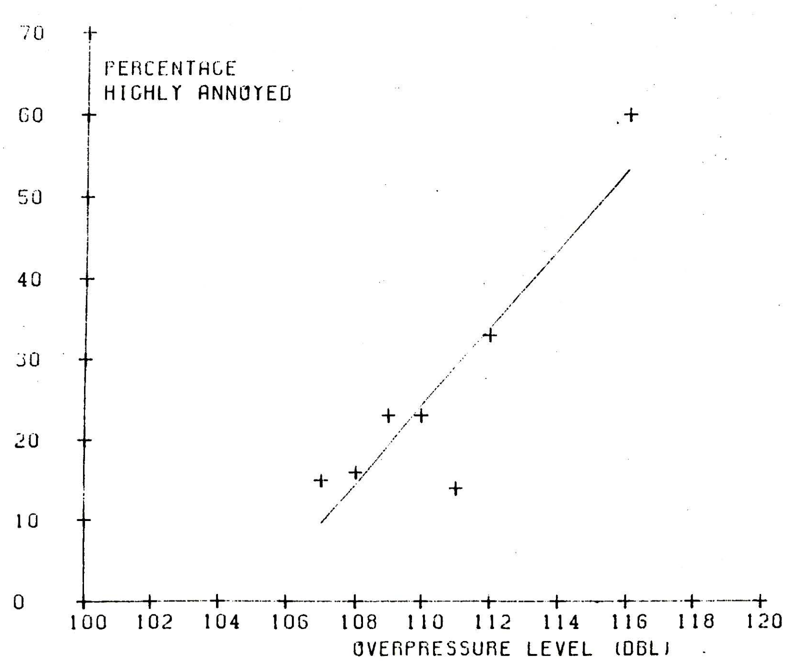

FIGURE 2 RELATIONSHIP BETWEEN PERCENTAGE OF RESPONDENTS HIGHLY ANNOYED AND OVERPRESSURE LEVEL.

5. CONCLUSIONS

The Study appears to confirm that blast overpressure causes some community reaction. Whilst the results of the survey indicated a significant percentage of residents highly annoyed, even down towards 105 dBL, care should be exercised in interpreting the results at the low percentage points of respondents highly annoyed. The results at least indicate that the 115 dBL limit widely used throughout Australia is a reasonable starting point which takes into account the practical difficulty of reducing overpressure levels from blasting, particularly in close sensitive areas.

Acoustics in the Eighties

ACOUSTICS IN HARD TIMES

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ACOUSTICS IN HARD TIMES

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ABSTRACT

In a time of economic bouyancy, such as the 1970's, various moves to improve the environment received widespread support. Legislation was passed to control pollution in its many forms, and in addition, health and safety in the work place became an important issue. Noise controls in the community and for hearing conservation purposes have been introduced in most States and Territories, and the necessary infrastructures and expertisc in government departments, and in relevant local authorities have been built up. Acoustic research, in academia., government and industry received encouragement and many acoustic consultancies were formed. For the most part the noise-makers, particularly the industrialists, have accepted the constraints placed on them,often at considerable economic cost. However, when there are great pressures to cut costs, and to make industry more competetive in world markets, there may well be pressure exerted for a relaxation of pollution control standards. Even if this does not occur overtly, there are likely to be delays in the purchase of new, quieter machines to replace old, noisy ones. When governments want to reduce staff numbers, those engaged in pollution control administration may seem to be the most expendable; scarce research funds are also more likely to be directed towards projects with a money-making, or efficiency improving potential It will be argued that such trends have long-term disbenefits and that acousticians should do all they can to prevent the hard-earned gains in environmental and workplace noise controls being eroded.

1. INTRODUCTION

It can be said with certainty that Acoustics in Australia, in the 80's (particularly in the areas of community noise, building acoustics and hearing conservation) is a very different matter from Acoustics in Australia in the 60's and 70's. In the 1960's a number of interested people around the country were conducting research and practicing in various branches of acoustics and several were also working toward the development of State-wide Acts and Regulations in the areas of community and occupational noise control and building acoustics, in the hope of achieving more formal recognition of some of the main acoustics problems. There had, of course, been several initiatives both in local government and in industry in these areas previously, but there was no general recognition of the problems nor general adoption of noise controls, although there was growing interest in the community in pollution in general, and, in the acoustics area in such problems as aircraft noise, noise from outdoor entertainment, etc. During the same period a number of consultants, academics and government research workers began to hold technical meetings covering various topics of interest in acoustics and as a result, the Australian Acoustical Society was formed.

During the 1970's much of this earlier groundwork came to fruition. Many of the States enacted noise control legislation in the areas of community noise and occupational noise, and there was some mention of noise control in building regulations. The Australian Acoustical Society, incorporated early in the decade, with Divisions in N.S.W. and Victoria only, expanded to Western Australia, and to South Australia. As a result of the legislative activity, specialised governmental agencies were formed, such as the noise control sections of the N.S.W.State Pollution Control Commission, the Victorian Environmental Protection Authority, and corresponding noise control sections in other State Environmental and Health Authorities. In addition, recognising the need for sound technical bases for measurement and assessment, the Standards Association of Australia formed an Acoustics Standards Committee, which set about drafting a number of standards, based partly on International Standards Organisation,(ISO) and International Electrotechnical Committee,(IEC) documents, but partly reflecting unique Australian conditions where necessary.

Research laboratories were busy, and active groups worked, for example, at CSIRO Division of Building Research in Melbourne, at Adelaide University and at the Experimental Building Station in Sydney. Grants were available for research in various areas of acoustics and formal graduate courses in acoustics were commenced, complementing the existing possibilty of obtaining higher degrees in acoustics by research.

However, all good things come to an end, and the expansionary 70's have given way to the contractions of the 80's. How will the achievements of the previous decades be maintained?

2. POLLUTE OR PERISH?

Consider the case of a manufacturer. On the one hand economic pressures encourage the greatest output for the smallest costs; on the other hand, a multitude of regulations, licences, etc. add to costs without perceived economic benefit. Placing and maintaining enclosures and other noise control equipment around a machine not only adds capital costs to the production process, but may even appear to slow down production because of restricted access to the machine. Psycho-acousticians will suggest that improved working conditions improve productivity, and it might be expected that widespread introduction of hearing conservation programmes would reduce compensation costs and premiums - however, it is extremely difficult to prove such benefits in dollar terms, particularly to a company accountant. In prosperous times, companies may be happy to adopt measures that will result in improved working conditions - albeit because they may find it difficult to retain capable employees if they do not. However, when Australian manufacturers are struggling to be competitive on international markets, they may well point to the more rugged working conditions in less-developed countries as a reason for higher costs of the local product. Of course, there have always been many companies showing a strong concern for employee health and safety, and they did not need the compulsion of regulations to reduce noise levels in the workplace. There are also still very many companies, usually with only a few employees, to whom the whole area of noise legislation is still a mystery, and who have not yet been brought under the regulatory net.

Noise control in industry is often a slow, difficult process. It only takes one noisy machine to increase the levels over a wide area, and until such a machine can be modified or replaced, any benefits from other, quieter machines, will not be realised. Replacement of properly functioning machinery is not usually the first priority when times are hard! On the other hand, a new machine may be more efficient, so the old machine replacement may be advanced in time. Unfortunately, as there is still a general lack of noise specifications and labelling, new machines may be no quieter than old ones.

Industry also may be responsible for emitting excessive noise levels to the surrounding community. In most parts of Australia, there is legislation in place which should prevent this occurring. However, it is highly unlikely that a marginally economical industry will be forced to install expensive noise control devices to protect one or two nearby residents, particularly if it can be argued that the industry will be forced to close resulting in more jobs being lost.

Not so many years ago, long arguments preceded the introduction of Australian Design Rules covering noise emission of new vehicles (which lag by some years, usually, the corresponding requirements in Europe). The gradual decibel-by- decibel reductions finally agreed to are having only a marginal effect on the very high traffic noise levels experienced by many people living near main roads. (Lawrence, 1985). Even the small reductions in noise levels that were expected to occur will be delayed in times of economic uncertainty in two ways - firstly, the rate of replacement of older, non-regulated vehicles by new ones is reduced, and secondly, there is likely to be even fiercer resistance by manufacturers to continued lowering of noise limits. Theoretically, in-service vehicle noise limits should be able to prevent individual noisy vehicles continuing to be driven unmodified, however, these regulations can only be effective if they are properly and extensively enforced.

Although the governmental agencies that have been formed to administer the noise control acts and regulations in the various States have gathered together well-qualified, professional staff, they would be the first to admit that they have nowhere near sufficient people to carry out all the work that they should be doing. This affects assessment, and the control and monitoring of new and existing projects. The lack of sufficient manpower is particularly noticeable in the various areas where inspection forms part of the process of enforcement.

3. MONEY-MAKING RESEARCH?

There is a disturbing trend emerging amongst grant-giving organisations (which are primarily governmental in this country) to limit grants to those people and research areas which can be readily seen as being of immediate economic benefit to the country. Again, it is difficult for acousticians to assert that a quiet Widgit is going to be one of the major solutions to the country's balance-of-payments deficit. Not only do we need to have in place our own noise-labelling requirements for appliances, but to succeed in overseas markets there also need to be many countries with similar requirements. Even in those areas where there are substantial world-wide noise emission requirements, for example, for motor vehicles, we have little chance of competing because, apart from the lack of home-based engineering design input in this area of manufacture, as mentioned earlier our own regulations are usually less stringent than those overseas anyway.

Some medical-acoustics research has been very successful commercially - the bionic ear being an outstanding example, and there are good prospects for underwater acoustics as well. However, the "consumer-oriented" research - theoretical and practical research into noise control in industry; community and transportation noise research; development of building materials and systems and services for noise control; psychoacoustics;

concert hall design and auditorium acoustics; the development of new methods of measurement and assessment, are all suffering from a chronic shortage of funding. This not only affects (and depresses) current personnel, but it is severely limiting the recruitment and training of others.

4. THE GREYING OF ACOUSTICIANS IN AUSTRALIA

Many of the Members of the Australian Acoustical Society have been members since its foundation, or shortly afterwards. Although the Society has continued to increase its membership over the years, it is a fairly slow process. What is of particular concern is the non-replacement of key members of the profession when they retire. This has left two of our major acoustic facilities virtually unoperational - I refer of course to the National Acoustical Laboratory at Chatswood and to the CSIRO Division of Building Research at Highett. The latter laboratory has a proud record stretching back to the earliest days of postwar acoustic research, attracting internationally recognised workers such as Werner Lippert, Arthur Nickson and others. As Fricke (1985) commented:

> CSIRO Division of Building Research is available for undertaking testing but, because of the lack of manpower and the effort requird to bring the facility back 'on-line' it is unlikely it will be used again by CSIRO.

The National Acoustical Laboratory was finally opened in 1986, some 6 years or so late. It is one of the largest acoustical facilities of its type in the world - and also without doubt - the emptiest, lacking not only manpower, but also the instrumentation able to take advantage of the exceptional facilities. This can only be judged as a gross and short-sighted waste of money. No matter the rights and wrongs of the intitial decision to build the laboratory in its present form, the Australian acoustical community should be able to take advantage of its capabilities for testing and research in many areas of importance, including those that could best be described as in the "public-interest", but which would have difficulty in attracting direct commercial sponsorship.

Another consequence of manpower and money shortages is the difficulty in attracting and retaining sufficient experienced acousticians to take part in acoustics and vibration standards drafting activities. The growing importance of acoustics and vibration has been recognised by the Standards Association of Australia by the formation of the Acoustics and Vibration Standards Board, which is responsible for a growing number of standards covering instrumentation, and measurement and assessment techniques. Some of standards are in the forefront of world standardisation, for example the Aircraft Noise Intrusion - Building Siting and Construction standard, which was developed with the cooperation of the Department of Aviation and which may be used by local government, architects and planners to assess and ameliorate aircraft noise problems.

However, not only are there fewer people than desirable for standards drafting, the necessary research to support their technical bases, for example, the theoretical analysis and measurement of the noise reduction provided by Australian building systems and elements, is proceding very slowly.

5. CONCLUSION - WHAT HOPE FOR THE FUTURE

Undoubtedly, the efforts of a considerable number of acousticians and others over the last two decades or so have placed Australia in a good position with respect to noise control regulations, standards and professional expertise. One-third of the delegates present at the 10th International Congress on Acoustics, held in Sydney in 1980 were Australian acousticians, and there is always a respectable number of Australians at ICA, Internoise and other such conferences. For how long Australia will be able to maintain its place in the international acoustics world is difficult to predict in the current economic climate. Increasingly our community seems to be run by accountants interested only in the current year's "bottom line".

We should perhaps remind ourselves of the bad old days of the 19th Century, when industrial progress proceeded without the benefit of environmental, workplace or building health and safety considerations. If we allow ourselves to become complacent, in fact, if we do not form ourselves into a strong "lobby" group then we may well find that our hard-won gains in making the world a quieter and more pleasant place in which to live or work will be eroded. Perhaps the only way that we can succeed is to make a big **NOISE!**

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Acoustics in the Eighties

Session 1B: Propagation and Attenuation I

Acoustics in the Eighties

THE ABSORPTION OF SOUND BY AIR

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THE ABSORPTION OF SOUND BY AIR

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ABSTRACT

The mechanisms of the absorption of sound by air are described. These mechanisms are the quantum mechanical effects of vibrational and rotational relaxation and the classical mechanical effects of viscosity and heat conduction. The history of the development of the absorption formulae and the work of Kneser, Evans and Bazley, Harris, Monk, Delany, SAE, ANSI and ISO are reviewed. The different air absorption formulae are used to remove the air absorption from the total absorption, of a bare 600 m³ reverberation room, which has been determined by reverberation time measurements at 35 different values of temperature and humidity. This enables the calculation of the wall absorption coefficient of the reverberation room at each value of temperature and humidity. The air absorption formulae are judged by their ability to produce the smallest standard deviation of the wall absorption coefficient across the 35 different values of temperature and humidity. This is done on the assumption that almost all the variation in the total absorption is due to the air absorption. The 1986 draft ISO air absorption formula gives the lowest standard deviation compared to the other formulae at all but three of the measurement frequencies. At these three frequencies it just fails to give the lowest standard deviation.

1. INTRODUCTION

The aim of this paper is to compare the performance of a number of different formulae for predicting the absorption of sound by air. The performance of the formulae is judged by their ability to reduce the variation of the total measured sound absorption in a 600 m^3 reverberation room across a range of temperatures and humidities. First the theory and the different formulae are reviewed. Then the results of the experimental work are described.

2. REVIEW OF THEORY

Diatomic molecules such as oxygen and nitrogen have seven degrees of freedom, three translational, two rotational and two vibrational. The two vibrational degrees of freedom arise from the kinetic energy and the potential energy of the single vibration mode. There are only two rotational degrees because the atoms act as point masses and there is no moment of inertia about the axis joining the two atoms.

The molar specific heat at constant volume for each fully excited degree of freedom is R/2, where R is the universal gas constant. However, quantum mechanics shows that the energy levels are quantised. The translational energy levels can be considered to be continuous because the levels are so close together, but the rotational and vibrational energy levels are significantly quantised. The probability of these energy levels being excited depends on the number of molecules which collide with them with energy greater than the transition energy between levels.

At low temperatures, the vibrational and rotational energy levels are not significantly excited and so the molar specific heat at constant volume of a diatomic gas is about 3R/2 corresponding to the three translational degrees of freedom. As the temperature increases, the rotational degrees of freedom become excited and both oxygen and nitrogen have molar specific heats at constant volume of about 5R/2 at normal room temperatures. At even higher temperatures the vibrational degrees of freedom become excited and the molar specific heat at constant volume of diatomic molecules approaches 7R/2.

Since the energies of molecules in a gas are distributed over a wide range, the vibrational degrees of freedom make a small contribution to the molar specific heat at constant volume even at room temperature. The contribution for a particular type of molecule in the gas can be calculated using the Planck-Einstein relation (Bass *et al.* 1984). It depends only on the characteristic vibrational temperature of the particular molecules the temperature of the gas, the mole fraction of the particular molecules, and the universal gas constant. The characteristic vibrational temperature can be determined very accurately using spectroscopic techniques.

Because a sound wave is adiabatic, it is a temperature wave as well as a pressure wave. It is the effect upon the temperature wave of the very small part of the specific heat that is due to the vibrational degrees of freedom that is responsible for most of the absorption of sound by air that occurs in the audio frequency range. The heat takes a finite time to enter and leave the vibrational degrees of freedom. This means that it leaves out of phase with the temperature wave and thus attenuates it.

The relaxation frequency is the frequency at which the maximum absorption per wavelength occurs. The absorption per wavelength takes the form of a bell-shaped curve when plotted against frequency. The absorption per wavelength is proportional to the frequency at frequencies below the relaxation frequency, peaks at the relaxation frequency, and is proportional to the reciprocal of the frequency at frequencies above the relaxation frequency.

When we convert to absorption per metre by multiplying by the number of wavelengths per metre, we find that the absorption per metre is proportional to the square of the frequency below the relaxation frequency. At the relaxation frequency it is equal to half its maximum value, and above the relaxation frequency it asymptotes to twice its value at the relaxation frequency.

The shape of the absorption per wavelength curve can be interpreted as follows. For frequencies below the relaxation frequency, the lag of heat going in and coming out of the vibrational degrees of freedom is not long enough to produce much absorption. Above the relaxation frequency the lag is too long to allow very much heat to go into the vibrational degrees of freedom.

The maximum absorption per wavelength can be predicted from that part of the molar specific heat due to the vibrational degrees of freedom. This can be calculated from spectroscopic measurements. Unfortunately, the relaxation frequencies cannot be predicted accurately enough theoretically. This is because a large number of different energy exchanges between molecules are involved and the reaction rate constants are not known accurately enough. Both the oxygen and nitrogen vibrational relaxation frequencies depend on the absolute humidity. This is because the water molecules occur in the excitation and de-excitation reactions of the vibrational degrees of freedom of oxygen and nitrogen.

Although the rotational degrees of freedom are fully excited at normal room temperatures, there is a very short lag associated with the flow of energy to and from these energy stores. The relaxation frequencies associated with rotation are greater than 10 MHz. Thus, at frequencies less than 10 MHz they produce an absorption of sound which is proportional to the square of the frequency.

Sound is also absorbed by the classical mechanisms of viscosity, heat conduction and diffusion. These classical mechanisms produce an absorption of sound which is also proportional to the square of the frequency, and can thus be combined with the rotational relaxation absorption into the so-called 'classical' absorption term.

At low frequencies, the absorption of sound by air is dominated by the nitrogen relaxation which increases as the square of the frequency before asymptoting to a fixed value. As the frequency increases further, the oxygen relaxation becomes the dominant absorption mechanism. Again it increases as the square of the frequency before asymptoting to a constant value. Increasing further in frequency the 'classical' absorption term takes over and increases as the square of the frequency. Thus, the absorption of sound increases as the square of the frequency with several steps as we change from one dominant absorption mechanism to another. The position of these steps and the magnitude of the absorption depends strongly on the absolute humidity.

3. REVIEW OF FORMULAE

Because the attenuation of sound by air at audio frequencies is small, its measurement had to wait until the development of electronics in the first part of this century. It was soon realised that the classical mechanisms underpredicted the absorption of sound in the audio frequency range. In the 1930s this 'anomalous' absorption was explained by Kneser (1933) and others as being due to the vibrational relaxation of the oxygen molecules. Unfortunately, on the basis of limited experimental data, Kneser concluded that the oxygen relaxation frequency varied as the square of the absolute humidity.

Evans and Bazley (1956) showed that the oxygen relaxation frequency varied as the absolute humidity to the power 1.3. Their equation is in reasonable agreement with currently accepted values over most of the humidity range. However, their experimental results showed that the oxygen relaxation underestimated the absorption at 1 and 2 kHz for high humidities.

In the 1960s Harris (1966) attempted to overcome this problem by obtaining an experimental curve of the ratio of absorption to maximum absorption for that frequency versus the ratio of humidity to the humidity at which the maximum absorption for that frequency occurs. This was quite successful in extending the prediction of sound absorption to lower frequencies. Unfortunately Harris also produced a formula for the relaxation frequency of oxygen which differed significantly from Evans and Bazley and currently accepted values. Thus, Harris extended the prediction ability to lower frequencies, but introduced inaccuracies by moving away from the Evans and Bazley formula for the oxygen relaxation frequency.

In 1964 the Society of Automotive Engineers (SAE) followed Harris's approach, but produced a different experimental curve, and they compounded Harris's mistake with the oxygen relaxation formula by reverting to Kneser's original formula, where the oxygen relaxation frequency depends on the square of the absolute humidity. Unfortunately this SAE document was reissued in 1975 (Society of Automotive Engineers 1975) and has been adopted in the current ISO Standard on aircraft noise (International Organization for Standardization 1978).

Delany and Bazley (1970) also followed Harris's approach and produced yet another experimental curve using the original Evans and Bazley data. Fortunately they retained Evans and Bazley's formula for the relaxation frequency of oxygen.

Piercy (1969) removed the need for Harris's *ad hoc* approach by showing that the extra unexplained absorption was due to the vibrational relaxation of nitrogen. It is interesting that when I visited Bazley in 1978 he told me that they had asked physical chemists whether the nitrogen could be responsible for the extra absorption and were told flatly no!

Monk (1969) produced another formula for the oxygen relaxation frequency. This formula was modified and used in conjunction with the nitrogen relaxation by Sutherland *et al.* (1974). This work was slightly modified to become the ANSI standard (American National Standards Institute 1978).

Bazley (1976) added the nitrogen relaxation to the Evans and Bazley formula. Unfortunately he made a mistake with the magnitude of the maximum nitrogen absorption. Bazley also modified the 'classical' absorption term to agree with more recent work.

The oxygen relaxation formulae of Evans and Bazley, Monk, Sutherland *et al.* (1974), and ANSI are in reasonable agreement with each other over most of the absolute humidity range. The main disagreement between ANSI and Bazley (with the maximum nitrogen absorption corrected) is in the nitrogen relaxation formula. If the percentage absolute humidity is h, ANSI uses 350h while Bazley uses 190h for the nitrogen relaxation frequency in hertz. Bass *et al.* (1984) commented that recent work in moist nitrogen had suggested 200h.

In 1986 a new draft ISO proposal (International Organization for Standardization 1986) adopted 280h for the nitrogen relaxation frequency. This new draft also modifies the constants in the oxygen relaxation formula compared to the ANSI standard on which it is based.

4. PERFORMANCE COMPARISON

To test different air absorption formulae, reverberation time measurements were made in the 600 m^3 reverberation room at the CSIRO Division of Building Research. These measurements were made in 1970 before the room was equipped with diffusing panels which substantially lowered its reverberation times. The measurements were made in third-octave bands using random noise. The third-octave band centre frequencies used were 1.6 kHz and the seven between 3.15 and 12.5 kHz inclusive. A total of nine decays at each of these eight frequencies were measured for each combination of temperature and relative humidity. These nine decays consisted of three decays at each of three microphone positions. Measurements were made for the 35 different values of temperature and relative humidity that are shown in Table I.

The total sound absorption was calculated using the Sabine equation from the average of the nine measured reverberation times. The different air absorption formulae were used to remove the air absorption from the total absorption and thus give the residual wall absorption. The wall absorption coefficient of the room was calculated by dividing the residual wall absorption by the surface area of the room. For each air absorption formula the standard deviation of the wall absorption coefficient across the 35 different values of temperature and humidity was calculated. This standard deviation was also calculated for the case of no correction for air absorption, i.e., the air absorption was set to zero. This case was taken as the reference case and the standard deviations for the other air absorption formulae were normalised by dividing by the standard deviation for this reference case. These normalised standard deviations are shown in Table II. The best air absorption formula is judged to be the one with the lowest normalised standard deviation. This is because this equation best predicts the variation of air absorption over the range of temperature and humidity across which the measurements were performed.

TABLE I

CLIMATIC CONDITIONS

Temperature (°C)	17.6	18.1	17.8	17.8	18.6	17.8	18.3
Relative humidity (%)	61.3	64.0	69.6	73.2	77.4	79.3	84.6
Pressure (hPa)	1016	1016	1016	1016	1015	1016	1014
Temperature (°C)	20.0	21.1	21.1	20.7	21.1	21.1	21.3
Relative humidity (%)	62.6	63.6	66.8	71.9	78.8	85.6	92.7
Pressure (hPa)	1012	1013	1010	1011	1009	1008	1008
Temperature (°C)	23.9	24.3	24.6	23.9	24.0	23.9	24.0
Relative humidity (%)	61.0	71.1	75.3	79.0	85.0	90.4	94.3
Pressure (hPa)	1015	1016	1016	1016	1016	1015	1016
Temperature (°C)	27.2	26.5	26.1	26.1	26.5	26.7	26.8
Relative humidity (%)	47.9	56.3	61.9	78.0	80.2	83.2	83.3
Pressure (hPa)	997	1000	1017	1015	1014	1014	1015
Temperature (°C)	30.0	29.8	29.9	29.7	29.7	29.9	29.7
Relative humidity (%)	56.8	60.0	62.3	67.2	70.7	74.3	80.5
Pressure (hPa)	1001	1000	1000	1000	998	998	998

TABLE II

NORMALISED STANDARD DEVIATIONS

Third-octave band centre frequency (Hz)	1600	3150	4000	5000	6300	8000	10000	12500
Evans and Bazley	1.39	1.84	1.13	0.83	0.55	0.61	0.33	0.46
Monk	1.32	1.58	0.93	0.66	0.40	0.46	0.28	0.47
Harris	1.08	0.70	0.60	0.64	0.63	0.56	0.55	0.61
SAE	0.44	1.02	1.09	0.93	0.76	0.61	0.56	0.62
Delany and Bazley	0.68	0.57	0.48	0.44	0.30	0.45	0.26	0.45
Sutherland *et al.*	0.41	0.90	0.71	0.50	0.34	0.32	0.30	0.50
ANSI	0.44	0.96	0.73	0.50	0.33	0.32	0.29	0.49
Bazley	0.62	0.91	0.65	0.54	0.37	0.49	0.28	0.46
Bazley (corrected)	0.29	0.56	0.47	0.44	0.30	0.45	0.26	0.45
ISO	0.33	0.48	0.42	0.35	0.22	0.35	0.24	0.46
ISO (modified)	0.27	0.44	0.40	0.38	0.25	0.39	0.24	0.46

The first two rows are Evans and Bazley's air absorption formula and Evans and Bazley's air absorption formula with Monk's expression for the oxygen relaxation frequency. These are the only two formulae in the table with no allowance for the nitrogen relaxation and this is shown by values greater than 1 at the two lowest frequencies. This means that at these frequencies the use of these formulae has actually increased the standard deviation rather than reduced it. At the two highest frequencies the standard deviations are as small as the other formulae, which indicates that the oxygen relaxation frequency expressions are both fairly good. However, Monk has lower values at all frequencies, except the highest, indicating his oxygen relaxation expression is slightly better.

The next three rows all use Harris's method of coping with what we now know to be the nitrogen relaxation. Both Harris and SAE have normalised standard deviations for the highest two frequencies that are larger than those for any of the other formulae. This is because of their incorrect oxygen relaxation expressions. They also produce values greater than 1 at some of the lower frequencies. This is also presumably due to their incorrect oxygen relaxation expressions, since Delany and Bazley produces much lower values with the same type of model. Also SAE produces greater values than Harris except at the lowest frequency, which is to be expected since they reverted to the earliest and most incorrect expression for the oxygen relaxation frequency. Delany and Bazley have the best of the formulae that do not explicitly include the nitrogen relaxation.

The next six rows all include the nitrogen relaxation. The first two of them are Sutherland *et al.* and its modification which became the ANSI standard. It is interesting to note that except for two frequencies, these two formulae do not perform as well as Delany and Bazley's.

The next row is Bazley's version which is Evans and Bazley with the nitrogen relaxation included. Surprisingly it does not perform as well as Delany and Bazley except at 1.6 kHz. This puzzle was investigated further and it was discovered that Bazley had used the wrong value for the maximum nitrogen absorption. Bazley's formulae with the correct nitrogen maximum absorption is shown in the next row. Except for one frequency this corrected Bazley formula performs better or as well as any of the formulae so far discussed.

The next row shows the draft ISO formula. This is the most successful air absorption formula yet produced. It produces the smallest standard deviation at five of the frequencies and is close to the smallest standard deviation at the other three frequencies. The last row shows a modification of the ISO formula. The nitrogen relaxation frequency was changed from 280h to 200h. This reduced the standard deviation at the lowest three frequencies, but increased it for the middle three frequencies with no change at the top two frequencies. Thus, overall this modification is no better than the draft ISO formulae.

5. CONCLUSION

It is to be hoped that the draft ISO proposal will quickly supersede the other air absorption formulae. However, it should be noted that even with this ISO formula the residual standard deviation is only reduced to between one-half and one-quarter of the value with no correction for air absorption. Thus, it is still preferable to keep temperature and humidity constant when measuring acoustical absorption coefficients in a reverberation room rather than making air absorption corrections for changed climatic conditions.

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SOUND ATTENUATION RATES NEAR THE GROUND

F. Fricke and R. Treagus

SOUND ATTENUATION RATES NEAR THE GROUND

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ABSTRACT

One of the greatest sources in variation in outdoor sound emission levels is the atmosphere. For nominally identical atmospheric conditions attenuation rates may vary dramatically, especially near sunrise and sunset. This variability should be considered indicative of the errors inherent in existing prediction techniques. In this paper results of field measurements are presented which indicate the attenuation rates and their variability are a function of time of day, relative humidity and cloud cover. The paper also compares measured and predicted sound levels in a limited number of cases. These comparisions suggest that European-based prediction methods may not have sufficient atmospheric stability categories for Australian conditions.

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1 INTRODUCTION

There are now several theoretical models available for estimating sound propagation outdoors eg. Manning (1981),Tonin (1985),VDI (1976),Kragh (1982). These models are all similar in concept but differ in the number of factors they consider and their degree of sophistication. Attempts have been made to estimate the errors inherent in such models e.g Manning (1981), Jakobsen (1983), Mizia (1986), but such evaluations are extremely difficult to undertake because of the large number of variables which have to be considered and monitored. As a consequence it would seem unlikely that these theoretical models will ever be systematically evaluated.

If these "European" models are to be used in Australia there needs to be some attempt to assess them in tropical and sub-tropical conditions. This assessment, in its simplest form, need only determine the accuracy of predictions under local atmospheric conditions. This was attempted in a number of simple cases (single source, flat or undulating grassed surface and little or no wind). Even with these simplifications measurements were difficult to obtain.

An alternative approach is to look at the significance of factors not considered in the theoretical models. This was attempted in the scattering of sound. It seems as though scattering is important, at least for a single directional source. The interaction between atmospheric and ground conditions was also investigated.

In the past simpler theoretical models have been used. Th simplest of these models considered hemispherical spreading only. More recently Kenna et al (1986) have found the best fit to their field data was given by:

Attn dB(A)=6dB(A)/dd + 3dB(A)/km

This type of simplification may have restricted application (eg. large distances, undulating ground) but is far easier to apply than the more sophisticated computer models which require ground contours and detailed atmospheric conditions. If it could be shown that a simplified model gave results which were sufficiently accurate, or of a similar degree of accuracy to the more complex models, considerable computational effort could be saved.

2 COMPARISON OF MEASURED SOUND LEVELS WITH CONCAWE PREDICTIONS

A comparison of measured sound levels with those predicted using the CONCAWE model Manning (1986) was made to give an indication of the accuracy of the predictions under local meteorological conditions.

Measurements were made over flat or undulating grassland over a distance of 1600m. The receiver height was 1.2m. The source was a gas gun impulse source. (The height of the source was approximately 0.6m). Because of the nature of the source and the difficulty of defining its sound power, exact agreement with CONCAWE predictions would not be expected but a consistent error might be.

The simplest indication of the inherent errors in the CONCAWE model is obtained if the difference in sound levels, measured at two times during the day, are recorded. Table 1 shows the difference in sound levels measured at 7.45 am and 8.55 am over the same ground and for nominally the same atmospheric conditions

Table 1.
DIFFERENCE IN MEASURED OCTAVE BAND SOUND LEVELS FOR TWO NOMINALLY INDENTICAL SITUATIONS.

Octave Band Centre Frequency (Hz)	63	125	250	500	1k	2k
Difference in Sound Level measured at 7.45 am and 8.55 am (dB)	14	14	13	13	11	7

As the CONCAWE model indicates no change of sound it appears that errors of at least 10dB(A) could be expected in the predictions over distance of 1600m.

Further comparisons were made under different meteorological conditions. These comparisons are shown in Table 2. Here the assumption has to be made that the CONCAWE model holds for impulse sources and that sound levels measured at 25m from the source can be used to estimate the sound power of the source.

TABLE 2
DIFFERENCE BETWEEN MEASURED AND CONCAWE PREDICTED SOUND LEVELS

	Oct Band Centre Frequency (H_z)					
SITUATION	63	125	250	500	1k	2k
1) Meteorological Category 5 $L_{meas}-L_{concawe}$(dB)	8	9	4	3	11	21
2) Met.Cat. 3 $L_{meas}-L_{concawe}$(dB)	-3	-2	-2	-4	-2	+4
3) Met.Cat.6 (over water) $L_{meas}-L_{concawe}$(dB)	3	2	-4	6	1	7
4) Met.Cat.4 (over water) $L_{meas}-L_{concawe}$(dB)	-15	-10	-23	-22	-21	-14

The tentative conclusions from these very limited measurements are:

(i) That the six CONCAWE meteorological categories are insufficient.

(ii) The CONCAWE predictions apply best for downward propagation and upwind propagation with a weak inversion.

(iii) The CONCAWE predictions are least successful when the temperature inversion is strongest.

3 SCATTERING OF SOUND

Scattering of sound can be cause by the atmosphere, the ground, vegetation and terrain. Scattering will have a more pronounced effect on directional sources than on omnidirectional ones. Some prediction methods, such as CONCAWE, do not consider the directivity of the source. This could lead to prediction errors. Other prediction methods do consider the directivity of the source but do not consider scattering by the ground and atmosphere. As scattering of sound would increasingly reduce the directionality of the sound with distance this could also lead to errors. That scattering of sound by the atmosphere does occur is not in doubt as atmospheric sounders operate on this principle. What is in doubt is the magnitude of the scattering.

Several papers on the effects of atmospheric scattering have been published. Ingard and Maling (1963) suggest that although the effect of turbulence on sound may be small in free space in the presence of a boundary it can cause marked fluctuations in the received level of a pure tone. Rudd (1973) showed that attenuation by turbulent scattering could be of the order of 4-6 dB/100m if the atmospheric boundary layer is considered thin. Rudd indicated that if the turbulent boundary layer is of finite thickness the wave would need to be scattered several times before leaving, thus reducing the attenuation with distance.

While Ingard and Rudd suggest that scattering is important Brown and Clifford (1976) who based their work on the spread of a beam, show that scattering by turbulence is negligible; of the order of 1dB/km. Scholes (1968) too suggested that attenuation due to turbulent scattering is negligible, compared with other factors, in the near horizontal propagation of sound close to the ground.

In order to obtain an independent assessment of the importance of scattering and to determine the magnitude of errors, when the source directivity and scattering are neglected, measurements were made over grass using a directional source. Sound level measurements were made at 50m 450m and 800m from a gas gun with the gun facing towards the source and away from the source. The results are of limited use for a number of reasons. Some of the most important of these are that only two distances from the source were considered, that only one source directivity and one set of atmospheric conditions were used and that scattering due to the atmosphere could not be separated form other scattering effects.

The results of this study are shown in Table 3. Table 3 shows that scattering of sound is an important factor in outdoor sound propagation when the source is directional. This is broad agreement with Ingard (1963), and Rudd (1973).

In the present case the scattering could be due to the atmosphere, the ground or both. Further measurements are being planned in order to separate ground and atmospheric effects and to determine a correction factor for scattering. In the meantime situations with omnidirectional sources or multiple sources can probably be handled without considering scattering. Over large distances it will be the low frequencies which propagate and as, at these frequencies, the source will tend to be omnidirectional scattering can also be neglected.

TABLE 3
SOUND SCATTERING AT TWO SITES

SITE 1						
1/3 oct centre frequency (Hz)	63	125	250	500	1k	2k
Level diff. ΔL_1= $L_{0°}-L_{180°}$ (dB) at 50m	13	15	20	16	7	5
Level diff. ΔL_2= $L_{0°}-L_{180°}$ (dB) at 800m	5	5	12	12	10	5
$\Delta L_1-\Delta L_2$ (dB)	8	10	8	4	-3	0
SITE 2						
Level diff. ΔL_1= $L_{0°}-L_{180°}$, (dB) at 450m	16	19	21	19	8	11
Level diff. Δ_{L2}= $L_{0°}-L_{180°}$, (dB) at 800m	13	13	16	14	13	13
$\Delta_{L1}-\Delta_{L2}$ (dB)	3	6	5	5	-5	-2

Note that at site 2 the sound level measurements at 0 and 180 degrees were made approximately one hour apart which could account for the negative differences at 1 and 2kHz. What is also worth noting is that at both sites the maximum scattering effect occurs around the frequency at which $L_{0°}-L_{180°}$ is greatest, as expected. $L_{0°}-L_{180°}$ is not an ideal measure of directionality but considering the other limitations of the present measurements it is thought to be sufficient.)

It would seem from these results that scattering in a relatively still atmosphere and over flat open grassed surfaces could be of the order of 1dB per 100m, even at low frequencies, when the front/back directionality difference is of the order of 20dB. This figure, at best, must be considered a ball-park one as the results are limited and as the attenuation rate per 100m could be expected to decrease with distance as previous scattering reduces the directionality of the source.

4 ATTENUATION RATE VARIATION DURING THE DAY

In an earlier paper, Fricke (1986) results of attenuation rates in forests were presented which showed a dependency on relative humidity. This relationship is shown in Fig. 1. As relative humidity near the ground is largely a function of the time of day (being highest before dawn and lowest in the afternoon) attenuation rates should be a function of the time of day also. This relationship is of course only likely to apply when there is little or no wind and when there is no rain.

Fig.2 shows the sound attenuation over grassland (up to 400m from the source) as a function of the time of day. There appears to be no correlation but when the data is massaged to eliminate points obtained when the windspeed at 1.2 m above the ground was more than 1m/s in the direction of or against the direction of propagation and to eliminate values obtained around sunrise and sunset there is a well defined trend (Fig. 3.). The relationship between attenuation rate and time of day shows the attenuation rate is greatest in the morning and lowest in the afternoon. Attenuation rates presented are the average of five 60 second Leq_S. The source in this case was a horn driver 2m above the ground. The signal used was a 'chirp'.

The reason for this result is not at all clear. Superficially, the opposite trend would be expected; increasing attenuation with time of day. This is because the ground is heating up and the temperature gradient above the ground could be expected to become more extensive and more pronounced. There are two explanations of the reduction of attenuation with time of day which have some merit but which have yet to be investigated:

(i) That the temperature gradients close to ground are the dominant influence on the sound attention and that these gradients are greatest in the early morning soon after sunrise.

(ii) That the ground effect dip in the spectrum is a function of the temperature gradient near the ground.

The first explanation has some supporting evidence from Jacobs (1986). Jacobs stated that the atmospheric conditions near the ground around sunrise and sunset are very unstable. These unstable conditions would result from intense temperature gradients. The dramatic changes in sound attenuation rates around sunrise and sunset, as shown in Fig.2 is also an indication of this instability. Temperature inversions have been found to increase sound levels even though the increases are highly variable, e.g. 2.2dB, Embleton (1982), and 30dB, Sills (1982).

There is also some supporting evidence for the second explanation. Figs. 4 & 5 show that the spectra at 50m and 400m change significantly with time. At 50m the dip is always apparent and the frequency of the dip shifts slightly. At 400m the dip is much less apparent but the spectral changes are large.

5 DISCUSSION AND CONCLUSIONS

From the limited data presented it would appear that outdoor sound level predictions based on the CONCAWE approach should be treated with caution. There are a number of possible reasons for doubting the efficacy of these computations.

(i) Limited meteorological categories.

(ii) Ignoring of scattering by the atmosphere and terrain.

(iii) Ignoring interaction effects such as that between the ground and meteorological effects.

(iv) Higher insolation rates in Australia than in Europe.

The first two of these reasons have been covered to some extent in this paper already. In some model studies it has been shown that by heating a surface can be increased or decreased. It appears as though the frequency at which the ground interference dip occurs is substantially affected by the temperature gradient close to the ground.

In the model studies the 1/3 octave sound levels measured 30mm above a flat carpeted surface and 1 m from the source increased and decreased by up to 4dB when the surface was heated. A narrow band analysis gave increases and decreases of more than 10dB. Although refraction effects on their own may be small compared with ground effects, as Parkin (1965) maintains, the interaction of ground and refraction effects seems to be significant and should be included in prediction methods. (This interaction between wind and ground effects has already been shown theoretically, by Madry (1986) & Rasmussen (1986).

The possibility that higher insolation rates in Australia may make European prediction methods inappropriate has not been investigated. It deserves further attention as does the shading effect of vegetation on the temperature gradient near the ground and hence the temperature/ground interaction. What also needs greater attention is the use of simplified models. There is a strong possibility that very simple models such as that proposed by Kenna et al (1986) are just as accurate as complex computer models of the CONCAWE type, especially if the simple models are applied to a limited range of situations. For instance the NAL model may be successful over distances of about 3km over undulating grassland. A model based on time of day may be better over distances of 0.5 km. What must be recognised is that errors of ± 15 to 20dB(A) can occur in predictions, at least for short durations. Expectations of errors of ± 1 or 2dB(A) are quite unrealistic.

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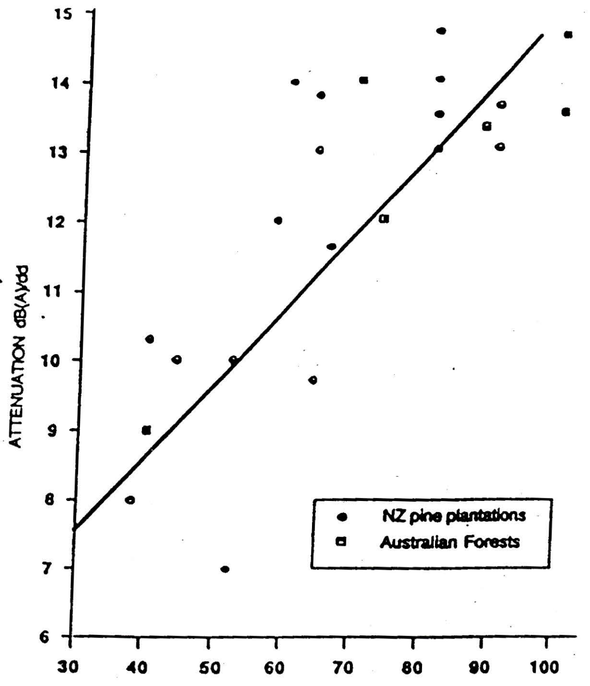

FIG. 1. Correlation of overall attenuation rate (dB(A)/dd) with RH (%)

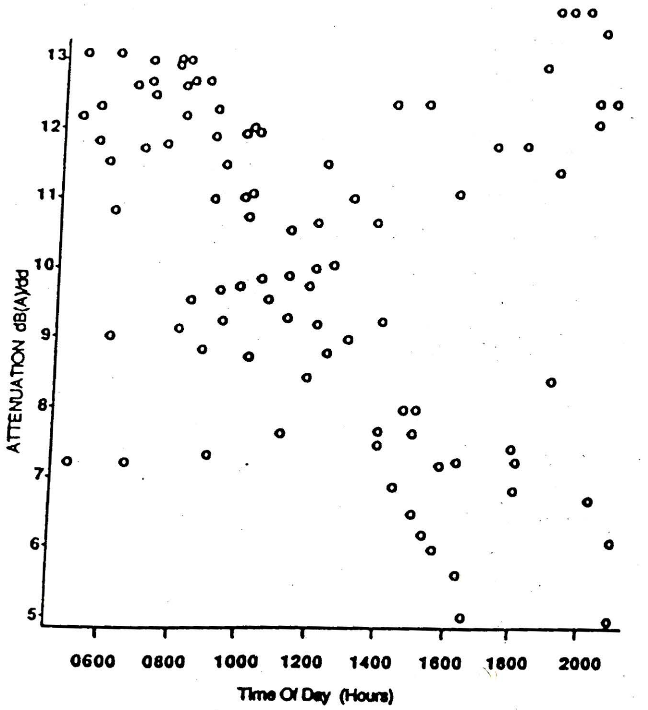

FIG. 2. Correlation of Grassland Attenuation (dB(A)/dd) with Time of Day: Raw Data

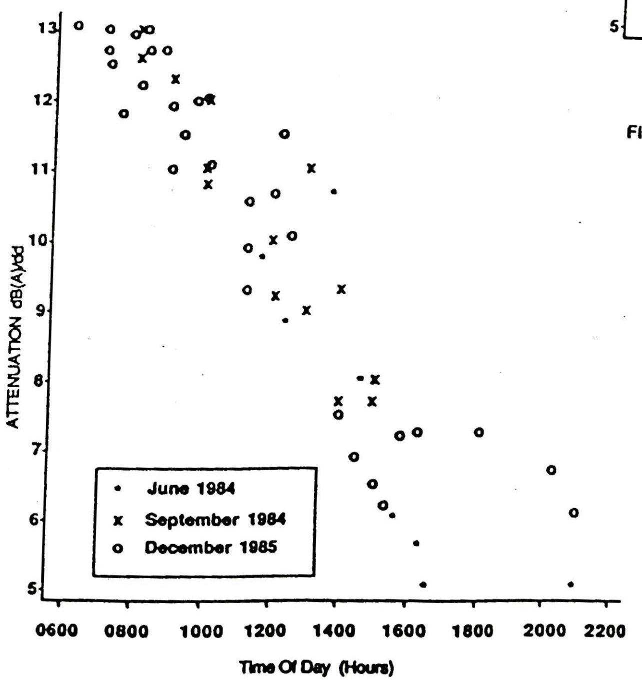

FIG. 3. Correlation of Grassland Attenuation (dB(A)/dd) with Time of Day: "Massaged" Data

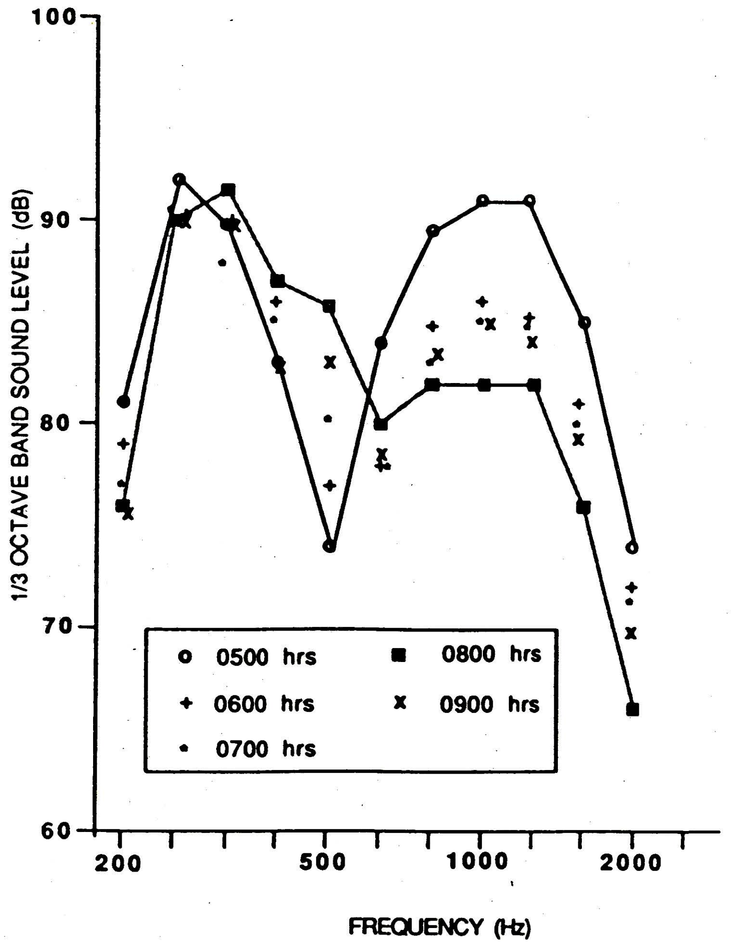

FIG. 4. 1/3 Octave Band Sound Levels Measured over Grass 50m from the Source.

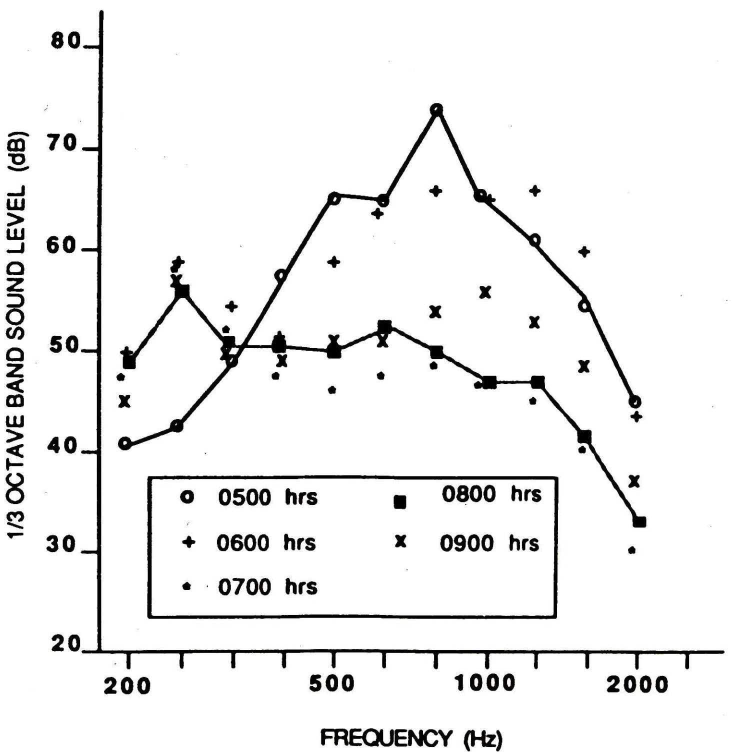

FIG. 5. 1/3 Octave Band Sound Levels Measured over Grass 400m from the Source.

Acoustics in the Eighties

SOUND PROPAGATION THROUGH THE ATMOSPHERE — A DETAILED FIELD STUDY

P. Peploe

INTRODUCTION

This study arose from the need to predict more accurately the effect of high-powered noise sources on the surrounding community. In particular, the Department of Defence was concerned about the effect of operating weapons and machinery on existing military ranges and bases.

Over the years, as the population of cities has increased, communities have become established close to military firing ranges and air bases. This closeness has forced the Defence Department to look at ways of utilising their equipment which cause least disturbance to the community and yet at the same time enable them to train effectively. To do this requires a detailed knowledge of the sound power and directivity of the equipment (with artillery weapons this includes the sonic bang of the shell and the shell impact) and the factors which influence the propagation of sound through the atmosphere.

In 1983 N.A.L. were requested by the Department of Defence to look at a method of predicting, with reasonable accuracy and consistency, noise levels at distances up to several kilometres from high-powered military noise sources.

OUTDOOR SOUND PROPAGATION

In order to assess the extent of noise exposure it is usual to draw contours of equal noise level on a map of the area of concern. These contours may be peak levels for impulsive noise, A-weighted rms levels for continuous noises or ANEF for aircraft noise.

The main difficulty in determining these noise levels is in knowing what factors affect the propagation of sound through the atmosphere and in determining the influence of each factor on the resultant sound level.

Some of the factors which influence the propagation of sound over long distances are: the height of both the source and the receiver above the ground; the position of barriers and reflecting surfaces in relation to the source and receiver; the terrain between the source and receiver; the ground surface and vegetation between the source and receiver; the meteorological conditions.

Some of these factors remain constant for a given location and fixed source and receiver heights. However, the ground surface and vegetation are affected daily by the weather and seasonally by the climate. Also, the meteorological conditions change continually.

It is because of these variations that these factors are difficult to quantify.

EXPERIMENTAL METHOD

Because these temporal variations play such an important part in determining the way sound propagates it would appear that a statistical method is the only way to predict satisfactorily sound levels at distances greater than a few hundred metres from a noise source. The method adopted to do this was, using automatic loggers, to gather a large amount of data on noise levels with distance, over various terrains at different times of the day and the year and for different climates. With sufficient data, one can then make a statistical estimate of the sound attenuation rate with distance for a particular terrain and vegetation type and for a particular climate. This information may be presented in the form of the percentage of time an attenuation rate is exceeded for a particular range of distances.

Obviously the data base would need to be large to cover typical terrains used by the Department of Defence and a range of times of day, seasons and climates.

EQUIPMENT USED

A. Sound Source

A sound source was required which would provide levels up to 60 dBA at the furthest distance of interest so that the source was not masked by the ambient noise. The furthest distance selected for the study was 4 kilometres, which implies that the sound source must produce at least 140 dBA at 1 metre.

After examining various types of source the most suitable sound source appeared to be a horn loudspeaker. The horn loudspeaker selected was a unit made by Community Light and Sound of Pennsylvania, U.S.A. Two such units were purchased. The stated power output of the speaker, for a maximum of 200 watts input, gives a calculated level of more than 140 dB at 1 metre, on axis, from the mouth of the horn. The stated frequency bandwidth is 200 Hz to 2000 Hz which is narrower than preferred. However, since the purpose of the study is to predict noise levels which will be experienced by residents (i.e. dBA levels) then any low frequencies produced by a sound source would be severely attenuated after A-weighting and high frequencies would be attenuated due to atmospheric absorption and it was therefore considered that this limited bandwidth would not compromise the experiment.

An oscillator producing a logarithmically swept tone from 2000 Hz to 200 Hz was connected to a 200 watt amplifier to power the loudspeaker.

Field tests of the source indicated that the maximum output of the loudspeaker was 6 dB less than expected from manufacturer's figures. The on-axis frequency response was also not as flat over the operating band as that stated by the manufacturer. However, the loudspeakers performed satisfactorily and reliably in the field.

B. Sound Measurement

Metrosonics Metrologgers type dB-301 were selected for the continuous logging of sound levels. These instruments were set to record 1 minute A-weighted Leq's which they can store in internal memory for up to 8 hours. At the furthest distances, Bruel and Kjaer type 2203 sound level meters fitted with type 4165 microphones were used as inputs to the Metrologgers because of their lower inherent electrical noise. At the end of a measurement period a Metrosonics Metroreader type dB-651 or dB-652 was used to read the memory contents of the loggers and print out the levels. Using the RS-232 interface on the Metroreader, these data were also transferred to micro-cassette via a Hewlett Packard HP75 portable computer.

Large foam windscreens were made to protect the microphone diaphragms from wind turbulence.

To improve discrimination between the source signal and ambient noise, filters have been designed to fit between the 2203 sound level meter and the Metrologger. The filters have a bandwidth of 200 Hz to 2000 Hz.

Kudelski Nagra IVSJ tape-recorders were used to gather information on the manner in which the source spectrum changes with distance.

C. Meteorological Measurement

In order to examine the degree to which meteorology affects the propagation of sound over a distance of several kilometres, a system produced by AIR Corporation of Colorado, U.S.A. was purchased. Called a Tethersonde it consists of a tethered 4.9 metre long, cigar shaped, helium filled balloon below which is attached a lightweight met transducer package which relays data via a radio link to a receiver on the ground. The receiver decodes the data and it is transferred via an RS-232 output and a Sharp PC 1500 portable computer to a printer and to a cassette for storage.

The height of the balloon is controlled by an electrically operated winch. Operation of the balloon is limited to a maximum height of 1000 metres and wind speeds below 10 metres per second.

Data gathered using this system are wind speed, wind direction, air temperature, wet bulb temperature and atmospheric pressure. From these parameters relative humidity and measurement height can be calculated.

Three ground meteorology stations were dispersed along the propagation path to monitor variations in wind speed, wind direction, air temperature and humidity.

D. Analysis Equipment

A Tektronix type 4052A desktop graphics computer is used to edit and analyse the data from both the sound loggers and the meteorological system.
Graphical presentations of 1 minute Leq's versus distance for a measurement period may be produced as well as graphs of attenuation rates for different times of the day, different times of the year and different locations.
Using temperature and wind speed profiles, sound rays may be calculated and plotted for any elevation and direction.

A Bruel and Kjaer type 2131 real time analyser controlled from the Tektronix 4052A is used to analyse the audio tape-recordings.

MEASUREMENT PROCEDURE

The loudspeaker was mounted 2 metres above the ground, in a yoke on top of a box trailer so that it could be traversed both vertically and horizontally. It was tilted up at an angle of about 10 degrees to give a vertical coverage of approximately 20 degrees to the horizon and a beamwidth of approximately 40 degrees.
The sound source was cycled 5 minutes on, 5 minutes off, for up to 8 hours, which is the maximum operating period for the loggers when recording 1 minute Leq's. Cycling the loudspeaker protected it from overheating and enabled ambient noise levels to be recorded.

For the initial set of field measurements at Singleton army range in N.S.W., loggers were placed at distances of 50, 500, 1000, 2000 and 4000 metres from the source in line with the centreline of the loudspeaker. Because of problems with insufficient source power and terrain problems, these positions were changed to 50, 400, 800, 1600 and 3200 metres. Later, in December 1985 at Singleton, extra loggers were used at distances of 100 and 200 metres from the source. Microphone heights were set at 1.2 metres and microphones were protected with large windscreens.

Tape-recordings were taken at distances of 50, 400, 800, 1600 and 3200 metres from the source, once every hour.

Where possible, the met balloon was flown and data gathered simultaneously with the sound level data.
Ground met was gathered every half hour at the source, at the 1600 metre position and at the 3200 metre position.
Cloud cover was photographed every hour from the source position.

Weather permitting, data were gathered over a 6 day period during a field trip. For three of these days, measurements were started an hour before sunrise and, for the other three days, measurements continued until an hour after sunset. Rain, strong winds and storms often modified this programme.

RESULTS

Some results are shown in the accompanying tables and graphs.

Figure 1 shows a typical day's measurements; Figure 2 shows the effect on levels of a change in atmospheric conditions in the late afternoon.

Figure 3 shows percentile attenuation curves for all days and all locations measured. The solid line represents an attenuation rate of 6 dBA per doubling of distance and 3 dBA per kilometre.

Figure 4 shows a typical printout of the meteorological data gathered by the Tethersonde.

Figure 5 gives an example of sound ray traces calculated using temperature and wind profiles for a nominated direction with respect to the source.

A comparison of third octave spectra at various distances from the source is shown in Figure 6.

SUMMARY

A. Field Locations

To date, measurements have been made at the following locations for a week each during the months indicated.

Singleton: June 1984, September 1984, December 1985.
Port Wakefield: November 1984.
Woomera: December 1984.
Innisfail: May 1986.

Measurements are proposed for the following locations for a week each during the seasons indicated.

Singleton: Autumn.
Port Wakefield: Winter.
Woomera: Winter
Innisfail: Wet (November).
A cooler temperate location (to be determined), for all four seasons.

B. Field Measurements

Much time has been spent on perfecting the methods of generating the noise and of collecting the sound level and meteorological data.

Noise level data can be gathered satisfactorily at distances up to 3.2 kilometres. A higher powered sound source would be required to take measurements at greater distances.

Two people are required to operate the meteorology balloon, which can be unstable under some conditions and can only be flown in wind speeds below 10 m/s. A more stable system requiring only one person to operate is desirable.

C. Data Reduction

Logger Data: All software has been written to store, edit and graph the logger data. Full editing has yet to be done, although this is expected to have little effect on the overall results.

The A_{90} figures for all days and all locations (Figure 3) show a remarkable similarity to the simple prediction of 6 dBA per doubling of distance plus 3 dBA per kilometre.

Tape-Recording: All tape-recordings to date have been analysed to obtain third octave spectra. Detailed examination of the data has yet to be done.

Meteorological Data: Software has been written to transfer meteorological data to the Tektronix 4052 desktop computer. Improvements to the ray tracing program are required so as to best utilise these data.

D. The Future

Already a substantial amount of data have been gathered and some overall trends have been noted. At the conclusion of the study, with a much larger data base, it should be possible to predict the effects on sound propagation due to the time of day, time of year or type of terrain. In addition, the field data can be used to test the accuracy of predictions of various theoretical and empirical sound propagation models.

With the advent of more powerful artillery weapons, there is a need to conduct a study of impulse noise propagation along similar lines to the continuous noise study.

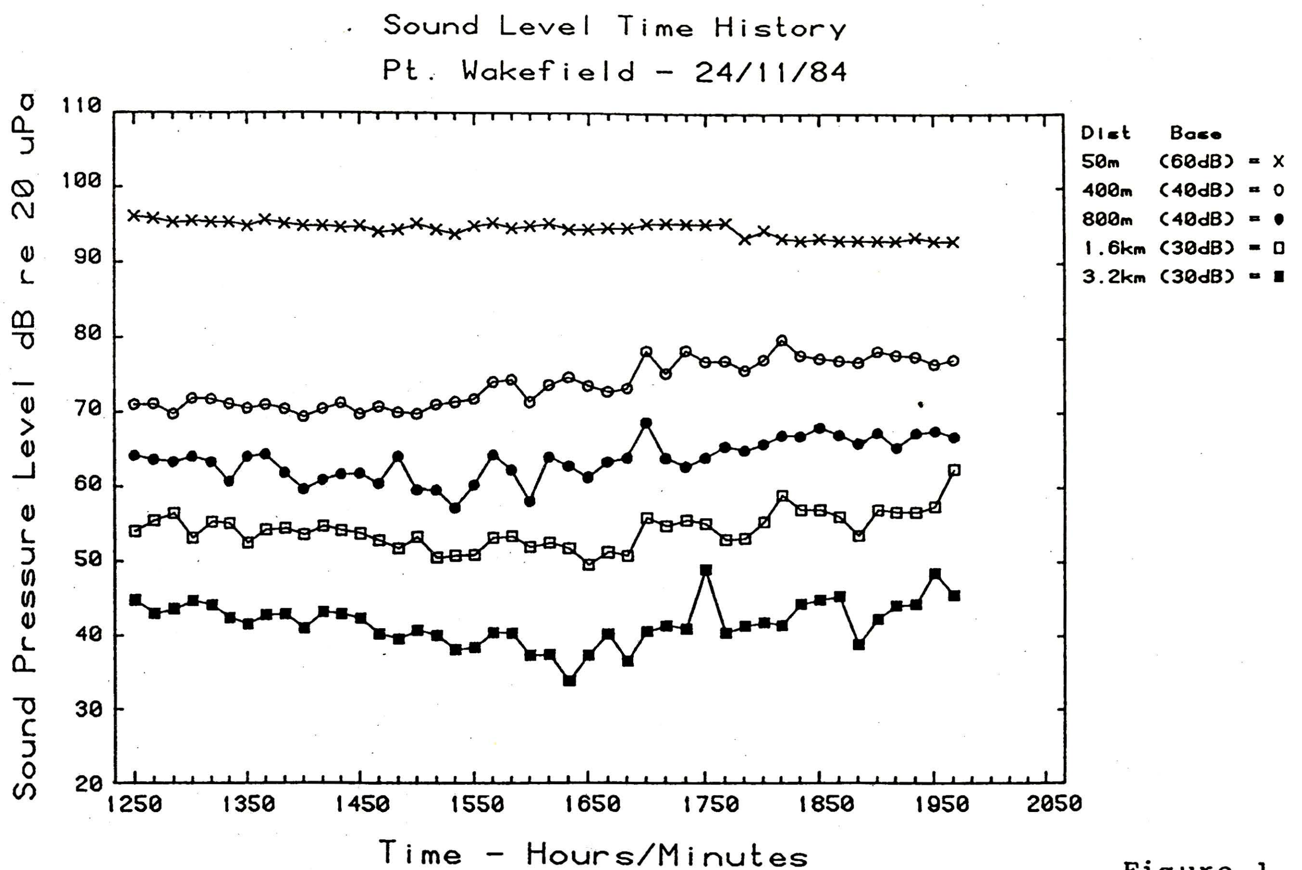

Figure 1

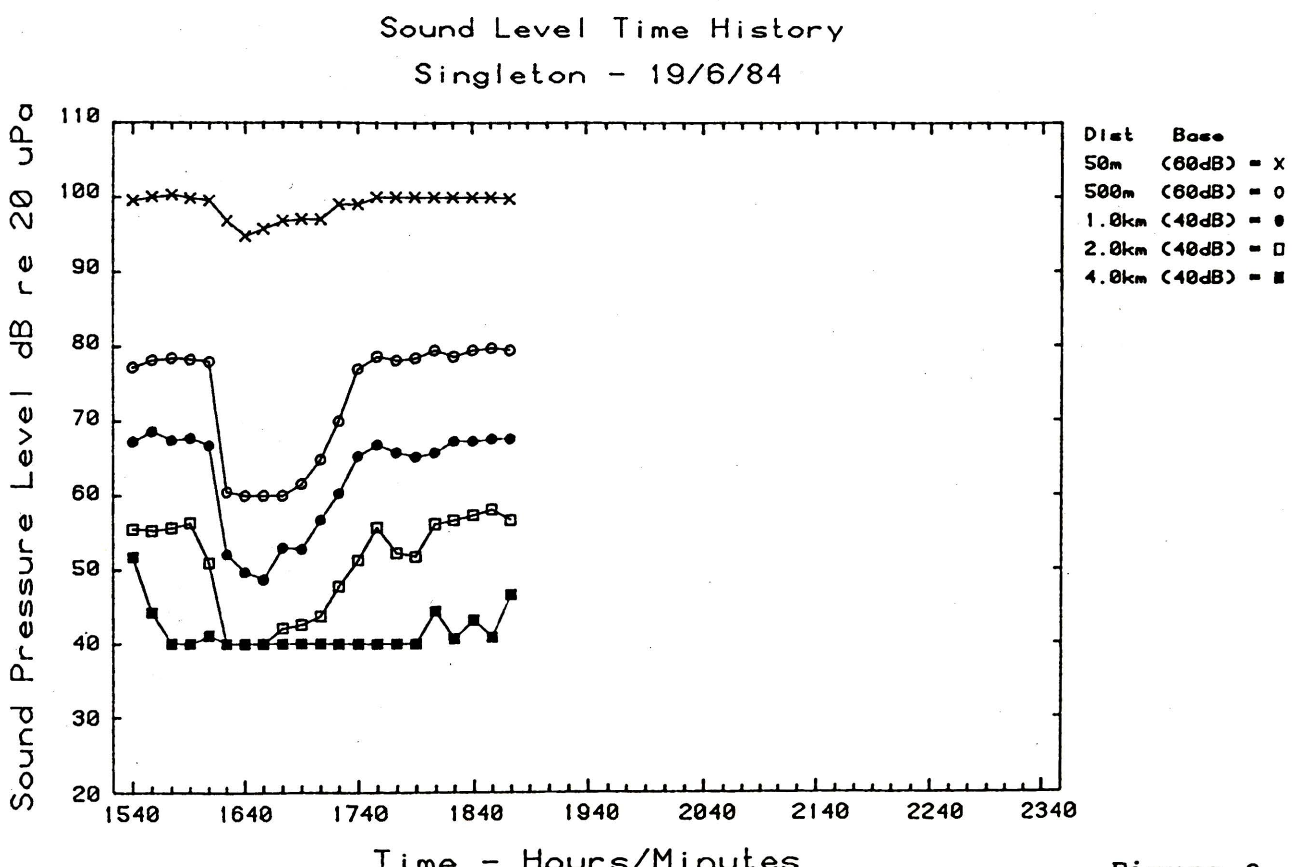

Figure 2

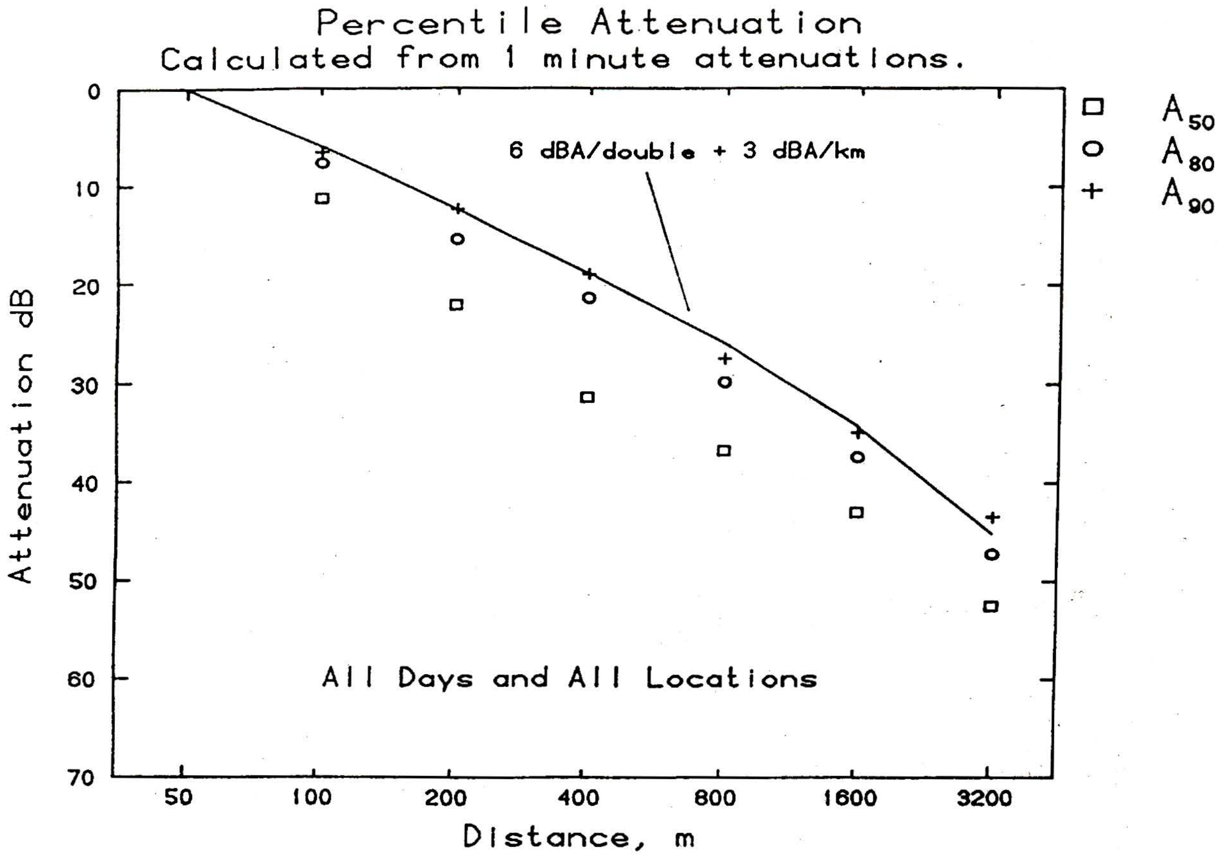

Figure 3

```
SINGLETON PROP.
DATE : 24th OF JUN
1984

*-:-*-:-*-:-*-:-*
DATA
AT INITIALISATION
DATE : 24th OF JUN

TIME      : 1146
TEMP DRY :  15.8 C
TEMP WET :  10.2 C
REL HUM  :  48  %
ATM PRESS: 1011 mB
HEIGHT    : 0 m
BATT CHG :  86 %
WIND SPD : 2.4m/s
WIND DIR :  339 d
WIND DIR : NNW
P2 =   11.41


         DATA
TIME      : 1148
TEMP DRY :  14.9 C
TEMP WET :  9.5 C
REL HUM  :  49  %
ATM PRESS: 1009 mB
HEIGHT    : 18.1 m
BATT CHG :  86 %
WIND SPD : 7.2m/s
WIND DIR :  333 d
WIND DIR : NNW
P2 =   13.5725
```

Figure 4

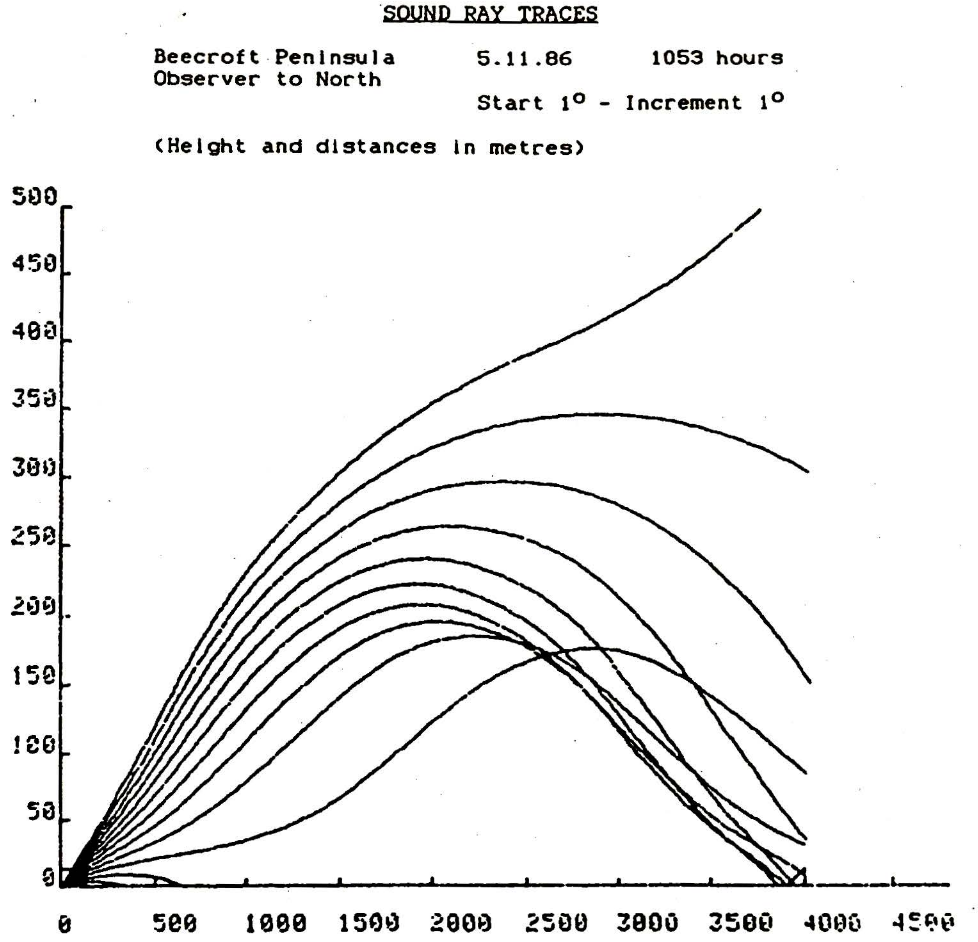

Figure 5

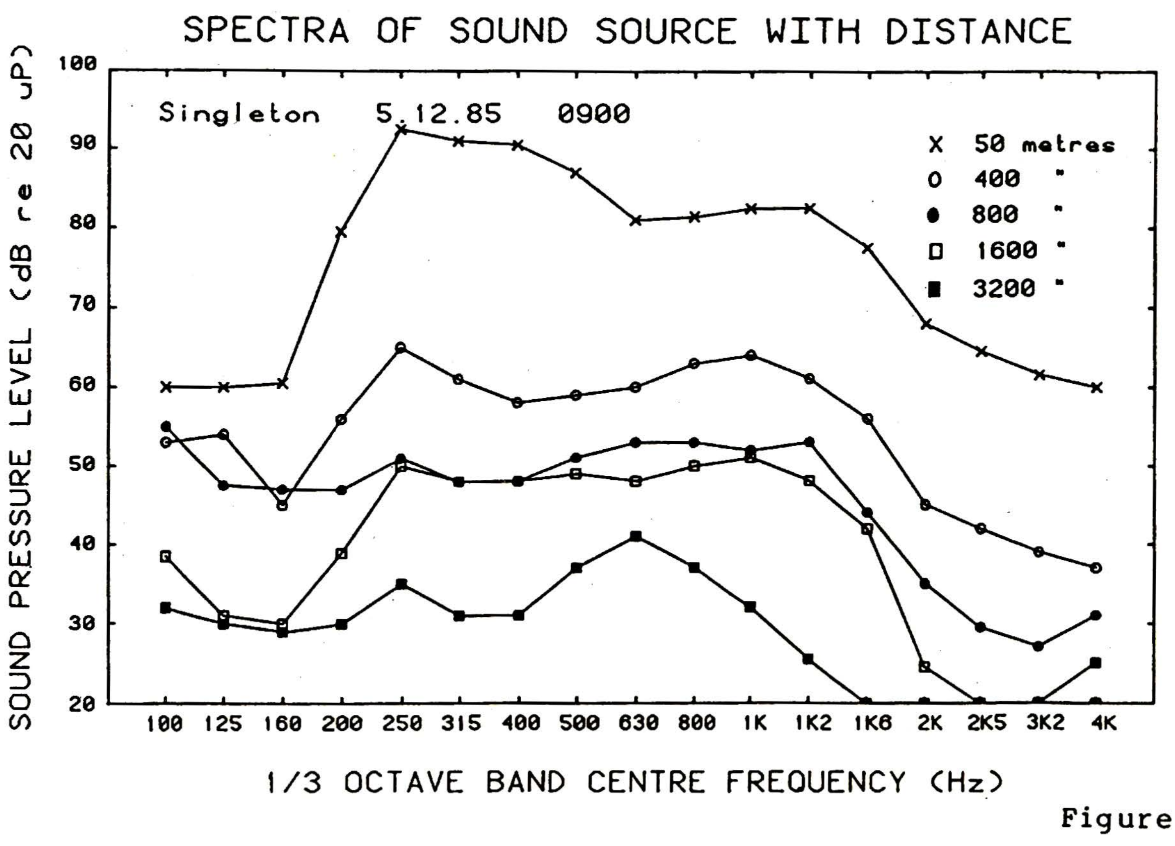

Figure 6

Acoustics in the Eighties

PREDICTION METHODS FOR OUTDOOR SOUND PROPAGATION

A.J. Madry

PREDICTION METHODS FOR OUTDOOR SOUND PROPAGATION

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N.S.W. 2006

ABSTRACT

This paper discusses the various methods of predicting sound levels outdoors and areas where inaccuracies are likely to occur. A theoretical model is presented based on a digital ray tracing method and the effects of diffraction are included by using diffracting rays. The model can simulate the effect of fluctuating atmospheric conditions and predict the range of sound levels likely to occur.

1. INTRODUCTION

Some effort has been put into developing models to predict sound levels caused by a noise source at some distance away from this source. There are basically two approaches that may be taken. One is to base predictions on measurements made at actual sites, the other is to use a theoretical approach. A combination of these approaches has been used by some. A model is described here which has a theoretical framework but attempts to simulate the conditions under which measurements are made.

2. DISCUSSION OF PREDICTION METHODS

The accuracy of any prediction method depends on what the predicted quantity is. This point is raised by Moerkerken (1985) in a review dealing with the Dutch propagation model. In this model the predicted quantity is the long term equivalent sound level. Predictions are based on empirical data gathered at various sites and an averaging over all possible conditions is made. This will include the effects of refraction and turbulence and variations of ground impedance due to seasonal changes of moisture content.

Such an approach is aimed at minimising the inaccuracy. Moerkerken points out though, that to compare with the predictions more than one measurement is needed, even for one particular meteorological condition. For downwind conditions the long term equivalent sound level can vary up to 8dB. For upwind conditions larger variations may be expected.

The empirical approach is very appealing for the reason that predictions made this way are known to be physically realisable. Measurements of the sound level of a known source are made at certain locations on a certain site under various conditions. These conditions may be divided into categories (e.g. upwind, downwind, neutral). Prediction of attenuation of a source may be obtained from this data. Effects difficult to deal with theoretically such as atmospheric turbulence are already incorporated.

Terrain profile and ground cover will vary widely from site to site so it should not be expected that predictions for one site should be valid for another. Any accuracy quoted for an empirical prediction method is really only relevant to the site where the measurements were made. Most prediction models available are based on data from European sites. One should not expect such models to be applicable to Australian sites. What may be described as grassland in Europe may be quite different in acoustic properties to what we consider grassland here, (Madry, 1986).

There may be many factors affecting propagation in one particular case. The more of these there are the more complicated it becomes to gather specific experimental data. An example is if there is some barrier between source and receiver. It is feasable to get experimental results for various barriers under one particular weather condition. However once various weather categories are to be included the amount of data to be gathered multiplies by the number of categories considered. The time needed for such a procedure is prohibitive.

Once the physical situation becomes more complicated the advantages of a purely theoretical model become apparent. Varying the conditions is done easily. Whether such a model gives realistic results will only be found by comparison with measurements in particular cases.

For a theoretical model to be genuinely accurate one would need detailed information about all the parameters affecting sound propagation. This would mean having meteorological equipment measuring wind and temperature profiles positioned at closely spaced intervals along the propagation path. Such a procedure would be impractical. Realistically the best one could hope for is information about the meteorological variables at say two heights at one or two locations along the propagation path. Values at other locations must be obtained by interpolation and can be expected to fluctuate due to unstable atmospheric conditions.

3. ANOTHER PREDICTION METHOD

Theoretical methods suitable for use in outdoor sound propagation have been studied (Madry, 1986). The numerical integration method of Rasmussen (1985) is accurate and can deal with refraction, the ground effect, changes in ground impedance and barriers. It requires a flat ground surface and computational time limits its application to shorter distances (less than 400m). An important factor not included is the effect of atmospheric turbulence.

The ray tracing method has a definite advantage in terms of computational ease and also in terms of explaining the physical processes occurring. However, there are problems in dealing with some situations. The ray method leads to regions known as shadow zones where no rays pass (e.g. behind a barrier or where there is some limiting ray caused by upward refraction). This can be dealt with using the geometrical theory of diffraction. Diffracted rays penetrate these regions. Another problem is that focussing of rays can lead to prediction of infinite amplitudes. The usual methods of calculating the amplitude are not valid in these regions.

A ray tracing method has been implemented based on the method used by Andersen and Kak (1982) for investigating the effect of a two dimensional refractive index field on ultrasonic imaging. This is suitable for investigating the effect of fluctuating meteorological profiles on ray paths. Rays are stepped out by small increments determined by solving a partial differential equation using values of the refractive index obtained by interpolation from a two-dimensional grid. Any particular meteorological profile can be set up by specifying the speed of sound at the discrete positions of the grid.

A ray that strikes the ground is reflected at an angle equal to the angle of incidence. The phase and amplitude of reflected rays is modified by the spherical reflection coefficient. Huisman and Martens (1986) have speculated that this may not be accurate and suggest that the plane wave reflection coefficient may be applicable for cases when rays striking the ground are parallel.

A ray striking a barrier is stopped in its path and the top of a barrier becomes a source of diffracted rays. The diffracted rays must be considered separately. The amplitude of diffracted rays varies with distance differently to conventional rays. A comparison between the number of rays when there is no refraction and when there is refraction is still valid. The amplitude and phase of these rays are also modified after striking a barrier

An example of a ray plot for a receiver downwind of a source is given in fig. 1a. It can be seen that as distance increases the ray pattern becomes much more complicated. Rays that have been reflected from the ground more than once reach the receiver. The method of calculating the effect of a direct ray and one reflected ray with some correction for windspeed is an over-simplification.

To determine the raypaths linking a source and receiver the model requires the user to identify the angles within which rays leaving the source will reach a receiver. This can be done using a broad sweep of rays as in fig. 1a. An accurate estimation can be obtained by performing a fine sweep over a smaller angular range. The 'shooting method' described by Andersen and Kak (1982) can be employed. The height of a ray at the distance of the receiver is plotted as a function of angle. The linking ray launch angle is determined by interpolation (see fig. 2).

Once the ray paths are identified the amplitude and phase are found using the method of Huisman and Martens (1986). The spread of a beam of rays, A_1, in a particular refractive index field at a given distance is compared with the corresponding spread, A_0, under free field conditions. These can be determined from the data in fig. 2. The attenuation due to spherical spreading of a point source in free field conditions is given by:

$$\text{attenuation} = 10 \log (4\pi R^2)$$

where R is the distance from source to receiver. The correction to this to account for spreading or focussing of a beam of rays is:

$$\text{beam correction} = 10 \log (A_1/A_0)$$

Except near regions of focussing or in highly non-linear gradients this factor is small.

The phase in many cases however, is significant. The phase is given by:

$$\text{phase} = (2\pi/\text{wavelength}) \times \text{pathlength}$$

Each of the sound paths linking a source and receiver produces a sound pressure at the receiver. These are combined to give the resultant level. The phase relationships between the paths can be quite critical and determine whether interference is constructive or destructive. The difference in the result can be of the order of 10 dB. A significant change in the pathlength, with respect to the wavelength, can be caused by only a slight change in windspeed for distances greater than 100m. It can be seen that the ray plot of fig. 1a, which is for a light downwind is very different to what one would expect for the zero wind case.

In practice as the meteorological profiles fluctuate so too do the raypaths. A realistic way of modelling the effect of the atmosphere is to vary the refractive index profiles and compute the different raypaths. The resultant sound pressure levels are calculated and may be averaged to give a prediction value. This seems a reasonable way to theoretically predict levels when one only has limited information about meteorological variables. The predicted value should be compared with a measured value taken as an average over some time interval, over which the profiles input to the model may be thought to occur. The range over which the sound pressure level may fluctuate can also be estimated this way.

It is often assumed that one should take the average profile as input for a model to give the best prediction. This is not correct. One should use various instantaneous profiles as input and take the average of the levels produced. This approach has been proposed by Huisman and Martens (1986).

In an experiment, pure tones were propagated at distances up to 200m. It can be seen from fig.3 that the sound pressure level fluctuates widely for the light wind conditions that were encountered. Changes in the coherence of direct and reflected rays could explain these fluctuations. There are also fluctuations in the phase and amplitude of individual rays due to the fact that wind and temperature vary randomly about a mean value as a result of atmospheric turbulence. It is still unclear as to the influence of this on results such as in fig. 3.

4. APPLICATIONS OF THE MODEL

The main use of the model presented here is in identifying propagation paths. Refraction of sound from inversions in upper layers can be predicted. Rays which in neutral conditions would only reach a receiver by a diffracted path over a barrier may be found to link a source and receiver directly when there is downward refraction (see fig. 1b).

The model cannot predict sound levels accurately under any conditions. In regions of focussing or shadow zones the method can only predict unusually large or small sound levels respectively. The location of these regions however is often of interest. The model is useful in the prediction of the variation of levels that may be expected.

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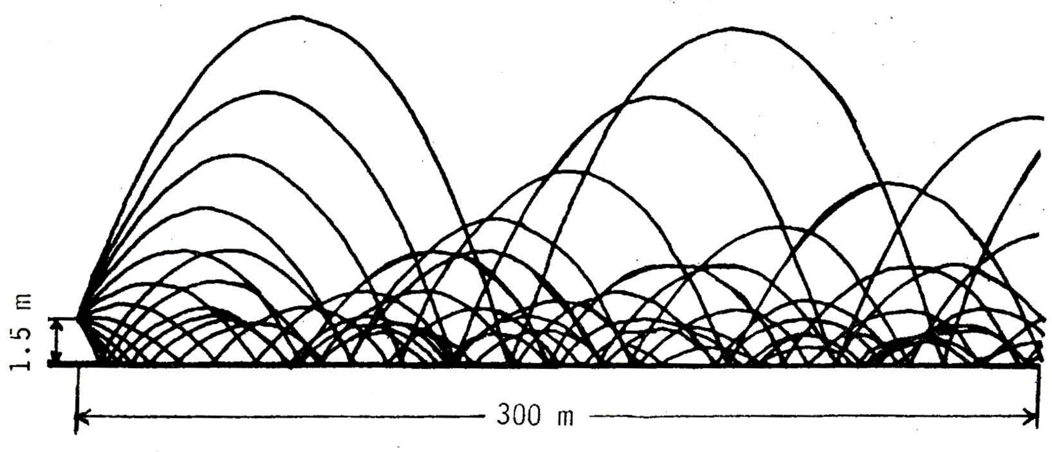

Figure 1a. Plot of rays for a receiver downwind of source. Windspeed = 2 ms^{-1}. The vertical scale has been expanded for clarity.

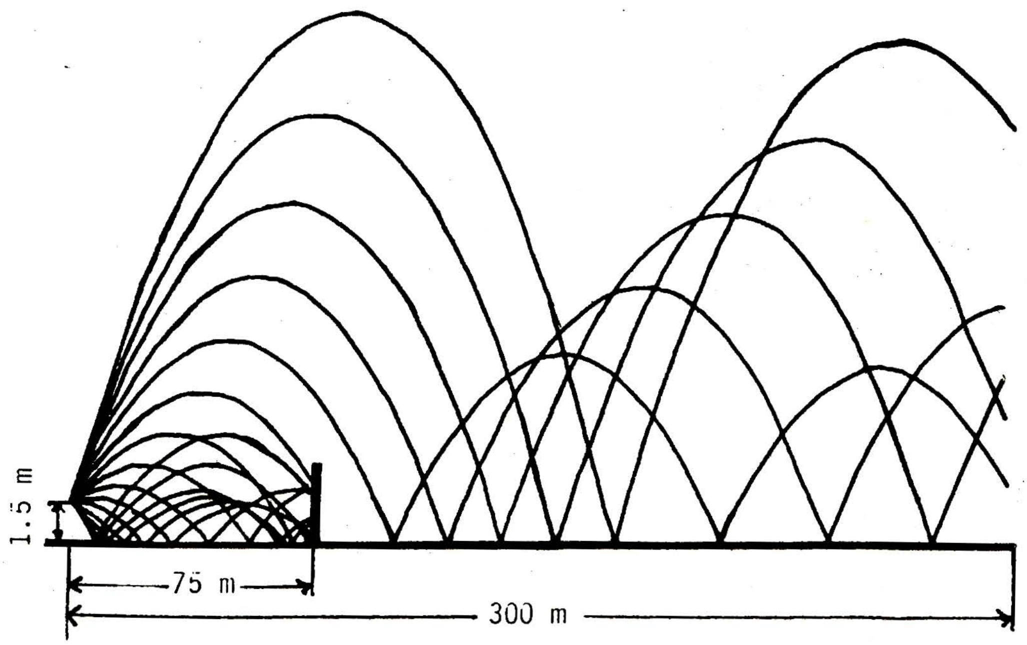

Figure 1b. Ray plot for conditions of fig.1a over a larger angle with a barrier inserted.

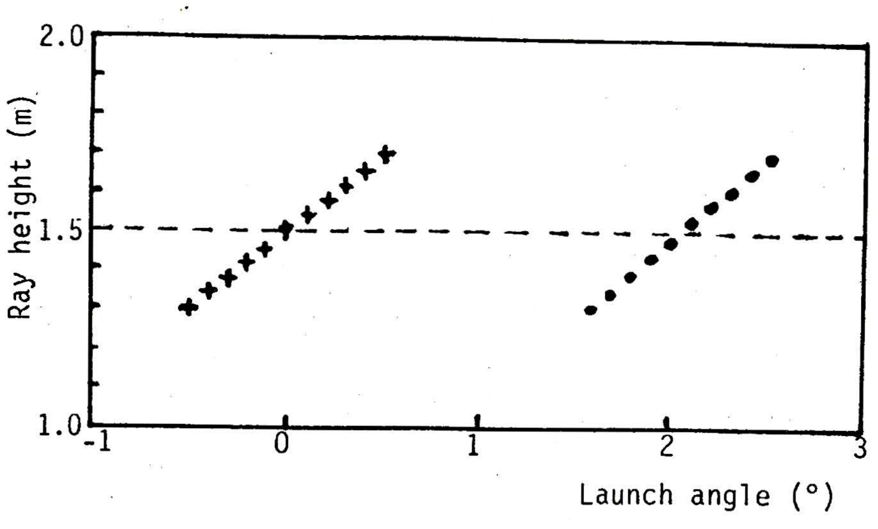

Figure 2. Ray height vs launch angle. Receiver distance = 25 m, source height = 1.5 m. ● windspeed = $1ms^{-1}$ receiver downwind of source, ✚ zero wind.

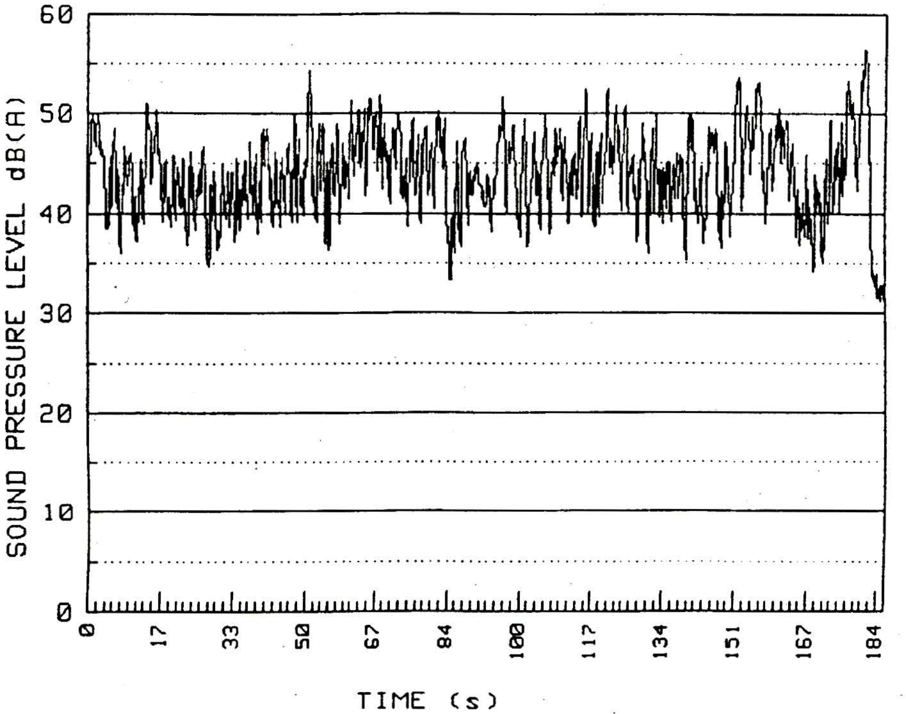

Figure 3. Sound pressure level vs time of a 1 kHz pure tone propagated 200 m over grass.

Acoustics in the Eighties

Session 1C: Music and Theatre Acoustics

Acoustics in the Eighties

EXHIBITION ACOUSTICS — THE MUSEUM AS THEATRE

P. Griffiths

EXHIBITION ACOUSTICS:

THE MUSEUM AS THEATRE

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ABSTRACT

In their contemporary context, museums are required to be more than a collection of objects. They are now conceived as complex environments providing many multi stimulus experiences for visitors. A museum and its exhibition experience can be considered as being similar to a theatre and an associated theatrical experience. Just as all theatrical performance contains an acoustical component, acoustics can also play a major role in the success or failure of an exhibition experience. This analogy between theatre and exhibition has been used as the conceptual basis for the acoustic design of the highly acclaimed Australian Pavilion for Expo 85, designed by the Department of Housing & Construction. Further refinement of this acoustical approach is being used for a number of museums and exhibition centres currently being designed.

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Richard Johnson, Denton Corker Marshall Pty Ltd.

1. INTRODUCTION

In their contemporary context, museums are required to be more than a collection of objects. They are now conceived as complex environments able to provide many multi-stimulus experiences for visitors, thereby attempting to extend and broaden the traditional concept of the exhibition experience.

Sound and the way it behaves are integral components of any environment. When considered as design elements which can be controlled and manipulated these many facetted acoustical aspects become particularly powerful tools in the development of the proposed exhibitions. Not only can acoustics be part of an exhibit, but it can also be the exhibit itself.

2. EXHIBITION ACOUSTIC DESIGN

What are we trying to achieve with exhibition acoustic design?

This question must be answered in two areas:

(a) Building acoustics concerned with the provision of an acoustically suitable environment for the exhibition function. This can be interpreted as the passive component of the overall acoustic design in the sense that once created it remains constant for the Museum visitor. This component is seen as a function of the building fabric and structure in controlling undesirable extraneous noise and in determining the internal acoustic performance of the building and major galleries.

(b) Exhibition acoustics, concerned with the exhibition experience. This is regarded as the active component of the acoustical design in that it builds upon the basic building acoustics and results in a high degree of interaction with the museum visitor.

The Museum and the exhibition experience can be considered as a theatre and an associated theatrical experience. In a broad sense, a building and its many spatial components will present an overall performance to an audience, and just as an audience interacts with a performance in a more conventional theatre, so too can an audience be stimulated to interact with the Museum performance. However, unlike the conventional theatre where the audience tends to be statically located and the performers dynamic, the Museum audience will be dynamic and the Museum performance static. It follows that the Museum/Theatre becomes a series of performance spaces that appear to the audience as a progression of theatrical events within an overall context. These events must be a success both individually and as a component of the theatrical whole.

The theatrical analogy demonstrates the dual acoustic role of the museum. In a theatre a suitable acoustic environment must be provided as the basis for the success of an undisturbed performance. Such an environment is provided by the physical context of the building or area in which the performance takes place. All theatrical performance contains an acoustic experience. Quite often the success or failure of the performance will rest upon such an experience.

Similarly, the Museum must provide a basic acoustic environment that will allow the exhibits to be experienced in comfort. In addition, the exhibits themselves can have or can be acoustic components forming part of a multi-stimulus environment.

This view of the Museum as theatre and exhibition as performance is fundamental to the exhibition acoustic design concept.

3. BUILDING ACOUSTICS

The acoustic design of a Museum or exhibition building must provide an environment which will allow visitors the full potential to experience the exhibitions without interruption, disturbance, annoyance, and stress brought about by extraneous, undesirable and uncontrolled noise sources.

This aspect will be familiar to all acousticians involved with architectural acoustics and covers such areas as traffic noise control, rain noise control, reverberation control, control of echos in large galleries, air conditioning noise control, internal acoustics for specialist areas such as theatres and electro-acoustic design.

For many museum and exhibition areas, the concept of building acoustics must be extended to include crowd noise control, acoustical control within architecturally defined areas such as visitor rest areas and meeting rooms, acoustic control of areas designed for large scale audio visual presentations, and acoustic control of areas required to be suitable for live music performances.

4. EXHIBITION ACOUSTICS

Exhibition Acoustics is concerned with the enhancement of exhibition appeal. This can be most effectively achieved by emphasising objects and acoustical environments by integrating room acoustics, sound control and production for all museum components, including the overall, the major galleries through to singular exhibits and exhibit elements.

This approach allows for the development of an acoustic environment which is unified, a component of, and complementary to all aspects of an exhibition, whilst still recognising the spatial and acoustic properties of the building as a whole and the major elements thereof. This approach should be seen as the optimisation of a number of acoustic elements with the physical constraints of the building and the exhibition objectives.

Two acoustic design parameters exist. These are:

(a) Sound (the absence or presence of sound, its character, its function).

(b) Room acoustics (how the sound behaves in any space).

The basis of exhibition acoustics design is that these parameters be controlled to provide:

(a) Unifying exhibit acoustical themes.

(b) Complementary acoustic interaction between spatial elements.

(c) Preferred circulation of museum visitors.

These points summarise the functionality of the exhibition acoustic design concept. Themes create orientation, context and a sense of unity; complementary interaction avoids conflict and confusion between exhibits and exhibit areas; preferred circulation establishes order, progression and contrast of exhibits and exhibit areas.

5. AUSTRALIAN PAVILION, EXPO 85, TSUKUBA, JAPAN

The official theme for Expo 85 was "Dwellings and Surroundings - Science and Technology for Man at Home". In accordance with this overall theme, the themes of the Australian Exhibition were developed sequentially as shown in Table 1.

TABLE 1

THEMES OF THE AUSTRALIAN EXHIBITION

Main Theme	Subtheme
1. Living - Past	A - Origins: - Aboriginal Culture - Colonisation - Immigration
2. Living - Present	B - Australian Dwellings and Surroundings C - Technology in the Service of Australians
3. Living - Future	D - The Inventive Australian
4. Summary Statement - "Living in Australia, Past, Present and Future.	E - The People, Lifestyle F - Fauna and Flora.

The Acoustic Design Brief was based on the following exhibition requirements:

(a) Simplification and limitation of the range of exhibited material in order to prevent visitors leaving with a confused image of the Exhibition.

(b) Provision of various levels of information within each exhibit.

(c) Satisfying the audience's expectation of entertainment as well as information.

(d) Control of ambient noise levels and exhibit noise levels to alleviate museum fatigue.

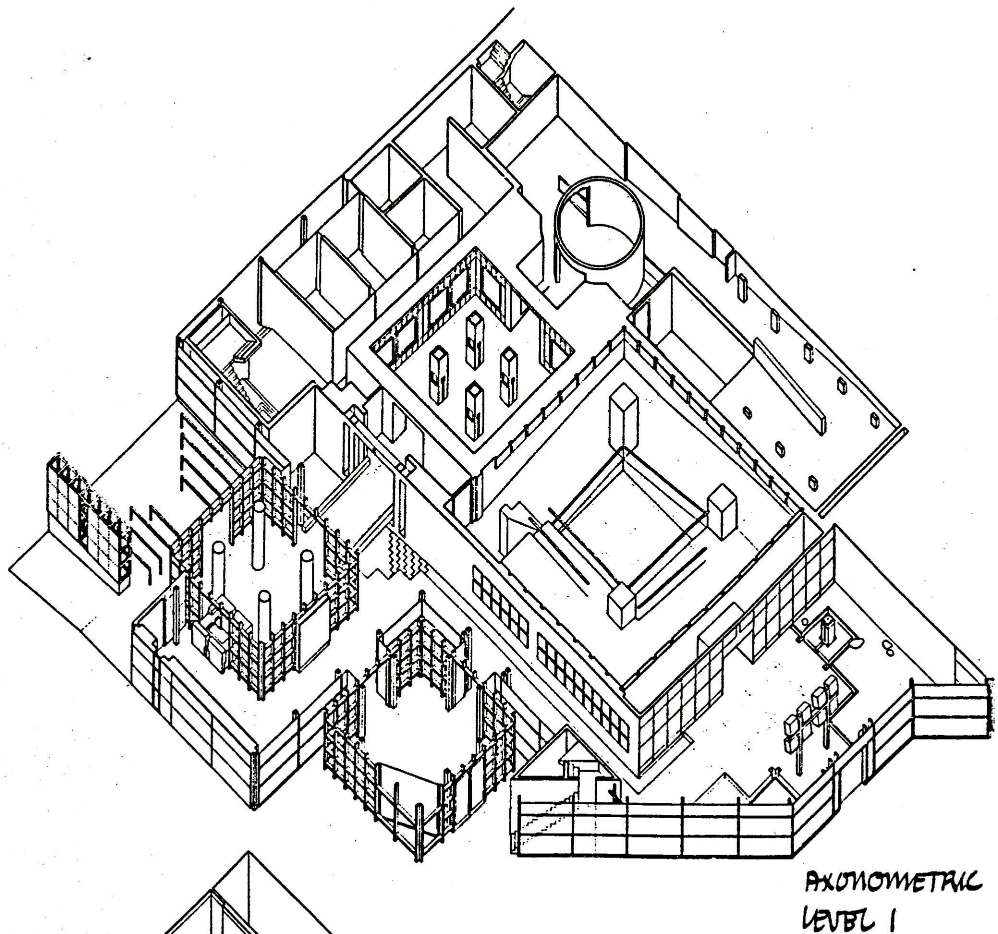
For this Exhibition, two major problems were identified as existing beyond the traditional noise controls required. These were:

(a) Exhibition galleries could not be acoustically isolated in terms of noise transfer because of open doorways and passages to allow visitor movement.

These galleries are shown in Figure 1.

(b) The high impact audio-visual space required large areas of glazing for projected images. This was in conflict with the requirement of a low reverberation time in order to maximise the effect of the soundtrack reproduction.

This Audio-visual Space is shown in Figure 2.



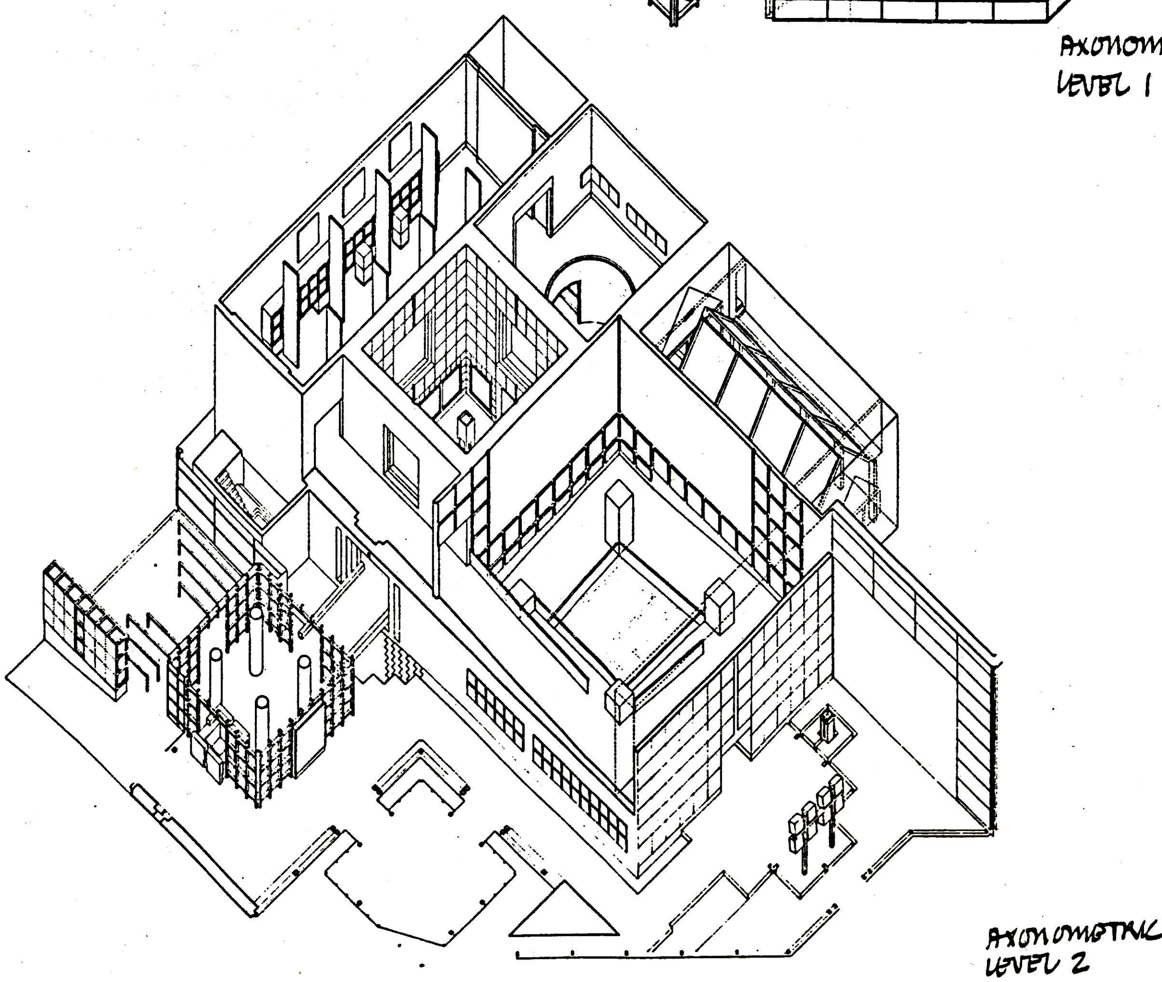
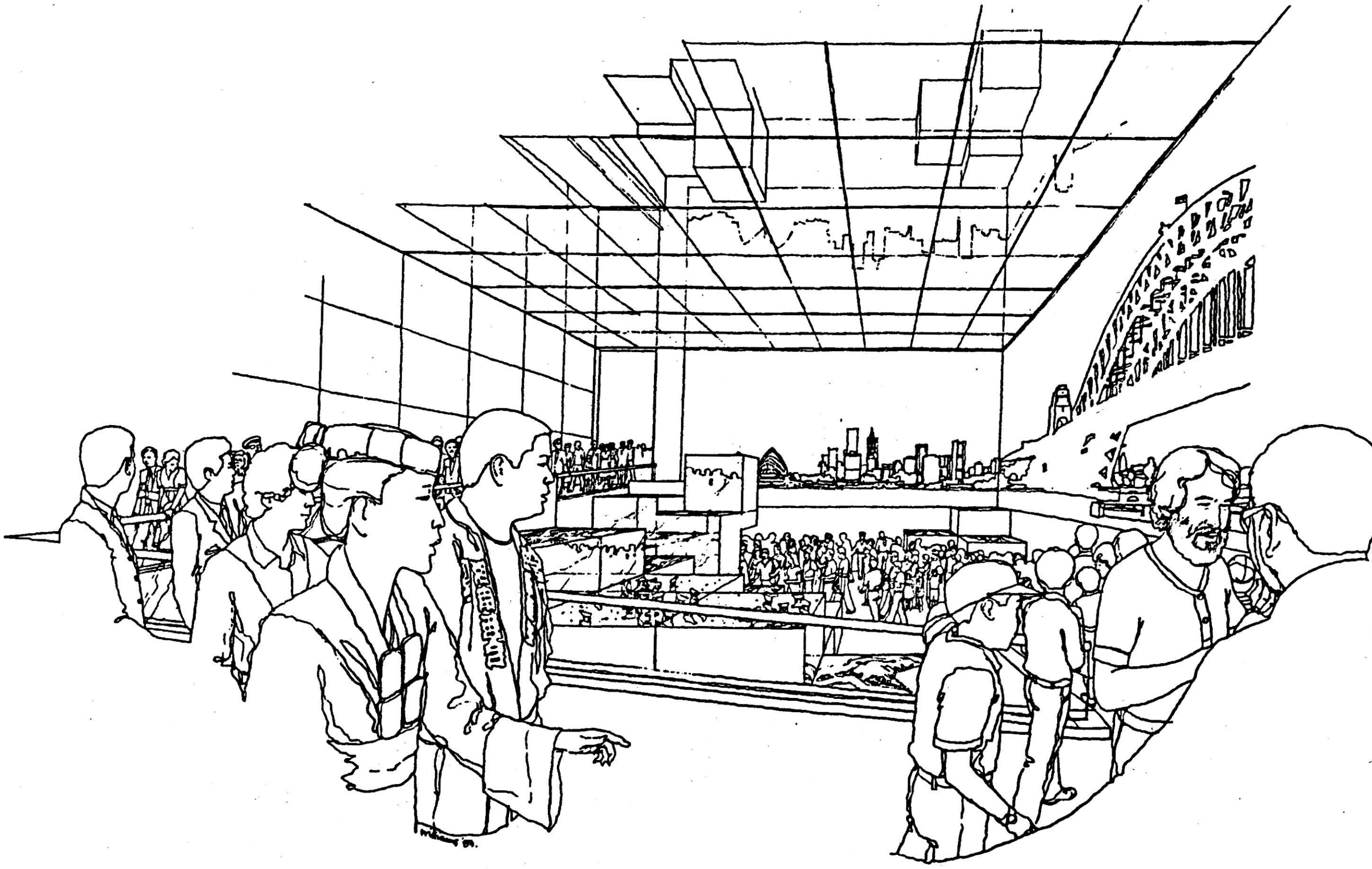
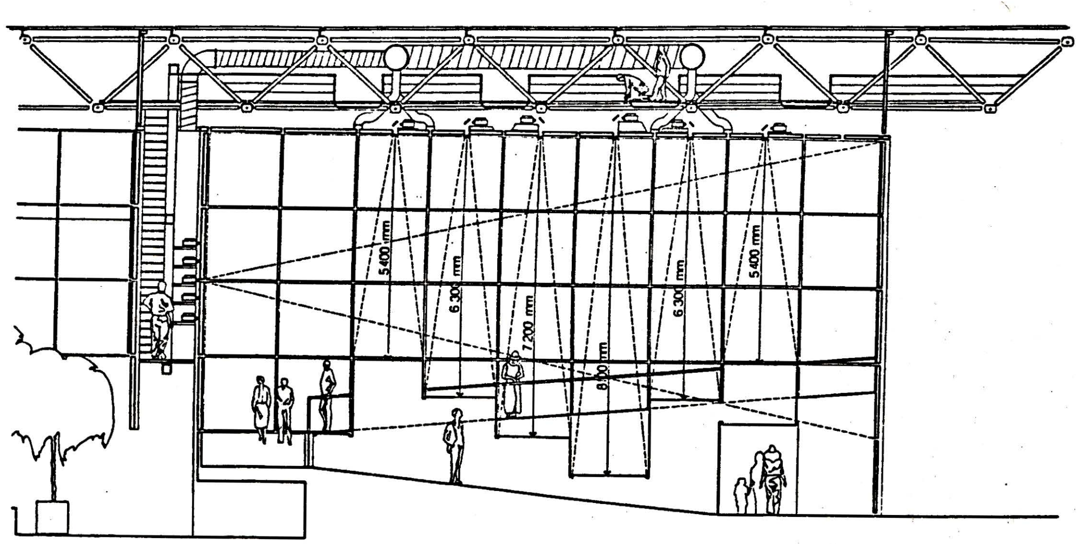


FIGURE 1 : AUSTRALIAN PAVILION, EXPO 85



PERSPECTIVE VIEW
AUDIO-VISUAL THEATRE



SECTION THROUGH
AUDIO VISUAL THEATRE

FIGURE 2 : AUDIO-VISUAL THEATRE

5.1 Reverberation Time Control

The reverberation times of subsequent galleries were designed progressively lower ranging from a maximum for the Entrance Hall to a minimum for the large audio-visual space, the major space in the pavilion. The conflict between reflective screens and absorbent panels in the Audio-visual Space was resolved by having screens on two adjacent walls and absorption on the opposite walls. Whilst the floor was reflective (except for carpet on the ramps) the ceiling was made as absorbtive as possible with fibreglass behind panels of reflective Mylar. The Mylar was fixed on aluminium frames with gaps between frames.

5.2 Sound Control Between Galleries

As acoustic isolation between galleries was limited, a decision was made at an early design stage to allow various audio signals from areas in the exhibition to co-exist rather than attempt to consider each audio visual component in isolation.

This led to the following set of rules for the Audio Visual Producers:

(a) All audio components must be considered as a part of an overall production, not a series of separate productions.

(b) Programmes containing speech should be avoided and must not be designed to exist ajacent to other speech oriented programmes.

(c) Music and sound effects tracks from adjacent sources should be rhythmically and harmonically related.

The other major problem was the audio component of the audio visual space. This was to be of high level for maximum impact and was not to be in conflict with the overall audio track being played throughout the pavilion. This was achieved by composing an overall sound track of which the pavilion component and the audio visual space component were rhythmically and harmonically related parts able to be heard simultaneously without conflict but also able to stand alone.

6. CURRENT PROJECTS

Other Museums and exhibitions currently being designed taking account of acoustic problems are the Power House Museum, Stage 2 and the First State 88 Exhibition which will open the Darling Harbour Exhibition Centre in Sydney.

6.1 Power House Museum

The Power House Museum is distinctive because of the very large volumes of the main galleries, the many audio-visual components throughout, the number and diversity of exhibit themes, and the building itself being a major element of exhibition.

However, the same basic rules apply to the exhibition acoustic design with regard to the blending of acoustic sources rather than isolation, and the need for audio design integrity at the design and production stage of audio-visual presentations.

6.2 First State 88 Exhibition

The design of this Exhibition represents a highly theatrical base, to the extent that much of the exhibition design is created as stage sets. A great emphasis has been placed on high impact entertainment exhibits which will exist with very little acoustic separation between exhibit component areas. The approach has been to design the audio components as if the visitor will be "moving between the instruments of an orchestra". In conjunction with this design approach, varying levels of acoustic absorption have been designed into the various exhibits to create differing sound fields as appropriate to the particular exhibit theme.

7. CONCLUSION

The Exhibition experience can be brought to a high impact level by treating the Exhibition enclosure and the Exhibition acoustic components in ways similar to theatre acoustics. Reverberation and room acoustics require careful manipulation in order to create varying environments consistent with the appropriate Exhibition thematic concept. Varying noise sources (both extraneous and those generated by the Exhibition components themselves) must be controlled and related to provide sound fields that are co-ordinated, complementary to and sympathetic components of the Exhibition as a whole, rather than causing fatigue and confusion to the Museum visitor.

NON-AUDITORY INFLUENCES ON THE PERCEPTION OF SOUND IN CONCERT HALLS

M. Palavidis and F. Fricke

NON-AUDITORY INFLUENCES ON THE PERCEPTION OF SOUND IN CONCERT HALLS

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ABSTRACT

The perception of sound in a concert hall is a very individual experience. Consensus may be had on a number of aspects of a performance and a performance space but the overall impressions of the audience will not generally be polarized. How do we explain this divergence of the listener's auditory perception?

Given that the acoustic design of the concert hall is satisfactory, it is logical to conclude that the individual's perception of sound will depend upon factors other than auditory ones. One must consider the influence of polysensory impression and interactioon; in particular it is important to consider that response to the sensory environment inside a concert hall will be the product of a limitless range of physiological and psychological experiences. A person's past experience of music and concert halls may also be important in the determination of the quality of sound.

In an attempt to evaluate the influence of the phenomena on the perception of sound a questionnaire was designed and distributed to audiences attending musical performances. The results of the questionnaire highlighted a number of interesting relations on the polysensory perception of sound.

1. INTRODUCTION

The aim of this paper is to introduce the vast amount of information avaliable on the subject of non-auditory influences on the perception of sound. It is the theme of this paper to relate this information to the perception of sound in concert halls. However, in researching this topic, no information was found that directly linked bi-modal perception to the design of concert halls. Most of the work performed in this area relates to branches of Psychology and Neurophysiology concerned with perception and behaviour. In particular the sub-topics of inter-sensory perception, sensory interaction and synesthesia are dealt with.

The material presented in this study will be supported by the results of a psycho-acoustic survey of audiences attending musical performances held in the Everest Theatre chamber music hall at the Seymour Centre in Sydney. The results of this survey highlighted a number of interesting correlations between the audience's auditory experience of the performance and other sensory and environmental influences.

2. PERCEPTION

One area of thought that this paper is concerned with is a view some philosphers and psychologists advocate; all information from the environment reaching the brain must first pass through sensory receptors and their neural pathways. It is not the intent of this paper to establish the accuracy of this dictum. The school of thought which holds that all information must pass through our senses provides a structure in which the physical world provides stimulus. This subsequently results in a person receiving sensations, and the interpretation of this effect is perception. Perception can therefore be defined as the response of a person to a stimulus. It is a process which includes sensation, memory and thought and which results in meaning such as recognition, identification and understanding.

Sensory information is collected in an on-going dynamic process involving the sensory receptors which continually attend to change in the environment. The senses work together, as required, actively looking, listening, touching, smelling and tasting. Seeing a fire, for example, involves much more than than the eyes and the limited information they provide. One also feels the heat and hears the cracking of the wood as it is slowly consumed by the rage of the flames. Experiencing fire is a multisenory process. The senses work together as needed and, through the process of perception, information is intergrated with memory and cognition. It is believed that a similar multisensory process is involved in hearing a concert. One does not just listen to the sound of the performance but one also needs to see the performance . This belief was dually reinforced by the findings of a social survey. This survey showed that listening to the performance was not a simple case of hearing its sound. If this contention is correct it has important inferences for the design of concert halls and the study of room acoustics.

3. UNITY OF THE SENSES

In the preceeding section the relationship between stimulus and sensory response was introduced and briefly discussed. It is the mechanism of this process that we now focus upon. Specifically, it is the relationship of cooperation and interrelation between the individual senses in the process of perception or multisensory reponse.

Shigehisa, P.M.J., Shigehisa,T., and Symons(1973) define sensory interaction as, " a stimulus in one modality being effective in altering sensitivity in another modality, either increasing or decreasing it". This doctrine has also been described by Gilbert (1941), as the facilitation or inhibition of the sense modalities under the effects of heteromodal stimulation. Marks (1978) in his treatise on the unity of the sense finds that this phenomenon is a contemporary formulation of Aristotle's 'sensus communis', a notion of a common sense that intergrates and regulates the activities of several senses. Marks also aligns this theory with those postulated by Locke, Galilieo and Kant. Their theories held that different senses provide the same information about the features of our world.

Research into intersensory effects or cross modal equivalence of the senses has been underway for over a century. Urbantschitsch (1888) attempted to investigate the effects of stimulation, in each of the sensory modalities, in order to study their effects on the other modalities. The findings of these experiments are looked upon with some reserve as the results were not very consistent and the controls were almost entirely lacking. Urbantschitsch did succeeded in calling attention to the fact that heteromodal stimulation may effect sensitivity, but he did not indicate the direction of the effect, whether it facilitated or inhibited sensitivity.

A comprehensive study was compiled by Gilbert (1941). This summarized the large body of experimental work that had been done on the unity of the senses up to that time. Gilbert limited the scope of his work to experiments that demonstrated the effects of heteromodal stimulation on sensitivity - the so called 'dynamogenic' effect of auxiliary stimulation. Dynamogenic is a term coined by Johnson (1920), and is used to designate instances where stimulation in one modality increases or decreases sensitivity or acuity in another. Gibert reports that experiments showed that subjects became less sensitive to auditory stimulus under the influence of an auxiliary stimulus. In these experiments electric shocks were used as a primary stimulus to measure the effects on auditory sensitivity under increased or decreased shock application to the subjects. The greater the shock the less sensitive the subject became to the auditory stimulus.

Later, Jacobson (1911) was able to show that sensitivity to the adequate stimulus is decreased, by simultaneous heteromodal stimulation. Jacobson's work focused on a subject's perception of sound when acted upon simultanoeusly by pressure i.e. weight sensation. He found that subjects judged weights as being heavier when they were accompanied by a given sound. And that a given sound was judged louder than another sound when a simultaneous pressure occured with the latter than when it did not. Jacobson concluded .... " We are permitted to infer that the conscious intensity of sound may be reduced by concomitant pressure sensations." Gilbert likewise concludes, that a sufficiently intense stimulus will momentarily reduce sensitivity in another modality and increase it after an optimum interval. Alternatively a less intense heteromodal stimulus will momentarly increase sensitivity.

4. VISION AND AUDITION

In the interaction of the visual and auditory senses, vision has been found to predominate. Marks (1978) explains this phenomenon by what he refers to as "law of minimum perceptual effort". Marks states this simply as: "people try to make sense out of what they sense, and make simple sense at that". This statement needs to be seen in the light of a person receiving two or more sets of stimuli from the environment at once. Under this effect of heteromodal stimulation the conflict between the two senses is resolved by the subjugation of one of the sensory modalities. Marks has found that vision generally predominates, waggishly stating," Apparently, seeing is believing"!

In the perception of space vision has been found to predominate the other modalities, besides touch. Spatial information is normally derived from the different senses and is coordinated so that the direction perceived by the visual and auditory senses appears to arrive from a single spatial representation. Witkins, Wapner and Leventhal (1952) looked at how far a sound had to be moved off the middle line so as person could tell it was off center. They found for a person to perceive that a sound had been displaced, the displacement had to be greater when the sound source appeared visually at the centre. The conclusion was that a person tended to hear sounds from where they were seen.

Pick, Warren and Hay (1969) also looked at sensory conflicts in the judgement of spatial direction. Their findings agree with the observations of Witkins et al. Auditory localization is greatly affected by the visual orientation of a stimulus. On the other hand, visual localization was not determined by the position of the auditory stimulus.

The results of our acoustic survey highlighted the importance of the patron being able to see the performance. People who had their view to the stage obstructed were also likely to find the theatre and the sound of the music as being unacceptable. For example, repondents who found their view to the stage obstructed were also found the seat they occupied uncomfortable, the room temperature as being too cold, and the brick wall of the theatre as being asesthetic only unsuitable for a music space.

Another section of the survey examined the lighting levels of the performance. Those respondents who felt the lighting was adequate, also found the seats they occupied comfortable, the sound of the music to be clear, bright and appealing. These results point to the strong influence of vision on the other sensory modalities. In particular we see that the respondents found it important to see as well as hear the concert. Those subjects who experienced difficulty in seeing also tended to have negative sensory experiences in the other modalities.

5. THE NEED TO SEE WHAT YOU HEAR

In the previous section we disscussed the dominance of vision over the other sense modalities. In particular we saw the need for people to see what they hear. Researchers have shown that this need is as much a psychological manifestation as an intrinsic physiological dictum. The work of Geissler (1915), demonstrates the psychological influence of vision on the perception of sound. Geissler experimented with auditory localization under various instructions - including the instruction to expect a sound in a certain half or quarter of a field. He found that subjects localized the sound best when it came from the expected quarter. However the error increased by 30 to 40% when the sound came from an unexpected quarter.

Subjects were also asked to outline what occured under the instruction to 'expect' the sound from a certian quarter. Most reported that much of the expectation involved visual imagining of the expected direction. Geissler concluded that the shift in localization under expectation may be partly explained as the result of.."fitting the sound into the visually imaged situation." The results of Geissler's experiment interestingly highlight the reliance of people on the visualization of auditory experience. This reliance factor is seen in the results of the present survey. The participants' responses clearly expressed the need to see the performance.

Ewert (1930) and Peterson.J, and Peterson.J.K,(1938) both note the fact that auditory localization tends to be directed by visual pattern. That is, sounds are heard as coming from the visually perceived source, even when with the visual field reversed. The work of Young (1928) demonstrates the physiological affinity of hearing with vision. Young looked at the role of visual perception on auditory localization by artifically reversing the cues for auditory perception. He found that visual perception of the sounding object could override the reversed auditory cues. Young concluded that auditory localizing was a product of two distinct activities. One was purely audiomotor, which was not influenced by practice in continually wearing the pseudophone (used to reverse the auditory cues). That is, without the use of vision, hearing remained reversed for the whole experiment. The other activity is localization involving the cooperation of vision. The interaction of this activity resulted in auditory perception coming back to normal.

It is interesting to note that the reliance between vision and sound is conventionalized in our language and culture. Such a relationship many commonly used metaphor and colloquialisms. For example "I am going to see a concert tonight", is not an uncommon expression.

6. PERCEPTION OF AUDITORY AND VISUAL INTERVAL

In order to better understand the the relationship between vision and sound we need to investigate further the effects of their unity. O'Connor and Hermelin (1972) showed that vision is the sensory modality which provides an indication of the number of different objects or events which constitute a stimulus. For example,when listening to an orchestra it is possible to distinguish the different instruments by their individual sounds, but not many people have this ability. However listening and seeing the orchestra simultaneously engenders a more vivid and acute perception of the individual instruments and their respective sounds. This point is stressed by the previously established notion that vision is an important part of sound localization and therefore it can be argued that vision makes the auditory experience more realistic.

This argument is taken up by Vernon(1934), who discusses the perception of music in terms of the listener who is 'with' or 'without' musical training or understanding. It is acknowledged that every individual listener appreciates music differently from every other, and that this varies with time and the nature of the musical event. However Vernon echoes Gurney's (1880) theory that musical perception is, "either indefinite or definite". Vernon sees the indefinite listener(who is usually a musically untrained adult) as not perceiving the sounds as something objective in the external world. Instead the sounds are translated into organic sensations or visual images. The visualization of the experience is to this listener just as important as the auditory one.

Definite listening is seen as being approximately equal to our ordinary perception of language. The musician, owing to his fimiliarity with 'grammar' and 'syntax' of the music can follow and anticipate the structural relations, interwoven themes and harmonic progressions. This is similar to following a verbal argument. In this case the listener's auditory sense can act independently of the other sense modalities. However Vernon indicates otherwise, ..."The abstract perception of notes as auditory objects is an exceptional state of affairs, a level reached only by the most highly trained musicians."

Vernon concludes by eluding to the following; ....

"musical perception depends on an extraordinary diversity of physiological and psychologicial reactions,..... different listeners employ a variety of different modes of response in their perception of the same features of the music"

7. UNIVERSAL SENSORY ATTRIBUTES

Vernon (1934) discusses one other important aspect of auditory perception. This is the theory of intersensory dependence or consonance. He states, ..." The really important features of auditory perception must depend upon central processes." The central processes advocated here are equivalent to Aristotles theory of 'common sensibilities', i.e. different sense provide the same information about the features of our world. This can then be interpreted as indicating that sensory information or sensations collected by the senses is part of a common sensory attribute(s). Under certain conditions the qualities perceived by one sensory system are influenced by the stimuli reaching other sense organs. It has already been noted that auxiliary stimuli may act to either facilitate or inhibit the stimulation of other sense modalities.

Of great importance in the theory of consonance is the question of what aspects of sensory perception are comparable from one modality to another. The aspect of brightness has received the greatest attention from this point of view. Von Hornbostel (1927) described brightness as being a quanitity that is common to all the sensory modalities. In his experiments he contrasted auditory brightness and pitch. As would be expected, his results showed that high pitched sounds are bright and that low pitch sounds are dark or dull. It was found that brightness can also be associated with timbre and loudness.

More recently, Marks(1974) experimented with pure tones and white light. He found that if pitch is held constant and visual brightness increased subjects would increase loudness to match the two sensations. Similarly, light intensity was increased to match a higher pitch. Marks concludes that the brightest tones are high pitched and loud, while the darkest are low pitched and soft.

At this point it is interesting to note that Beranek (1962) incorporates the ideas of universal sensory attributes into his famous work on the auditoria acoustics. For example when discussing the subject attributes of musical acoustic quality in a hall, one of the qualities he described is brightness. Brightness is defined as..."bright clear, ring sound, rich in harmonics. It comes from the relative prominence of the treble and slowness of its decay." Beranek's definition clearly coincides with Marks experimental conclusions.

Brightness is a fundamental quality, consonant with all sensory modalities and is one of three basic dimensional sensory experiences that dominate our sensory experience, according to Hartshorne's(1934) theory. His theory places specific equivalences amongst qualities of virtually all sensory modalities. It is postulated that this equivalence is a result of adaptive evolution. He attempts to account for this phenomenology through postulating three fundamental dimensions that are common to all sensory experience. These are ; activity - passivity, joy - sorrow and intensity - faintness.

Hartshorne' attempts to relate modalities such as vision and sound by matching various sensory nomenclatures to these dimensions. For example, he matches the following pairs of colours as being contrasting to the postulated common sensory attributes; red - green to active - passive, yellow - blue to joy - sorrow and white - black to intensity - faintness. Low pitched sounds are found to correspond to sorrow and hence associated to blue - violet and high pitched sounds to yellow through joy.

Simpson, Quinn and Ausubel(1956) and Wicker (1969) reach similar conclusions to those of Hartshorne, on the bipolar dimension of colours. Their work was conducted independently of his fundamental dimensions. Wilson (1965) found that subjects do describe red, as a more lively and exciting colour than green.

This relationship between colour and mood response emerged in the results of the present pyscho-acoustic survey. It was found that a patrons response to the colours inside the theatre also tended to be similar to their auditory reponse to the music and tactile response to their seat. Patrons recorded that their response to the music tended to be affected by such things as the colour of the seats and carpet. Another interesting alliance that emerged was the response to the brick walls in the theatre. Patrons generally found them dull and austere. This tended effect their perception of the music in a similar manner.

8. CONSONSANT DIMENSIONS OF VISUAL EXPERIENCE

To what extent do the consonant dimensions of visual experience translate into the dimensions of auditory experience; that is what are the parallels between the visual and and auditory dimensions of sensory experience? It has already been seen that brightness is one of these dimensions. Marks (1978) describes two other fundamental attributes of sense as being quality and intensity, stating... "that the visual manifestation of quality is colour and the auditory manifestation is pitch." Ryan (1940) reports found intensity in other modes is also related to colour in vision.

Zietz (1931), systematically investigated the interrelations between the visual and auditory systems. Colour quality was found to be influenced by the auditory stimuli only when colour was unstable, and low in saturation. For this purpose he he used afterimages, colour wheels of low saturation and brief tachistoscopic exposures. Zietz concluded that low tones tend to make a colour darker, warmer, unclear and dirty, shifting colours toward the red, blue or violet. While high tones usually became brighter, colder, sharpley contoured, more solid and surfacey. High tones changed the colours toward green and yellow.

Marks, (1978) also finds this analogous relationship between specific colours with specific musical pitches. The most extensive empirical literature which discusses these analogies amongst the senses deals with synesthesia. Synesthisa refers to the production of sensations in one sensory system that are a result of sensations in another sensory system. Experimental evidence has shown that for some people the stimulation of the auditory system with musical tones can give rise to the vivid perception of colours. The phenomenon of synesthesia has been commonly dubbed, "coloured hearing".

Eckerson, Karwoski and Odbert (1942), looked at the musical associations of colour and mood, focusing on coloured hearing in the context of perception. They tested the colour and mood responses of subjects when listening to different pieces of music, which had been selected to represent a fair range of moods. Strong associations were found between colour and mood. These were seen to be partially a result of the tradition of our language and culture. Another interpretation offered by Eckerson et al is that the music establishes a mental state which induces mood response. This response was found to be parallel to our colour associations, which was in turn dependent on our cultural traditions.

Eckerson et al, also found this psychological response to be induced singularly by colour. Subjects were shown various colours in the absence of the music and asked to rate their mood response verbally. These correlated significantly with the other results. The results of these experiments were used to construct a mood circle. This represents the systematic response of subjects to music in terms of colour and mood. The following colour - mood correlations are represented on the 'mood circle'; Red = Exciting, Orange = Gay, Yellow = Playful, Green = Leisurely, Blue = Tender, Purple = Solemn, and Black is heavily concentrated at Sad. Colours like blue and purple tend closely to black. Where as colour like red and orange tend to yellow. A high correlation is found between these results and those postulated by Hartshorne.

9. CONCLUSIONS AND DISCUSSION

That vision effects audition is not in doubt. This is not only on the basis of the mark cited but also from the work of composers such as Scriabin, Beethoven, and Mozart. Scriabin went so far as to use a 'light keyboard' at the premiere of his composition "Prometheus". Kaleidoscopic colours were displayed at the precise moment that certain parts of the music were being played. Beethoven called the key of B minor "the black key" while Mozart used the key of A major as "a pattern of many colours". The language of music also abounds with words canoting colour: "bright" and "dark" sounds, the "chromatic" scale. "tone colour" and coloratura" passages.

It seems then that vision has an important impact on the perception of music. This should have important implications for the design of concert halls. With an opera or dramatic presentation the set and costuming is all-important for the rendering of the correct ambience and the ability of the audience to see an action and distinguish between characters on stage. Presumably the colour of materials used in the interiors of concert halls is also important, as is the lighting (both the colour rendering and the level).

This paper has been written in order to raise the awareness of designers and consultants to the necessity of an integrated approach to the design of rooms for music. The paper has also been written as a review of the type of information which is available on bimodal perception. Very little of this work is reported in acoustics journalism and texts and yet it has obvious relevance to much acoustics design and research work and audiometric testing.

It would appear from the results of the survey in the Seymour Centre that there are significant influences on the perception of acaustical conditions by the comfort of the seats, the colour of the room interior and the adequacy of sight lines. What is apparent from the review of literature is that there is very little information available which can be used directly by designers and that the application of the work on bimodal and heteromodal stimulation requires further study. However any work on auditory perception which does not take into account non-auditory sensory inputs must be treated with caution.

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PSYCHOACOUSTIC FACTORS IN MUSICAL HARMONY

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PSYCHOACOUSTIC FACTORS IN MUSICAL HARMONY

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ABSTRACT

Musical judgements were examined in view of possible acoustic and sensory influences. First, listeners rated how well pitches in the chromatic scale followed a major triad, a major third, and a perfect fifth. Ratings reflected harmonics and subharmonic tone sensations. Psychoacoustic models of the ratings, differing in the number of factors considered, were formulated and tested. The correlation between predicted and actual ratings for the most successful model was .88 ($p < .01$). In Experiment 2, triads, thirds and fifths were paired with each other at various transpositions with respect to one another. Listeners rated how well the second element followed the first. Psychoacoustic models were tested, in which ratings corresponded to the sharing of octave-generalised harmonics and subharmonic tone sensations. The correlation between predicted and actual ratings for the most successful model was .81 ($p < .01$). In a third experiment, listeners judged tonal movement in sequences that changed key. Sequences were judged as conveying less tonal movement if key changes related well to the structure of the overtone series, suggesting that acoustic factors may influence the perception of broad levels of musical structure. Finally, short harmonic sequences were analysed by calculating acoustic relations between sequence notes and the overall key. Average relations varied from bar to bar in a wave-like pattern. The role of such patterns in the perceptual abstraction of keys is discussed.

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1. INTRODUCTION

The idea that music is tied to mathematical relationships has a long history. The Pythagorians associated musical intervals with small integer ratios corresponding to the lengths of strings. Rameau (1722/1971), along similar lines, explained how a chord maintains its essential identity whether it is in root position or inverted. As an example, a C major chord is associated with its root note C, whether the chord notes have the ascending pitch order, C-E-G, E-G-C, or G-C-E. According to Rameau, a major chord and its inversions are related in that each contains pitches that represent the frequency ratios 3:4:5. This observation remains important. Since these ratios occur in the harmonic series, a chord may implicate a root note ('do') in the same way that a harmonic spectrum is heard with the pitch of the fundamental frequency (even when the fundamental is absent). Sensory mechanisms yielding sub-harmonic tone sensations, implicated in pitch perception, might also underly the perception of chord roots (Terhardt, 1974; 1977).

On consonance and dissonance, Helmholtz (1877/1954) argued that dissonance may occur when the spectral components of complex-tone pairs produce beats. Plomp & Levelt (1965) showed that the most dissonant pure-tone intervals coincide with maximum beating between the two tones. For complex-tone pairs, the number of audible beating partials is smallest when fundamentals are related by a ratio of small integers (Terhardt, 1974; 1977). The intervals and chords used in Western tonal music tend to avoid beating partials. Thus, the concept of sensory consonance as the absence of beating partials can explain why small integer pitch ratios are found in harmonic music.

It is evident that sensory and acoustic factors are relevant to certain aspects of harmonic music. First, because pitch relationships found in major chords mirror frequency relationships found in harmonic spectra, mechanisms involved in pitch perception may explain how the chord-root is heard as the most important note in a chord. Second, sensitivity to overtones and their interactions may help to account for why certain tone combinations are consonant and others are not. Still not clear, however, is the musical relationship between chords and subsequent pitches that represent overtones and subharmonic tone sensations. Such information may be relevant to the expectations that a chord may evoke in a musical context. As well, research has not established whether acoustic and sensory factors are relevant to broader levels of musical structure, such as chord changes and chord sequences.

This paper briefly describes some investigations into the relationship between psychoacoustics and musical harmony. In one approach, a probe-tone technique was used to assess the musical importance of pitches representing harmonics and subharmonic tone sensations in chords and intervals. In the second approach, chords and intervals were paired with each other, and listeners were asked to rate how well the second element fit with the first. Ratings were analysed in view of the sharing of harmonic components and subharmonic tone sensations. In the third approach, musically untrained listeners rated the extent of tonal movement they perceived in short excerpts from Bach chorale music. Ratings were assessed in view of the notion that sensitivity to the overtone series may influence the tonal movement perceived in key changes. In the final approach, highly trained musicians composed short harmonic phrases. The changing acoustic information suggested by the sequences was examined to address the question of how harmonic phrases might convey an overall sense of key or key change.

2. STUDY I: RATING OF PROBE TONES FOLLOWING CHORDS & INTERVALS

This work examined judgements of three musical elements: a major triad; a perfect fifth; and a major third. First, the energy present at each pitch-class in the triad or interval was determined (i.e., fundamental frequencies and overtones). Next, sub-harmonic frequencies which may be perceived as a result of sensory mechanisms involved in pitch perception were assessed (e.g., Terhardt, 1974). To examine the influence of these factors, listeners were asked to rate probe tones following each of the musical elements. 12 probe tones were used: one for each pitch in the chromatic scale.

Tones were produced by a DMX-1000 real time digital synthesizer, controlled by a PDP 11/23 computer. On each trial, listeners were presented (through Sennheisser headphones) a triad or interval 350 msec in duration, followed by a 500 msec pause and then a probe tone of 1 sec duration. All tones had a rise and decay time of 22 msec each. Tones had five partials, with the amplitude of each partial inversely proportional to the partial number. To emphasise pitch-class, rather than pitch height, probe tones were constructed by combining nine octave equivalents of a single tone, and placing a Gaussian waveform amplitude envelope over the entire set. The order of trials (three musical elements X 12 probe tones) was scrambled for each subject. Each presentation was randomly transposed with the condition that the lowest tone in the triad or interval was between G3 and C4. 15 musically trained listeners rated how well each probe tone fit each musical element (from 1-7).

Harmonics and subharmonic tone sensations were predicted to have an octave-generalised influence. Thus, ratings were predicted to be high when the probe tone had the same pitch as a harmonic component, or a subharmonic, of the triad or interval. (Due to this assumption, and the construction of probe tones, influences by sub-octave tone sensations could not be assessed. Therefore, only sub-fifth influences were considered).

2.1 RESULTS & DISCUSSION

Ratings of each probe tone following the perfect fifth, major third, and major triad are given in Table 1. For clarity ratings are displayed as if the lowest tone (the root) of the musical element were always C. For the perfect fifth, the two fundamentals of the presentation (C and G in the figure) were rated higher than other tones, $F(1, 14) = 53.00$, $p < .001$. Second, of the remaining probe tones, the probe tone at D was rated significantly higher than others, $F(1, 14) = 8.29$, $p < .05$. This finding is predicted from the fact that D is the pitch corresponding to the third harmonic of the note G. Probe tones at E and B were not rated highly, even though they represent the fifth harmonic of C and G respectively. Therefore, it appears that listeners were not influenced by frequencies beyond the third or fourth harmonic. For the perfect fifth, ratings did not appear to reflect subharmonic tone sensations. For instance, the probe tone at F (the sub-fifth of C) was expected to be rated somewhat highly, but was not rated significantly higher than other probe tones.

TABLE I

AVERAGE RATINGS OF PROBE-TONES FOLLOWING THREE MUSICAL ELEMENTS

probe-tone
(where the lowest tone of each musical element is C)

	C	Db	D	Eb	E	F	Gb	G	Ab	A	Bb	B
fifth	6.3	2.8	4.7	3.9	4.1	3.9	2.3	6.1	3.9	4.1	3.9	3.5
third	6.3	3.3	3.6	3.5	5.3	3.8	3.1	5.0	3.3	4.9	3.1	3.6
triad	6.3	2.5	3.5	2.9	4.7	4.7	2.9	5.8	3.5	4.1	4.3	3.3

For the major third, the two fundamentals of the presentation (C and E in the table) were rated higher than all others, $F(1, 14) = 68.42$, $p < .001$. Second, of the remaining probe tones, the probe tone at G was rated the highest, $F(1, 14) = 8.50$, $p < .05$. This finding is predicted by the fact that G represents the third harmonic of the note C. Third, of the remaining probe tones, the probe tone at A was rated higher than others, $F(1, 14) = 10.31$, $p < .01$. This finding is predicted on the basis that it represents the sub-fifth of E. As for the perfect fifth, the sub-fifth of C (i.e., F) was not rated highly.

For the triad, the three fundamentals in the presentation (C, E and G in the table) were given significantly higher ratings than other tones, $F(1, 14) = 34.37$, $p < .001$. Second, of these tones, C and G were rated higher than E, $F(1, 14) = 20.40$, $p < .001$. The latter finding is predicted by the fact that the notes C and G are mutually supportive: G represents the third harmonic of C, and C represents the sub-fifth of G. Of the remaining tones, the sub-fifth of C (i.e., F) was given the highest average rating, $F(1, 14) = 18.77$, $p < .001$.

To summarise, probe tones representing fundamental frequencies were rated highly in all three conditions. In general, probe tones representing near overtones were rated highly. However, ratings did not reflect an influence by the fifth harmonic. In some cases, probe tones representing the sub-fifth pitch of a fundamental frequency were given high ratings, suggesting an influence by subharmonic tone sensations.

2.2 PREDICTING RATINGS FROM PSYCHOACOUSTIC MODELS

As a second analysis of psychoacoustic factors, some simple models of the ratings were formulated. To most fully understand the relative importance of fundamentals, overtones, and subharmonics, six models were tested. The models differed in the number of factors considered relevant to the ratings. In the first model, ratings were predicted to be high only for probe tones representing fundamental frequencies in the presentation. In model two, ratings were predicted to reflect the first three harmonics of tones. In the third model, ratings were predicted to reflect all five harmonics. Models four, five, and six considered the same influences as sets one, two, and three respectively, but in addition, involved a predicted influence by sub-fifth tone sensations.

The influence of each harmonic component under consideration was assumed to be proportional to the amplitude of that component. (Recall that tones had five partials with amplitudes inversely proportional to the partial number). The influence of subharmonic tone sensations on probe tone ratings was less obvious. An informal analysis of the ratings suggested that sub-fifth tones were approximately as important as third harmonics. Thus, as an estimate, frequency components were predicted to implicate sub-fifth pitches with a perceived importance equal to 1/3rd that of the pitch of the frequency component itself.

Table 2 gives the correlation between predicted and actual ratings for each of the six models. Coefficient values are highly significant (in all cases, $p < .01$), including those for predicted ratings based on fundamentals alone. Considering an influence by fundamental frequencies alone, the average correlation between predicted and actual ratings is 0.81. Predictions are improved when sub-fifth tones are considered (sets 4, 5 & 6), and when three harmonics are considered. Predictions are not improved by considering all five harmonics. The model yielding the highest average correlation between predicted and actual ratings assumes an influence by the first three harmonics, and by sub-fifth tones (mean $r = .88$).

TABLE II

CORRELATIONS BETWEEN PREDICTED & ACTUAL PROBE RATINGS: 6 MODELS

Number of harmonics used in model

musical element	(no subfifth) 1	3	5	(with subfifth) 1	3	5
fifth	.84	.86	.86	.82	.86	.87
third	.78	.82	.79	.83	.88	.86
triad	.80	.78	.76	.89	.89	.87
Average corr	.81	.82	.80	.85	.88	.87

That average correlation coefficients were as high as .88 strongly suggests that the perception of triads and intervals is for the most part predictable from psychoacoustic factors. It is surprising, however, that the model considering fundamental frequencies alone yielded such good predictions. Given the success of this simple model in predicting ratings, there was little room for improving predictions by considering overtones and subharmonic tones. This ceiling effect makes the importance of near overtones and subharmonic tone sensations somewhat difficult to establish. However, it does appear that no improvement in predictions were made by considering the fifth harmonic: at most, listeners were influenced by the third harmonic and sub-fifth tone sensations.

3. STUDY II: JUDGEMENTS OF CHORD PAIRS

In the second approach, judgements of pairs of musical elements were examined. Three musical elements were used: a major triad; a major third; and a perfect fifth. All nine possible pairs were presented. Presentations consisted of a triad or interval 350 msec in duration, followed by a 500 msec pause and then another triad or interval. Tones had five partials, as in Experiment 1.

For each of the nine pair types, the second musical element was presented at each of the 12 possible transpositions with respect to the first musical element (in random order). Both upward and downward transpositions were presented. The lowest note of the first musical element was one of Bb3, B3, C4, G4, Ab4, or A4. 15 highly trained listeners were asked to rate on a scale of 1 to 7 how well the second musical element fit the initial musical element. It was predicted that ratings would reflect the overlap of harmonic and subharmonic pitches in the two elements.

3.1 RESULTS & DISCUSSION

Average ratings for each pair type are shown in Table 3. Ratings are presented as if the root (i.e., 'do') of the first musical element were always C. Averaging across pair types, similarities were found between ratings of pairs of elements and theoretical relationships between musical keys. First, pairs having the same root were given higher ratings than those having a root movement to the second (D in table), fourth (F), fifth (G), and flattened seventh (Bb), $F(1, 14) = 90.30$, $p < .001$. Second, pairs having a root movement to the fourth or fifth were given higher ratings than pairs having a root movement to the second or flattened seventh, $F(1, 14) = 65.23$, $p < .001$. These differences parallel the relationships between keys as described by music theory, supporting the close tie between chord and key relationships. Psychoacoustic models can also explain the findings since chords having the root notes of highly related keys tend to share more harmonic and subharmonic pitches than chords having the root notes of less related keys.

3.2 PREDICTING RATINGS FROM PSYCHOACOUSTIC MODELS

Six models of the ratings were tested. The models did not consider order information, but took into account only the overlap of harmonic components and sub-fifth influences. Model 1 considered just fundamental frequencies of the two musical elements. Model 2 considered the first three partials of the two elements. Model 3 considered all five partials. Models 4, 5, and 6 considered the same partials as models 1, 2, and 3 respectively, but these models also assumed an influence by sub-fifth tone sensations, as described in Experiment 1.

Predicted ratings of pair types were generated as follows: psychoacoustic models of each musical element were taken from Experiment 1. For each pair type, the two sets of predicted ratings were correlated at each of 12 possible alignments, one for each of the 12 transposition conditions in the presentations. The resultant 12 correlation coefficients were taken as the predicted ratings for that pair type. These coefficients were then, in turn, compared to actual ratings via another correlational analysis.

TABLE III

RATINGS OF PAIRS OF MUSICAL ELEMENTS

Lowest tone of 2nd musical element
(where the lowest tone of the 1st element is C)

pair type	C	Db	D	Eb	E	F	Gb	G	Ab	A	Bb	B
triad-triad	6.4	4.1	4.5	4.9	4.1	6.1	3.7	5.9	5.0	4.3	4.0	3.3
triad-third	6.3	3.5	4.0	3.8	3.4	4.0	3.3	3.7	3.3	3.0	3.7	2.7
triad-fifth	5.7	3.3	3.7	5.1	3.1	5.1	2.9	5.0	3.2	4.0	3.9	2.5
third-triad	6.7	3.9	4.9	4.1	4.9	5.8	3.7	5.4	4.9	5.7	4.5	3.5
third-third	6.1	3.5	4.7	4.3	4.0	4.3	3.5	4.9	4.5	4.6	3.0	4.0
third-fifth	6.4	3.9	3.6	3.1	4.9	4.7	3.3	4.7	4.0	5.2	4.5	3.0
fifth-triad	6.3	4.2	4.1	4.5	3.9	5.3	3.9	5.7	4.3	4.6	4.2	3.5
fifth-third	6.3	4.2	4.5	4.4	3.1	5.0	3.4	5.2	4.0	3.3	4.3	3.3
fifth-fifth	6.1	3.4	4.9	4.5	4.1	6.1	4.1	5.8	3.9	4.1	4.5	3.9
mean rating	6.2	3.8	4.3	4.3	4.0	5.2	3.5	5.1	4.1	4.3	4.0	3.3

Table 4 displays correlations between predicted and actual ratings for each of the nine pair types and each of the six psychoacoustic models. As in Experiment 1, predictions are very good even when just the pitches of the fundamental frequencies in the musical elements are considered. Predictions are improved by assuming an influence of three harmonics, and by assuming an influence by sub-fifth tone sensations. However, predictions are not improved by assuming an influence by all five harmonics. As in Experiment 1, the model that best accounts for the data assumes an influence by three harmonics and sub-fifth tone sensations (mean $r = .81$, $p < .01$).

TABLE IV

CORRELATIONS BETWEEN PREDICTED & ACTUAL PAIR RATINGS:
6 MODELS

Number of harmonics used in model

	(no subfifth)			(with subfifth)		
pair type	1	3	5	1	3	5
triad-triad	.77	.86	.81	.88	.92	.89
triad-third	.66	.68	.61	.68	.71	.64
triad-fifth	.63	.65	.62	.65	.65	.64
third-triad	.80	.86	.81	.88	.92	.88
third-third	.63	.72	.66	.74	.77	.74
third-fifth	.77	.78	.80	.78	.77	.81
fifth-triad	.85	.88	.86	.89	.88	.88
fifth-third	.58	.66	.59	.68	.74	.69
fifth-fifth	.86	.91	.88	.92	.94	.93
Average corr	.73	.78	.74	.79	.81	.79

4. STUDY III: JUDGEMENTS OF KEY CHANGE IN HARMONIC SEQUENCES

In music theory, the relationship between keys is described by the cycle of fifths. Major keys separated by an interval of a fifth or a fourth are adjacent in the cycle. As an example, the key of A major is adjacent to the key of E major. Adjacent keys are more musically related to each other than nonadjacent keys. In the cycle of fifths model, the musical relation between two keys does not depend on which key is presented first. However, Rosen (1971) has suggested that listeners are sensitive to the overtone series, and each key implicates *one* of its neighbouring keys with the third harmonic of the key centre, or tonic ('do'). The third harmonic of the key centre projects the adjacent key in one direction only (called the clockwise direction). If so, the psychological structure of the cycle of fifths should be generally unbalanced. Key changes suggesting movement in the clockwise direction should convey less tonal movement, because these key changes are consistent with the structure of the overtone series.

As a preliminary test of the above hypothesis, 20 musically untrained listeners were asked to judge to extent of tonal movement perceived in short excerpts from Bach chorales. Two sequences stayed in the same key, while eight changed key. On the cycle of fifths, sequences changing key were one of four types: modulating to a key one step in the clockwise direction (e.g., from C to G); modulating one step in the counterclockwise direction (e.g., from C to F); modulating two steps in the clockwise direction (e.g., from C to D); modulating two steps in the counterclockwise direction (e.g., from C to Bb). The apparatus used and the construction of tones was the same as in Experiment 1. Sequences were presented in random order to each subject twice. Judgements were made on a scale of 1-7.

4.1 RESULTS & DISCUSSION

Sequences modulating in the counterclockwise direction conveyed more tonal movement than sequences modulating in the clockwise direction. For sequences modulating one step on the cycle, the mean rating was 3.75 when the key change was in the counterclockwise direction, but only 2.90 when the key change was in the clockwise direction, $F(1, 19) = 8.67, p < .01$. For sequences modulating two steps on the cycle, the mean rating was 5.06 when the key change was in the counterclockwise direction, but only 3.83 when the key change was in the clockwise direction, $F(1, 19) = 10.27, p < .005$. These findings support the notion that the psychological relation between musical keys may be influenced by a sensitivity to the overtone series.

5. STUDY IV: THE CHANGING ACOUSTIC INFORMATION IN HARMONY

The final approach, still at an early stage, addresses the question of how global structure (i.e., key or key change) is conveyed in harmonic phrases. Two main issues are of particular concern. First, in so far as keys are conveyed in a great variety of music, in single melodic lines as well as in harmonic music, it is unlikely that perceiving an overall key depends on the presence of particular tones or tone sequences. More likely, the apprehension of an overall key involves a perceptual averaging of musical information. Second, research suggests that a sense of key may act as a means by which chords and tones are understood (Krumhansl, 1983). But since chords and tones are the very elements that give rise to the sense of key, there must be a balance between movement towards and movement away from the key. If musical information were always unrelated to the overall key, the sense of key would be lost. If musical information were always highly related to the key, there would be nothing musically for a key-sense to achieve.

To address these concerns, the present study looked at changes in acoustic information, averaged over important rhythmic units, in short harmonic sequences. Five students of composition were asked to provide two harmonisations of a given melody. The sequences varied between five and seven bars. Each student was given a different melody line. Melodies were composed by a music professor at the Conservatorium of Music, Sydney.

5.1 RESULTS & DISCUSSION

First, the key structure of the sequences was determined according to traditional music theory. Four of the sequences contained changes in key, while six were nonmodulating. Second, for each sequence, a measure of the acoustic relation between each note of the sequence to the key or keys of the sequence was determined. The acoustic structure of tones used in the previous studies was used as a basic model from which theoretical acoustic relations could be determined (i.e., 5 partials with amplitudes -> 1/n). The acoustic structure of a key was derived from the notes of the key's tonic triad (e.g., for the key of C major, the notes C, E & G). The acoustic relation between a sequence note and a key note was taken as the sum of weights given to all *pitches* shared by the two notes. The weight of a shared pitch was taken as the least of the amplitudes of the two components involved (where fundamentals were assigned an equal arbitrary value). For instance, the pitch of C is shared by the notes C and F (i.e., it occurs at the fundamental of C and at the third harmonic of F). The weight given to this shared pitch would correspond to the least amplitude of the two components involved, or the amplitude of the third harmonic of F.

Average acoustic relations between sequence notes and key notes were calculated for each bar. Figure 1 displays, for a selection of the harmonic sequences, the changing acoustic relation between sequence notes and the overall key or keys. Sequence 1 did not change key, but the acoustic relation to the overall key is seen to weaken initially, and then strengthen at the phrase ending. Sequence 2, two bars longer, shows a similar weakening of the key at the beginning and strengthening of the key at the end. In addition, however, this sequence has a strengthened key relation in the middle of the phrase. For each of the sequences, the average acoustic connection of sequence notes to the overall key shows a wave-like pattern. This wave-like characteristic provides the necessary balance between movement towards and movement away from the key. It may also be the case that such acoustic patterns are musically important in giving a gestalt quality to harmonic phrases.

Other sequences shown in Figure 1 give the average acoustic relation to two keys: one for the initial key of the sequence and one for the final key of the sequence. While the acoustic relations to both keys tend to maintain the wave-like pattern seen with sequences 1 and 2, waves reflecting the second key are more extreme toward the end of the phrase, with a sharp strengthening of the second key in the last two bars. Possibly, when listeners are meant to perceive a change in key, it is important that the second key is introduced in waves more extreme than those characterising the current key. Future work should examine the acoustic structure of numerous other harmonic phrases in order to assess the generality of these observations. If wave-like acoustic information characterises harmonic phrases in general, such knowledge may be useful to those interested in the perceptual abstraction of global structure from temporally organised events.

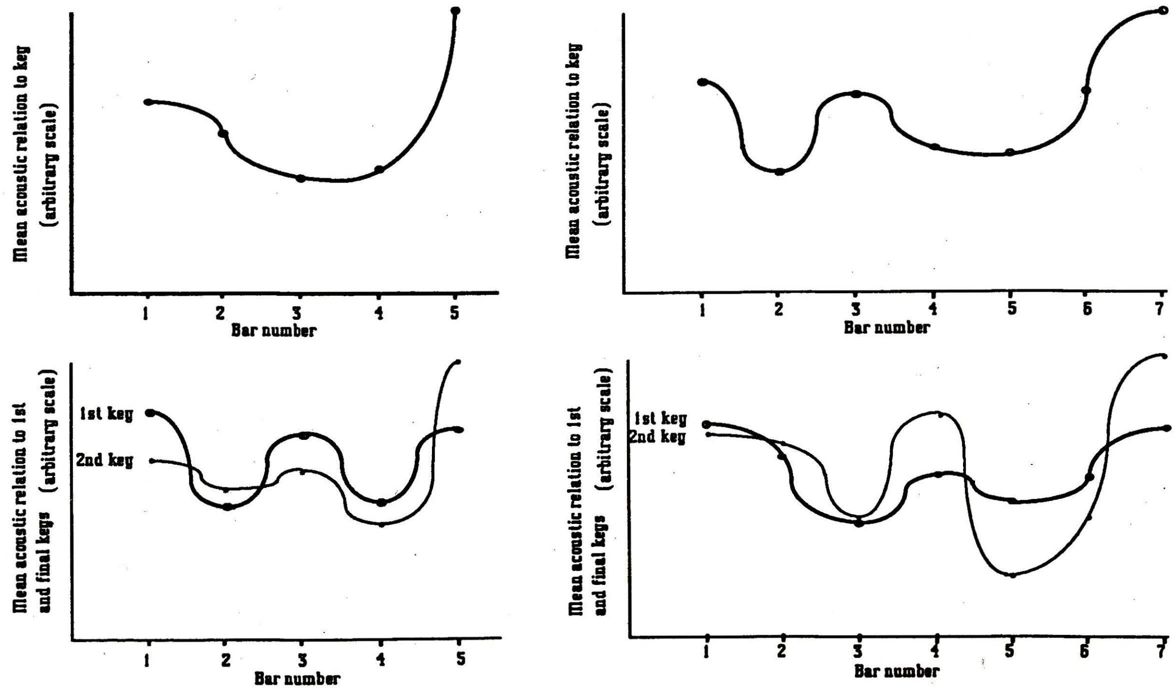

Figure 1. Average acoustic relations between sequence notes and the overall key or keys, plotted for four harmonic sequences.

6. CONCLUDING REMARKS

The above investigations, while but briefly reviewed, suggest an important role of acoustic and sensory factors in the perception of chords, chord changes, and harmonic sequences. It is notable, however, that different kinds of music could easily have arisen from such factors (Helmholtz, 1877/1954). Thus, to draw a strong connection between music and psychoacoustics does not suggest that styles of composition outside of Western tonal music are not musical. However, it may be useful, for those exploring new avenues of musical composition, to be aware of the potential for sensory and psychological processes to yield a link between acoustic structure and the enjoyment of music. Clearly, the structure of sound and our perceptual capabilities should be viewed as significant components of music listening.

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Acoustics in the Eighties

ENGINEERING CONSIDERATIONS IN ESTABLISHING A MANUFACTURING INDUSTRY FOR BOWED STRINGED MUSICAL INSTRUMENTS

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ENGINEERING CONSIDERATIONS IN ESTABLISHING A MANUFACTURING INDUSTRY FOR BOWED STRINGED MUSICAL INSTRUMENTS.

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ABSTRACT

The University of Tasmania's Conservatorium of Music and Department of Mechanical Engineering have embarked on a project aimed at manufacturing the violin family of musical instruments using indigenous timbers. Acceptable quality instruments have been produced by carefully matching the timber properties to the acoustic performance of the instruments. It remains to be seen whether the industry can be economically established.

ACKNOWLEDGEMENT: Financial support for this project has been provided by the Tasmanian Arts Advisory Board, the Tasmanian Development Authority and the University of Tasmania. The Authors wish to acknowledge the assistance of Mr Jan Sedivka, Master Musician in Residence at the University of Tasmania.

1. INTRODUCTION

In 1983 the University obtained a grant from the Tasmanian government to undertake a research project aimed at producing musical instruments, violins in particular, from locally available timbers. Local timbers were chosen because of their low cost and ready availability compared with the European timbers, spruce and maple, traditionally used for violin making; also there had already been some encouraging results with the use of Tasmanian timbers in guitars, pianos and violins. It was hoped that, ultimately, an industry could be developed to supply good quality instruments to the upper secondary school market in competition with instruments presently imported from Japan and China.

The major part of the grant was used to employ a young violin maker, Douglas Finlay, who with no formal training had come to the notice of the violinist and (at that time) Head of the Tasmanian Conservatorium of Music, Mr Jan Sedivka. Douglas had already made a playable violin from traditional timbers and was not bound to the conventions dictating that violins should be made from spruce and maple.

Instrument making is a complex mix of art and science. As engineers the authors leaned towards directing the research along established principles of acoustics. The crux of the problem was to make subtle changes in the shape of the violin to compensate for the differences in the properties of the local and traditional timbers. In this regard the project relied heavily on the work of Dr Carleen Hutchins and the Catgut Acoustical Society. (Hutchins 1981). There were other publications covering the art of violin making, in particular a book entitled "The Secrets of Stradivari" (Sacconi 1979) and others. The publications were some help, but in the final analysis the success of the project was largely due to the skill of the violin maker in being able to control the arching of the front and back surfaces (plates) and make sub-millimeter adjustments to the thickness of the plates.

The assessment of the finished violins was entirely art. Mr Jan Sedivka played the instrument and gave his opinion. This method of assessment, although not quantitative in any way, was perhaps the most appropriate in that the measure of any musical instrument is the price which it fetches on the market, and Mr Sedivka was able to value the instruments and give his opinion of playability.

2. METHODOLOGY

The project can be considered in three broad sections;

(a) feasibility of making instruments from local timbers,

(b) production techniques and,

(c) Material properties and their consistency and implication for quality control.

2.1 FEASIBILITY

The selection of suitable local timbers was on the basis of their physical properties; strength, weight and damping characteristics. Colour and appearance were considered to be of secondary importance. King William pine has been used exclusively as a substitute for spruce for the bellies, while Tasmanian blackwood, eucalypt, sassafrass and myrtle were tried in place of maple for the back, sides and neck/scroll cases. (Doe 1983). It quickly became apparent that King William pine, although a very resonant timber was much weaker than spruce in the direction across the grain; to allow for this the arching of the belly was increased so as to make up in shell strength what the timber lacked in elasticity in that direction. The selection for the back was more difficult. Here appearance and weight were more important than strength and resonance.

The front and back plates of a violin are made by carving a convex shape out of solid timber so as to set the desired arching. The inside of the plate is then carved to give the desired plate thickness in the range 3-5mm.

The traditional method of determining the "correct" thickness is for the violin maker to listen to the sounds (tap tones) from the plate as he raps the plate at different points. The maker also can twist and bend the plate to get a feel for its stiffness. These tap tones correspond to the resonant frequencies (eigenvalues) of different vibration modes (eigenvectors) of the plates. In particular the second resonant frequency (the cross mode) and the fifth resonance frequency (the ring mode) are particularly important in the "tuning" of the plates. Figures 1 and 2 show the second and fifth modes of a King William pine violin belly using the chladni technique of excitation by a loud speaker and mode visualisation with decorative 'glitter'. By careful adjustment of plate thickness the maker can set the tap tones to within a couple of hertz. Tone quality and response of the finished violin is known to depend on the relationship between the resonances of the back and belly.

Figure 1
Chladni pattern - 2nd mode.

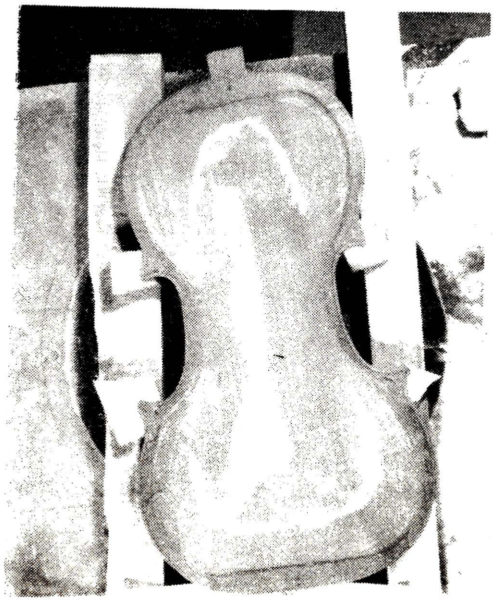

Figure 2
Chladni pattern - 5th mode.

2.2 PRODUCTION TECHNIQUES

The traditional craft method of making violins is very labour intensive. Shaping a plate or carving a scroll case can only be undertaken by skilled craftsmen. The labour input is reflected in the cost of hand made instruments which range upwards from around $1,500. From the outset, the aim of the project was to produce medium quality instruments at a competitive price. With a market aimed at school children at present supplied with factory made, imported instruments, much preliminary shaping by machine will be needed to minimise the cost. Preliminary work has been done on the routing of plates and scroll cases on the three axis numerically controlled milling machine at the Mechanical Engineering Department of the University of Tasmania using CAD-CAM techniques. Skilled craftsmen will still be needed for the final tuning of the plates and assembling the violins, but the roughing out of plates, scroll cases and other ancillary components on a NC machine will reduce the labour component substantially.

2.3 MATERIAL PROPERTIES

Currently in progress at the University of Tasmania is a project involved with determining the dynamic properties of native Tasmanian timbers. While the major thrust of this investigation is not aimed at the musical instrument study, its results are directly applicable. The properties of interest are the dynamic moduli, damping and the phase relations in the Poisson's ratios. Knowledge of these material properties will serve two purposes: firstly, since the intention is to use materials different from those traditionally used in violin manufacture there is a need to have a good idea of the differences between the properties of the different timbers so that these differences can be accommodated by changes in the arching and the thickness of the violin plates. Secondly, timber is a highly variable material - there is a need to know how to inspect a piece of timber so as to assess its suitability.

Progress on the measurement of the dynamic properties of timber has been slow and few results have been obtained. Some static tests have been conducted on small strips of timber both untreated and treated with a sealer. Two different samples of spruce were tested and compared with two King William pine samples. One of these pieces was visually selected by the violin maker (Douglas Finlay) and used for the bellies of several violins. The other piece was rejected by the violin maker. Both King William pine samples had considerably lower stiffness than spruce in both the axial and radial directions. The "selected" piece of King William pine was highly variable with very poor radial stiffness. The addition of the sealer had a marked effect in increasing the stiffness of the King William pine but very little effect on the spruce. The sealer reduced the variability in the pine. Sacconi (1979) and Caldersmith (1981) have mentioned the stiffening effect of a sealer in quality violins and suggested that Stradivarius may have deliberately used this trick. From the marked stiffening effect that the sealer has on King William pine it appears that a stiffening sealer may be essential for bellies made from this material. This is consistent with the observations made by Douglas Finlay in building his instruments.

3. RESULTS

Over the period from early 1983 to mid 1984 Douglas Finlay made eighteen violins, twelve from indigenous timbers. The early instruments experimented with different timbers (blackwood, sassafras, eucalypt and myrtle) for the back. The common criticism was that the instruments were not sufficiently bright - they showed a tonal quality similar to the viola. In the later instruments Douglas was able to set the arching and tune the plates using tap tone and chladni techniques to give clear modal shapes close to the recommended frequencies.

These later instruments were all from King William Pine and Tasmanian myrtle and were assessed by Mr Sedivka as being of professional quality. One of the instruments was played for a while in the Tasmanian Symphony Orchestra.

In mid 1984 the project came to the attention on a visiting violin maker and repairer who was impressed by the tonal qualities of the violins but critical of the finish of the violins and the departure from traditional timbers. This led to Douglas Finlay obtaining a Queen Elizabeth II Award and admission to the Newark Violin Making School for three years of formal training in violin making and repairing.

4. CONCLUSIONS

The project has established that violins can be successfully made from timbers other than those traditionally used for that purpose. Using non traditional timbers requires subtle changes to the arching and thickness of the front and back plates so as to compensate for the different timber properties. The project still has a long way to go before the aim of commercial production can be realised. A market survey needs to be conducted and customer acceptance assessed; some violin players are very biased towards traditionally made instruments. One of the problems is predicting how the new instruments will perform in five, twenty or even one hundred years. We are not aware of any method of answering this question. However we are confident of being able to make new instruments with playing qualities which compare favourably with traditional instruments costing $1,500 or more.

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Acoustics in the Eighties

Session 2A: Hearing

Acoustics in the Eighties

SOME FIELD OBSERVATIONS OF HEARING LOSS, HEART RATE AND BLOOD PRESSURE IN GUN CREW FIRING 105 mm HOWITZERS

N.L. Carter

SOME FIELD OBSERVATIONS OF HEARING LOSS, HEART RATE AND BLOOD PRESSURE IN GUN CREW FIRING 105 mm HOWITZERS

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ABSTRACT

Hearing levels, blood pressure and heart rate were measured in two gun crews before and after firing Hamel 105 mm howitzers. Each gun fired a total of 40 rounds during each of two test days, the guns often firing simultaneously. Two types of ammunition were used, that with the higher (noisier) charge being fired on the second day. All gun crew were fitted with E.A.R. earplug, and noise measurements were made at each of the crew positions. The results indicate no effect on hearing but possible effects on heart rate and systolic blood pressure.

1. INTRODUCTION

The L118/119 British Light Gun (Hamel gun) is due to come into service in the Australian Army in 1989. The results of measurements of the blast overpressures (BOPs) at Hamel gun crew positions have caused concern because of possible effects on hearing and other aspects of the physical health of gun crew, and NAL was requested to join the Directorate of Artillery and the Directorate of Army Health Services in a field study of possible effects on guncrew of firing the gun. This paper describes the effects of blast overpressure (BOP) on the hearing of guncrew, and presents some observations of heart rate and blood pressure of one gun crew before, during and after firing.

2. THEORETICAL BACKGROUND

2.1 EFFECTS ON HEARING

Evaluation of the hazard to hearing presented by firing the Hamel crew positions alone is not possible at present for the following reasons:

(i) The lack of a generally accepted Damage Risk Criterion (DRC) for hearing applicable to impulse noise.

(ii) Lack of an agreed methodology for estimating the amount of protection provided by hearing protectors against various types and levels of impulse noise.

(iii) Paucity of data on the relation of rate of fire to hearing hazard.

(iv) Lack of data on the effect of noise from neighbouring guns, such as may occur in training.

2.1.1 Damage Risk Criteria (DRC) For Hearing Loss from Impulse Noise

Current DRC differ according to the maximum level allowed without the use of ear protection, the method of measurement of impulse duration and the adjustment of permitted BOP according to the number of rounds fired per day. One standard (CHABA) describes the use of either of two methods of measuring impulse duration depending on whether the waveform (pressure by time) consists of a single peak or many, as may result from reflections of the primary wave from nearby surfaces. In practice noise from weapons may reasonably be described both ways and the resultant calculations of hazard may not agree.

In addition recent studies have suggested that the above criteria, based primarily on studies of the effects of noise from small arms on hearing, may be too conservative when applied to noise from large weapons. For example, Price (1982; 1986) has contended that the degree of hazard to hearing presented by impulse noise depends primarily on the location of the spectral peak of the impulse and its relation to a curve expressing the sensitivity of the ear to damage as a function of frequency (a critical level or CL curve), and has supported this contention by experimental work on animals.

2.1.2 Hearing Protection
Current DRC all include estimates of the amount of protection afforded by hearing protectors but these estimates vary greatly. Thus the British Standard (cf Forest, 1982) and that of the Australian Ordnance Council (1984) assume that hearing protectors provide the equivalent of 20 dB reduction in peak level of each impulse. The US MIL STD 1474B generally assumes that 29 dB of protection is provided by single hearing protectors such as V51-R earplugs and that a further 6.5 dB noise attenuation is possible from the use of double ear protection (plugs and muffs). German estimates depend on the type of protector and vary from 25 to 35 dB. These estimates are based in part on empirical studies (eg. Garinther and Hodge, 1971) but there is as yet no agreed method for measuring the attenuation of impulse noise by hearing protectors comparable to those available for steady state noise.

2.1.3 Rate of Fire
No current DRC for impulse noise deals with the effect of rate of fire, i.e. 10 rounds spread evenly over one day is regarded as equally as hazardous as 10 rounds fired in as many minutes.

2.1.4 Effect of Firing Two Guns Concurrently
In some circumstances two guns may be placed such that there may be occasional summation or cancellation of overpressures due to simultaneous or near simultaneous firing of the two guns; and/or the firing of the more distant gun may effectively increase the 'B' duration of the noise from the nearer gun. Overall evaluation of these effects on hearing is aided by a field trial although detailed evaluation of each factor is not of course possible under these conditions.

2.2 HEART RATE AND BLOOD PRESSURE

There is plenty of evidence for physiological responses to sound and noise, which involve reflexive and autonomic system responses, but very little of this information, other than that on the startle response, is concerned with responses to impulsive noise. Measures of blood pressure (BP) and heart rate (HR) were chosen for the following reasons:

(i) The literature on the non-auditory effects of industrial noise on people suggests that the cardiovascular system may be affected by noise.

(ii) Laboratory studies of monkeys have found permanent noise affects on blood pressure without concomitant effects on hearing.

(iii) Experimental studies with human subjects have demonstrated peripheral vasoconstriction in response to sounds as low as 70 dB.

(iv) A previous study by the author found consistent increases in heart rate in one gunner firing an M198 155 mm howitzer.

(v) There is a 'plausible mechanism' for short term effects of noise on heart rate and blood pressure which may contribute to chronic increases in blood pressure in the long term.

2.2.1 The Literature on Non-auditory Health Effects of Noise
In 1979 Welch reviewed translations of European research reports and found that all of forty studies dealing with the cardiovascular system found a deleterious effect "of one type or another". Subsequent reviews have concluded that these results could not be accepted at face value for methodological reasons. Nevertheless, no study found a beneficial effect of noise, as should have occurred if only random factors were operating.

2.2.2 Laboratory Studies of Animals
Numerous studies of the physiological responses of animals to noise have been carried out. We were influenced principally by the studies by Peterson et. al. (1981; 1984) on the effects of noise on heart rate and blood pressure in monkeys mainly because:

(a) these animals are more similar to humans than other species whose responses to noise have been studied;

(b) the studies were longitudinal, spanning 6-9 months of daily noise exposure;

(c) they included measures of hearing.

Peterson in et. al. found persistent and probably permanent increases blood pressure of up to 28% without any effects on hearing.

2.2.3 Peripheral Vasoconstriction
A number of studies showing peripheral vasoconstrictive (plethysmographic) responses at the finger tips to noise at levels as low as 70 dB have been reviewed by Burns, (1979). Constriction of the temporal artery elicited by steady state sounds over 95 dB does not appear to habituate to successive occurrences of the sound (Sokolov, 1963; Carroll, 1974).

3. INSTRUMENTATION AND METHOD

3.1 FIRING AND TESTING SCHEDULE

Firing was carried out on two test days with a lower charge being used on Day 1. The average interval between firings was about one hour with up to 11 rounds being fired on each occasion from each gun.

3.2 SUBJECTS

The subjects were 16 normal healthy volunteers. A maximum hearing loss of 25 dB was permitted in either ear up to 8000 Hz.

3.3 AUDIOMETRIC TESTING

The subjects were each given several tests before exposure to noise.

The procedure after each firing was for gun crew B to walk immediately across to the trucks for the hearing tests. Gun crew A remained at the gun for blood pressure testing, after which they too, removed ear plugs and walked across to the trucks for the hearing tests.

All troops were instructed and closedly supervised in the technique of fitting the E.A.R. earplugs.

3.4 HEART RATE MONITORING

Heart rate monitoring was carried out using Holter monitors. Heart rate and arrhythmia data were obtained from the (Mediloggers') tapes after the trial. The 'clocks' of the Mediloggers, the ambulatory blood pressure monitors, and the audiometric testers' wrist watches were all synchronised with the time/date generator of the video recording system (see below).

3.5 BLOOD PRESSURE (BP) MONITORING

The blood pressures of six men from the crew of gun A were measured when possible before and after firing. Blood pressure measurements were also carried out at various other times to provide an estimate of the typical BP values and their variation under trial conditions. Tests were also made before and after a series of simulated firings on Day 2 to estimate the effect of the physical activity involved in loading and firing the gun alone, i.e. without noise and blast. 18 BP readings on each subject were obtained on Day 1. 37 BP readings were made on each subject on Day 2.

3.6 THE VIDEO MONITORING SYSTEM

A video camera was mounted on a mast on top of a small van, focussed on the gun positions, and its output recorded on two 4-hour video recorders controlled to record sequentially. Time of day (in hours, minutes, seconds and hundredths of a second) and the date were recorded on the tape using a time/date generator.

3.7 NOISE MEASUREMENTS

Gun noise recordings and measurements at each of the gun crew positions were made by means of microphones mounted on vertical rods driven into the ground. The microphones were connected to charge amplifiers, sound level meters and a spectrum analyser. These measurements were carried out by the Advanced Acoustic Technology (AAT) section of NAL.

4. RESULTS

4.1 TEMPORARY THRESHOLD SHIFT (TTS) DATA

Temporary hearing losses (TTSs) were calculated for each frequency and ear by subtracting reference hearing levels established at the beginning of each test day from corresponding hearing levels measured after each firing. Histograms of the resulting TTs measures were drawn up but because little TTS was evident all TTS measures at each frequency were pooled for each test day. Figure 1 and 2 plots the means, standard deviations and >2 SD's of TTS from all firings for both test days, and the last firing of each day.

As can be seen from this figure there is no evidence for substantial hearing losses in protected ears due to the gunfire. Indeed there is a slight mean 'improvement' in hearing level during each test day, probably due to a training effect in these subjects. These comments must be qualified because of the small numbers of subjects used. The TTS data summarized in Figures 1 and 2 do not include losses sustained by subject B6, withdrawn after the first firing of Day 2, because he tried to refit the plugs during firing. His TTS data peaked at over 50 dB and demonstrate what can happen if errors in earplug fitting occur at crucial times.

4.2 HEART RATE

Following the trial, full paper printouts of the ECG's were obtained. Precise times of firing the guns were obtained from the video recording and marked on the pages of the ECG printouts.

Mean heart rate in beats per minute (b.p.m) was calculated for each subject for each 10-second interval, commencing two minutes before the first shot of each group of shots and ending two minutes after the last shot of the group. The averages, across subjects, of these 10-second mean heart rates were calculated. Mean heart rates during the simulated firing on Day 2 were also calculated.

Inspection of data suggested that firing the guns induces an increased heart rate, but the firing of gun B 7 metres away is not clearly associated with increased heart rate in the crew of gun A. Simulated firing, in which the physical activity of lifting and moving ammunition was the same as in normal operation of the guns, was not associated with an increase in heart rate. These impressions heart rate was again averaged over 20-second intervals before and after each shot and compared with the heart rate averaged over all 10-second intervals not within 20-seconds of any firing of gun A (simulated firing excluded). The respective means (HR) and standard deviations (HR) were 98.7 b.p.m. and 11.1, and 86.4 and 7.8 respectively, for Day 2. For

Day 1, mean of mean heart rates within 20 seconds of firing was 83.4 b.p.m., standard deviation 7.6 b.p.m. Similar data for intervals not within 20 seconds of any firing of gun A were 78.6 and 5.3. For Day 1 the greatest difference between mean heart rates was between intervals less and greater than 10 seconds from firing. Means and standard deviations were 86 (7.4) and 78.6 (5.3) respectively.

4.3 BLOOD PRESSURE

Systolic, diastolic and mean blood pressures measured 10 minutes or less before firing were subtracted from corresponding measurements made up to five minutes after firing.

For comparison, differences were also calculated between all successive pairs of the blood pressure measurements not used in the previous calculations - i.e. without intervening gunfire.

Blood pressures were also measured before and after a simulated group of firings on Day 2. In these simulations each round was passed to a member of the crew of gun B, instead of loading the gun, otherwise the drill was the same as for firing.

The six mean differences associated with gunfire were all positive, but those for Day 1 were small and were exceeded by the three mean differences not associated with gunfire. However, Day 2 systolic blood pressures before and after had a mean difference of 14.5 mm Hg, compared with -2.9 mm Hg for those not associated with gunfire and a mean difference of -4.25 mm Hg (SBP) associated with the group of simulated firings. Mean diastolic and mean, mean blood pressures associated with firing also yielded small positive values as compared with small negative ones from tests not associated with gunfire.

5. DISCUSSION AND CONCLUSIONS

5.1 EFFECTS ON HEARING

It is evident that the average effect of firing the Hamel gun on hearing is negligible provided E.A.R. earplugs are correctly fitted and worn whenever firing takes place.

Properly fitted E.A.R. earplugs provide adequate protection when firing the Hamel 105 mm howitzer for up to 40 rounds of ammunitions per day, when no more than 10 rounds are fired consecutively, followed by a pause of about one hour or more before the next firing.

The losses sustained by Subject B6 strongly indicate that a more reliable method of hearing protection, the use of which can be more easily monitored by supervisors, is required. At present the only reasonable alternative to earplugs appears to be earmuffs which incorporate small amplifiers and speakers to permit speech communication and sound localisation, while clipping high intensity sounds.

5.2 OBSERVATIONS OF HEART RATE AND BLOOD PRESSURE

5.2.1 Heart Rate

The application of statistical tests to these data to test for noise effects was not considered appropriate because of the small number of subjects and the many uncontrolled factors other than noise which could have affected heart rate. However, the following tentative conclusions seem justified:

(i) There is a reliable association between heart rate and firing Gun A, but not Gun B, in Gun A crewmen.

(ii) The peak heart rates associated with gunfire (averaged over 10-second intervals) are similar to those reported previously (Carter, 1986) where one subject fired single shots from an M198 howitzer separated by relatively long intervals of low physical activity. The 'baseline' mean heart rates were considerably higher in the present study, due to the physical activity required in multiple firing and other factors. The similarity of peak mean HR in the two studies suggests that the noise does have a reliable effect on heart rate which is partially 'masked' by changes in sympathetic tone due to these other factors (cf. Burns 1979).

6.2.2 Blood Pressure

No firm conclusions can be drawn, because of the small number of subjects and because adequate experimental controls were not feasible in the circumstances. Nevertheless the results do suggest that there may be an association between the peak sound pressure level of impulse noise and systolic blood pressure, an association which may have been clearer had it been possible to test blood pressure during firing.

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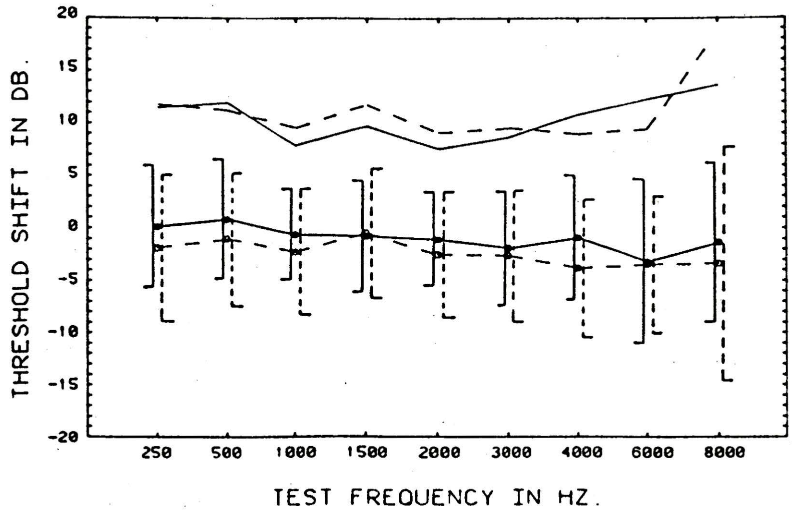

FIGURE 1 Means (filled circles), standard deviations (solid vertical bars), and 95% confidence limits (solid horizontal line) of all TTS measured on Day 1. Open circles, dashed vertical bars and dashed horizontal line represent corresponding TTS data from Day 2.

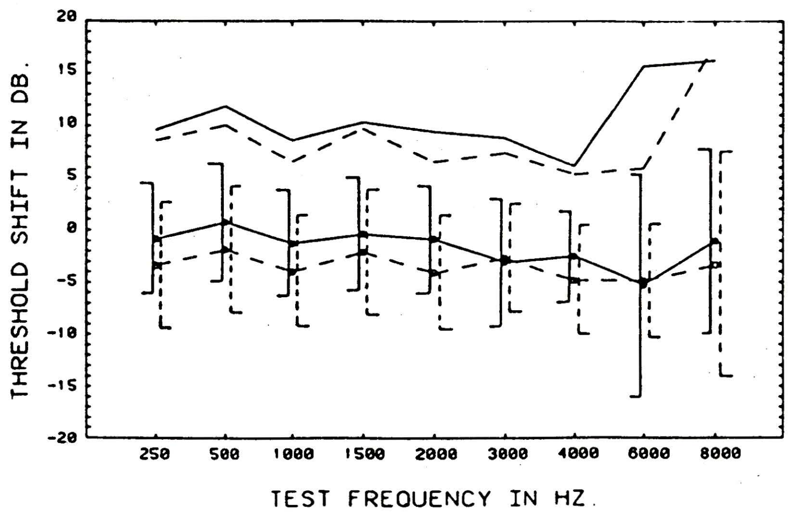

Figure 2 Means (filled circles), standard deviations (solid vertical bars) and 95% confidence limits (solid horizontal line) of TTS following the last firing of Day 1. Open circles, dashed vertical bars and dashed horizontal line represent the corresponding TTS data from Day 2.

Acoustics in the Eighties

NON-AUDITORY AIDS FOR THE HEARING IMPAIRED: MODALITY-SPECIFIC DIFFERENCES IN THE PROCESSING OF SPATIALLY AND TEMPORALLY PRESENTED INFORMATION

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NON-AUDITORY AIDS FOR THE HEARING IMPAIRED: MODALITY-SPECIFIC DIFFERENCES IN THE PROCESSING OF SPATIALLY AND TEMPORALLY PRESENTED INFORMATION

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ABSTRACT

To assess whether tactile displays have any inherent advantages over visual displays as non-auditory aids for the hearing impaired, the relative efficiency with which spatially and temporally presented information is processed by touch and vision was investigated. Subjects identified which of two vibrations or two luminous bars was most intense. Within each modality spatial, temporal, and spatiotemporal presentation methods were used. The tactile-temporal and tactile-spatiotemporal tasks were performed more quickly and more accurately than the tactile-spatial task, while the visual-spatial and visual-spatiotemporal tasks were performed more efficiently than the visual-temporal task. This suggests that touch processes temporally presented information more efficiently than spatially presented information, while the reverse is true in the case of vision. As speech is primarily characterized by acoustic changes over time, these results favour a tactile-based strategy for the provision of non-auditory aids for the deaf.

1. INTRODUCTION

Over the last decade there has been a resurgence in the development of tactile aids for the hearing impaired. Commercially available single and two-channel devices have been shown to provide significant benefits in the areas of lip-reading accuracy, speech production, and awareness of environmental sounds (Plant, 1983,1986; Spens and Plant, 1983). Even more promising are multi-channel devices such as the Queen's University Vocoder with which subjects have acquired closed set vocabularies of several hundred words (Brooks and Frost, 1983; Brooks et al., 1985,1986a, 1986b). In contrast, the development of visual aids has stagnated since the pioneering work carried out at the Bell Laboratories during the 1940's (Potter, 1945). This paper addresses the question of what it is that might make tactile aids superior to visual aids for the hearing impaired.

A sensible approach to the development of non-auditory aids for the hearing impaired is to concentrate on a modality whose processing abilities are compatible with the demands imposed by the nature of the speech signal. Perhaps the most fundamental property of speech is that it is characterised by changes which occur across time rather than across space. In this case, the capacity of vision and touch to process temporally presented information is central to an assessment of their suitability as modalities for non-auditory aids.

Several studies have investigated the processing of spatially and temporally presented information in hearing and vision. Metcalfe, Glavanov and Murdock (1981) presented subjects with either auditory or visual lists of words in a spatially and temporally distinct order, then asked the subjects to recall either the spatial or temporal order in which the lists were presented. Recall of temporal order was best in the auditory condition, while recall of spatial order was best in the visual condition. Thus it seems that temporally presented information is processed and/or retrieved more efficiently than spatially presented information when the input modality is audition, while the reverse is true in the case of vision.

O'Connor and Hermelin (1972) presented subjects with sets of three auditory or visual stimuli in which the temporal and spatial order of presentation were manipulated so that the middle stimulus in temporal terms never corresponded with the spatially central stimulus. When asked to recall the middle stimulus, subjects were more likely to identify the temporal median in the auditory condition and the spatial median in the visual condition. They concluded that auditory presentation encourages temporal processing, while visual presentation promotes spatial processing. This finding is consistent with the minimal processing principle (Thorson, Hochhaus, and Stanners, 1976), which states that when given the choice between coding formats a modality will adopt the more efficient alternative.

These two studies suggest that audition specializes in processing temporal changes, while vision processes spatial changes most efficiently. While these results are not unexpected when the nature of the stimuli normally processed by each modality are considered, they do suggest that vision may not be a suitable modality through which to present temporally structured information such as speech.

Similar data is not available for the sense of touch. Although several studies have suggested that similarities exist between auditory and tactile coding processes (Békésy, 1955, 1957a, 1957b, 1959; Handel and Buffardi, 1968), it is not clear whether these similarities extend to areas relevant to the communication of speech. The aim of this paper is to determine whether a comparison between touch and vision yields the same conclusion as the previous comparisons between hearing and vision. That is, using visual and tactile analogues of various patterned stimuli, it will determine whether touch (like hearing) deals more effectively with temporally presented information while vision deals more effectively with spatially presented information.

The efficiency with which spatially and temporally presented information are processed can be assessed by comparing the speed and accuracy with which visual and tactile displays are identified when they contain either spatially, temporally, or spatiotemporally presented information. If touch specializes in processing temporally presented information, then temporally presented stimuli should be identified more rapidly and more accurately than equivalent spatially presented stimuli. When the stimuli have equivalent spatially and temporally presented characteristics the minimal processing principal should apply. In this case a tactile-spatiotemporal display should also be processed more rapidly and more accurately than a tactile-spatial display because touch should focus on the more efficiently processed temporal characteristics of the spatiotemporal display. On the other hand, if vision specializes in processing spatially presented information, visual-spatial and visual-spatiotemporal stimuli should be identified more rapidly and more accurately than equivalent visual-temporal displays.

2. METHOD

2.1 SUBJECTS

Twelve subjects participated in this experiment. All were students at the University of Tasmania, and none reported any uncorrected auditory or visual deficits.

2.2 APPARATUS AND PROCEDURE

The stimuli used were 48 pairs of either visual or tactile pulses. One of the two pulses in each pair was always more intense than the other. The subjects' task was to judge which pulse in each pair was the more intense.

The tactile stimuli consisted of two 250 Hz square-wave vibrations presented via single Oticon bone-conduction hearing aids placed under the index and third finger-tips of the subjects' preferred hand. These signals were generated by an Applied Engineering Super Music Synthesizer controlled by microcomputer. The amplitude of the high-intensity pulse was either 33, 30, 27, or 24 dB, with that of the low-intensity pulse being set 6 dB below that of the high-intensity pulse. Pilot studies indicated that the relative sensitivities of the subjects' index and third finger-tips often differed, thus making it difficult to determine which pulse was more intense when a difference less than 6 dB was used.

As the vibrators generated discernible auditory signals, they were mounted inside an acoustically dampened box into which the subjects hand was placed. In addition, the subjects wore both protective ear-plugs and industrial ear-phones. With these three levels of protection, it was not possible to hear the acoustic signals generated by the vibrators.

Three methods of presenting the tactile stimuli were used. In the tactile-temporal condition the first pulse was presented simultaneously to both finger-tips for 550 ms, then, 300 ms after its termination, the second pulse was also presented simultaneously to both finger-tips for 550 ms. In the tactile-spatial condition both pulses were presented simultaneously for 550 ms to different finger-tips. In the tactile-spatiotemporal condition the first pulse was presented for 550 ms to the subjects' index finger-tip. Three-hundred milliseconds after the termination of the first pulse, the second pulse was presented for 550 ms to the third finger-tip.

The visual stimuli were pairs of luminous bars generated by a Sprite Graphics card controlled by microcomputer and displayed on a Sony CVM110 VDU. Each bar extended across 3.30 degrees of visual field with a height of 0.75 degrees of visual field. The luminance of the high-intensity bar in each pair was set at either 12.5, 11.6, 9.8, or 6.7 cd/m2, with the corresponding low-intensity bar being set at either 11.6, 9.8, 6.7, or 5.1 cd/m2 respectively. This difference in luminance provided an analogue of the amplitude differences between the tactile pulses.

As with the tactile stimuli, three different presentation methods were used with these visual stimuli. In the visual-temporal condition, the two bars were presented sequentially at the center of the display. Each bar was displayed for 550 ms with a 300 ms ISI. In the visual-spatial condition both bars were displayed side-by-side on the screen for 550 ms. Finally, in the visual-spatiotemporal condition, the first bar was presented for 550 ms on the left-hand side of the display. The second bar was then presented for 550 ms on the right-hand side of the display 300 ms after the termination of the first bar.

Each subject undertook 10 practice and 48 experimental trials in each of the six conditions. On each trial they were presented with the two pulses via the assigned method, then identified which pulse was the more intense by pressing one of two buttons. They were instructed to respond as quickly as they could. The button-press was operated with the subjects non-preferred hand because their other hand was always restrained in the vibrator-box. This was done to preclude any reaction time differences induced either by having their preferred hand restrained only in the tactile conditions or by responding with different hands in different conditions. Both reaction time and accuracy data were recorded. Reaction timing was commenced at the onset of the second pulse in the temporal and spatiotemporal conditions, and at the onset of the display in the spatial conditions. This ensured that there was a constant delay (550 ms) between the onset of timing and the termination of the display regardless of presentation method. After each trial the correct response was displayed on the VDU.

3. RESULTS

Mean reaction times for correct responses and mean $\underline{d}$'s for each presentation method are shown in Table I. Dprimes were calculated using the proportion of correctly identified stimuli in which the first pulse was more intense as the hit rate, and the proportion of incorrectly identified stimuli in which the second pulse was more intense as the false-alarm rate.

TABLE I

MEAN REACTION TIME AND ACCURACY FOR EACH PRESENTATION METHOD

	Presentation Method		
Modality	Temporal	Spatial	Spatiotemporal
		Reaction Time (s)	
Touch	0.66	0.84	0.71
Vision	0.75	0.59	0.58
		Accuracy ($\underline{d}'$)	
Touch	3.00	1.78	2.65
Vision	1.82	2.16	2.21

It is clear that, for touch, temporal presentation of the stimuli led to better performance than spatial presentation. This was true both for speed ($\underline{F}(1,11) = 29.87$, $\underline{p} < .001$) and for accuracy ($\underline{F}(1,11) = 139$, $\underline{p} < .0001$). That is, responding was both faster and more accurate with temporal presentation.

For vision, the opposite results were obtained. Spatial presentation of the stimuli led to better performance than temporal presentation. This was true both for speed ($\underline{F}(1,11) = 9.08$, $\underline{p} < .05$) and for accuracy ($\underline{F}(1,11) = 5.81$, $\underline{p} < .05$). That is, responding was both faster and more accurate with spatial presentation.

In both modalities, spatiotemporal presentation led to results very similar to those found with the preferred presentation method. That is, for touch the reaction times for spatiotemporal presentation were not significantly different from those for temporal presentation ($\underline{F}(1,11) < 1.0$, n.s.), but were significantly shorter than those for spatial presentation ($\underline{F}(1,11) = 7.76$, $\underline{p} < .05$). For vision, the reaction times for spatiotemporal presentation were not significantly different from those for spatial presentation ($\underline{F}(1,11) < 1.0$, n.s.), but were significantly shorter than those for temporal presentation ($\underline{F}(1,11) = 5.44$, $\underline{p} < .05$). In each case, the results for accuracy mirrored those for speed.

4. DISCUSSION

The tactile-temporal task was performed more rapidly and more accurately than the tactile-spatial task, while the visual-spatial task was performed more rapidly and more accurately than the visual-temporal task. As there was no evidence of subjects employing speed-accuracy trade-offs, the results must reflect some differences in task difficulty between conditions.

These differences in task difficulty may be attributed to differences in the coding strategy used by each modality, rather than to any procedural differences between presentation methods. The spatial and temporal tasks in each modality differed only in terms of those features which characterised them as being either spatial or temporal in nature (ie. being presented at different locations or at different points in time). All other stimulus parameters, such as duration and intensity, were constant across conditions. Further, these features which distinguished the spatial and temporal tasks were combined in the spatiotemporal tasks, which were performed as well as the 'preferred' tasks in each modality. The presence of 'non-preferred' information (spatial for touch, temporal for vision) thus did not disrupt or prevent efficient processing of the 'preferred' information, but merely led to less efficient processing when it was presented alone. Hence, these results support the hypothesis that temporally presented information is processed more efficiently by touch while spatially presented information is processed more efficiently by vision.

The similarity in performance between the spatiotemporal tasks and the 'preferred' tasks (temporal for touch, spatial for vision) has additional significance. First, both these results are consistent with the minimal processing principle. Second, while the visual data confirms O'Connor and Hermelin's (1972) finding that visual presentation encourages a spatial encoding strategy, the tactile data suggests that tactile presentation encourages temporal processing.

It can be concluded that temporally presented information is processed most efficiently, and is more salient, in the tactile modality, while spatially presented information is processed most efficiently, and is more salient, in the visual modality. As speech is primarily characterised by temporally changing features, these conclusions begin to provide the information-processing foundation to account for the practical finding that touch may be a more effective medium than vision for non-auditory aids for the hearing impaired.

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Acoustics in the Eighties

Session 2B: Propagation and Attenuation II

Acoustics in the Eighties

SELECTION OF ACOUSTIC BACKGROUND BY SPECTRAL CONTENT

B.G. Marston

SELECTION OF ACOUSTIC BACKGROUND BY SPECTRAL CONTENT

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ABSTRACT

Atmospheric absorption is an important aspect in the assessment of long range noise propagation (0.1 km - 10 km). It is a function of temperature, pressure, frequency and atmospheric composition. The effects of changes to these variables on predicted and measured dB(A) sound pressure levels are examined for the range 0°C to 40°C and for 5% R.H. to 100% R.H. The implications to the measurement of acoustic background and to the prediction of sound pressure levels are discussed with reference to the spectral content of a range of acoustic sources. Variations in predicted levels of between 2dB and 40dB have been observed and comments are offered on the reasons for these variations.

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1. INTRODUCTION

Despite extensive research and experience accumulated since the scientific basis of acoustics was established last century, our best predictions are still only reasonably approximate.

Over short distances (up to 100 metres) with well controlled conditions, reasonable accuracy can be achieved. Because so much of acoustics deals with short range propagation and because of developed accuracy of prediction in this area a false impression is created about the accuracy of noise prediction over long distance.

Implicit in all short range predictions are a number of assumptions which can be traced back to architectural acoustics. Short range predictions assume that every thing is static except for the propagating sound. The bounding surfaces are assumed to be uniform and the supporting medium, air, is considered to be homogeneous and still.

2. PREDICTION MODELS

The most basic and widely used propagation models use only distance attenuation and frequency dependent barrier attenuation to obtain a conservative estimate of noise levels. Any discrepancy between the predicted level and measured levels are put down to 'anomalous' attenuation. More sophisticated models categorise this 'anomalous' attenuation into ground effect, vegetation absorption and meteorological effects. Almost invariably these factors are treated by empirical tables based on site specific data.

The first two factors have received a great deal of attention and progress is being made in understanding the basic mechanisms. Ground effect appears to be simply ground impedence causing destructive interference between direct and reflected sound. Vegetation absorption appears to be a combination of ground impedence in vegetation modified soil, low frequency sound scattering and for very dense vegetation, high frequency absorption by leaves and branches.

The last factor, meteorological effects, is still only partially understood but there is growing awareness that the support medium, the atmosphere, plays an important part in the variability of outdoor sound levels.

One aspect that is rarely touched upon in predictive modelling is the role played by atmospheric absorption.

3. ATMOSPHERIC ABSORPTION

Atmospheric absorption is a re-distribution of energy in the acoustic wave from the pressure wave to potential energy in the vibrational modes of atmospheric molecules. Energy taken up during the compression phase is stored and released into the rarefraction phase diminishing the amplitude of the wave. In classical atmospheric absorption theory, energy is transferred by conduction and viscous losses. In modern theory, energy is also transferred to the vibrational modes of atmospheric molecules. Because of this there is a clear link between the state and composition of the atmosphere, and the atmospheric absorption. In all, twenty four interactions have been identified but only four gases play a significant role in the energy transfer.

The concentration of the first three, nitrogen, oxygen and carbon dioxide is for all intents and purposes constant. The fourth, water vapour, although only in small concentrations, has a powerful influence. The close proximity of one of the natural vibration modes of water with one of the vibration modes of the oxygen molecules allows water vapour to strongly influence the acoustic absorption properties of oxygen.

Carbon dioxide fluctuations are often credited with causing large variations in absorption but the carbon dioxide molecule only affects very low frequency sound. It should be remembered that in plant communities changes in carbon dioxide concentrations are accompanied by wide fluctuations in humidity.

General acoustics relied for many years on the work of researchers such as Knudsen (1933) and Kneser (1933). Kneser's Monogram of 1940 is still in wide spread use. More theoretical studies of atmospheric absorption took place in the areas of nuclear physics and chemistry and it was not until the 1970's that a systematic approach to calculating atmospheric absorption re-entered general acoustics in the form of works by Brazley and Piercy et al. It is now possible to calculate with reasonable accuracy the absorption over a range of temperatures, pressures and compositions.

4. ABSORPTION AND SOUND PROPOGATION

Implicit in almost all commercial predictions are a basic set of temperature and humidity conditions. The speed of sound, if stated, is either 340 metres per second or 343 metres per second. These correspond to temperatures of 15°C and 20°C. What is the significance of these values? Fifteen degrees corresponds to the world mean equilibrium temperature. Twenty degrees corresponds to a design temperature commonly used in air conditioning work. The relative humidity is usually either 50% R.H. or 70% R.H. The combination of 20° Celsius and either 50% R.H. or 70% R.H., falls within the 'human comfort' zone of psychometric charts. These values may be appropriate for architectural acoustics.

World temperature extremes range from plus 58°C to minus 88°C (Aziz, Libya, 1922; Vostock, Antarctica, 1962). Even within Australia temperatures range from plus 53°C to minus 22°C (Cloncurry, Queensland; Kosciusko, N.S.W.). It should be asked if predictions made for a temperate 20°C and 50% R.H. are applicable for the snow fields of the Snowy Mountains or the extremes of the Outback.

Relative humidity is a 'measure' of the moisture content of the atmosphere. One climatologist is credited with describing relative humidity as 'the meteorological equivalent of absolute humbug'. It can not be used in isolation and must always be qualified by an accompanying temperature. A more appropriate descriptor is moisture content in grams of water per kilogram of dry air (gm/kg). For a single value of moisture content the relative humidity can vary from almost zero to upward of 140%.

The moisture content is controlled by that source region of the air mass affecting the assessment area and the conditions that existed in that source region. Australia is influenced by maritime polar, maritime tropical and continental tropical air masses with moisture contents varying from 3 gm/kg to over 20 gm/kg. Cooling of the air mass below the temperature of the source area can reduce the moisture content and hence reduce the potential relative humidity allowing for extremely low relative humidities as the air mass warms.

Within any 24 hour period, within the same air mass, the temperature and humidity will vary along a single line of moisture content only jumping to a new line if condensation takes place or a new air mass moves into the area.

5. TEMPERATURE-HUMIDITY ATTENUATION PLOTS

For the standard temperature-humidity combination, the moisture content line is approximately 8 gm/kg. Figures 1 to 6 show plots of attenuation versus temperature and relative humidity. The left hand plots is a contour map of predicted noise levels. The right hand plots are isometric view of the same surface with the lowest level subtracted from the remaining values. The value at the top of each plot is the predicted value for 20°C and 50% R.H. with the positive and negative range of values. In all cases the 8 gm/kg line follows the trough across the left hand corner. At the lower value of 4 gm/kg the values form a ridge sweeping across from top left to bottom right. For convenience the values are limited from 5% to 100% R.H. and from 0°C to 40°C.

For the majority of situations the predicted values fall in the region where moisture content is greater than 4 gm/kg and the variation is only 2 or 3dB(A). For values where the moisture content is less than 4 gm/kg predicted values fall off. The rate of fall-off is dependent on the spectral content of the original spectrum. For spectra with a large high-frequency content the variation is quite wide. Spectra with a great deal of low frequency content show only small variation. Over the range of values shown the 63H_z frequency band has a variation of between 0.1dB per kilometre to 0.6dB per kilometre. At the other end of the spectra the 8H_z frequency band varies from 49dB per kilometre to 312dB per kilometre.

6. IMPLICATIONS

These variations affect both the precision of predictions of impact and the measurement of noise background. The standard temperature-humidity combination falls along the 8 gm/kg line which gives values less than the potential maximum at 0°C and 100% R.H. As the source-receiver distance increases the discrepancies between these two values increases. At a range of 10 kilometres this discrepancy would amount to as much as 5dB(A). This would seem to indicate that the standard combination of temperature and humidity should be changed to a lower temperature and higher relative humidity. On the other hand, the predictions could be confined only to those temperature-humidity combinations experienced in the area of interest. It might be possible in colder dryer areas to allow lesser noise reduction measures than would be indicated by the standard conditions.

Similarly noise measurements taken under cold dry conditions would produce lower background measurements than would normally be expected either hiding a normally intrusive noise source or producing an excessively low background criteria.

7. FUTURE WORK

This work will be combined with other factors in a ray tracing program to examine the interaction of atmospheric absorption with other meteorological factors encountered in the real world over a variety of conditions derived from aerological charts.

No long range noise prediction can be considered realistic unless both the range and frequency of occurance of noise levels can be fully specified. This work is attempting to bring this situation closer to reality.

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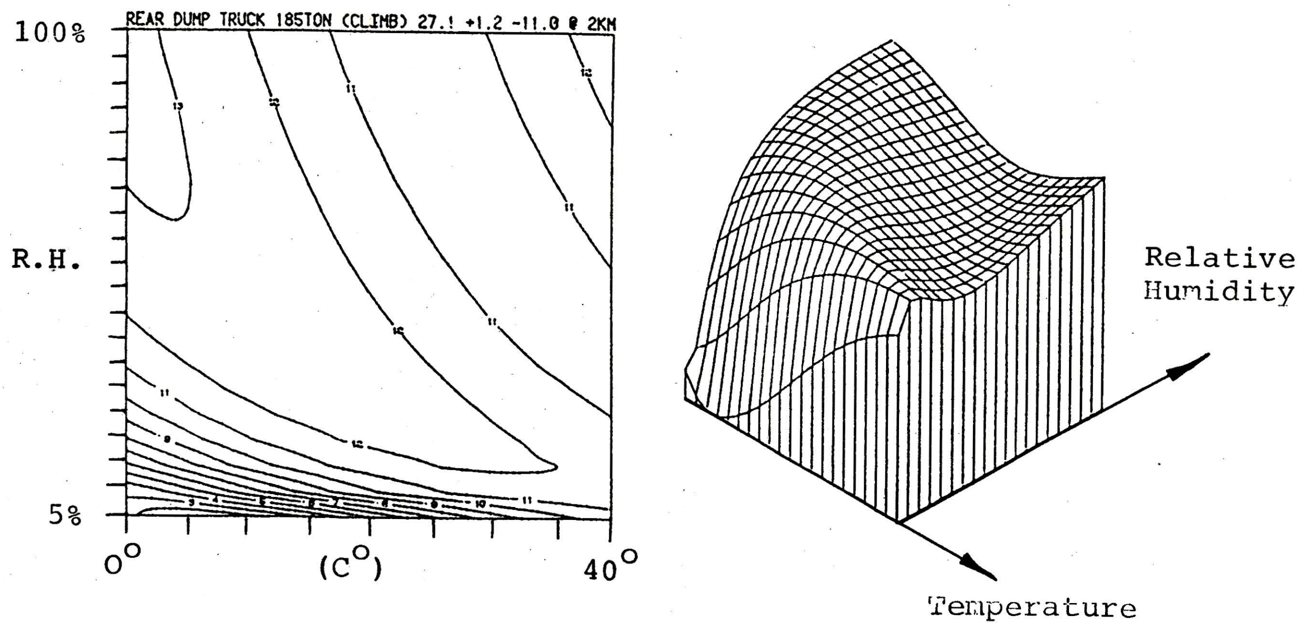

Fig. 1 - Predicted Levels at 2km for a 185 ton Rear Dump Truck

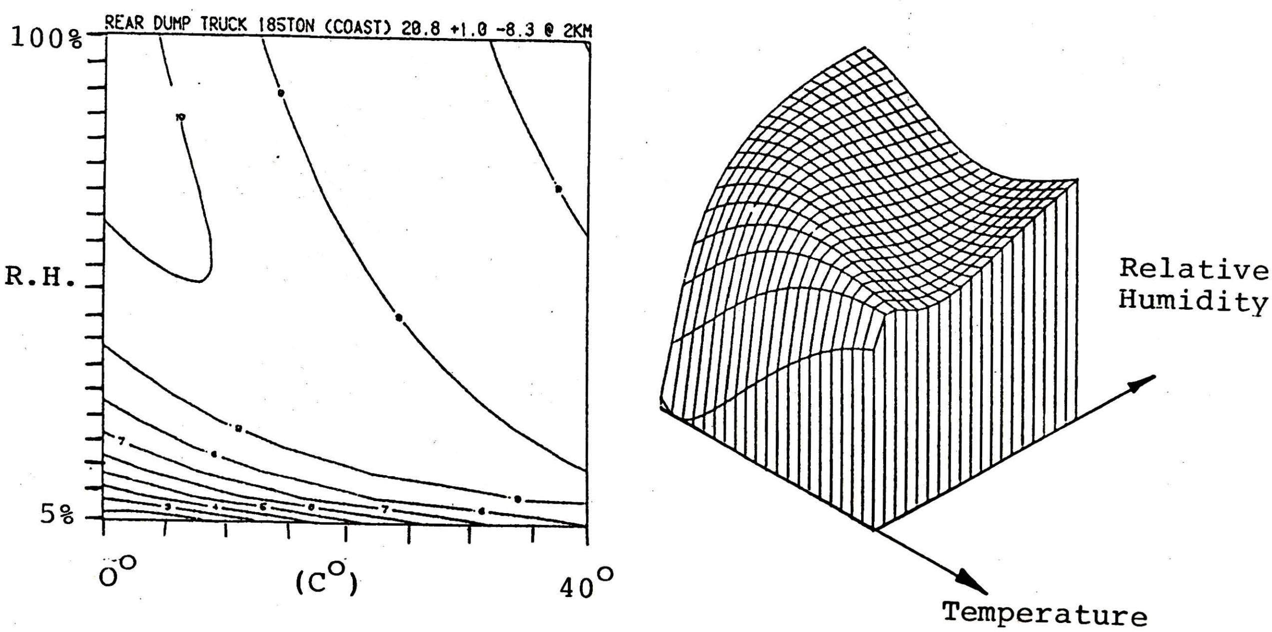

Fig. 2 - Predicted Levels at 2km for a 185 ton Rear Dump Truck

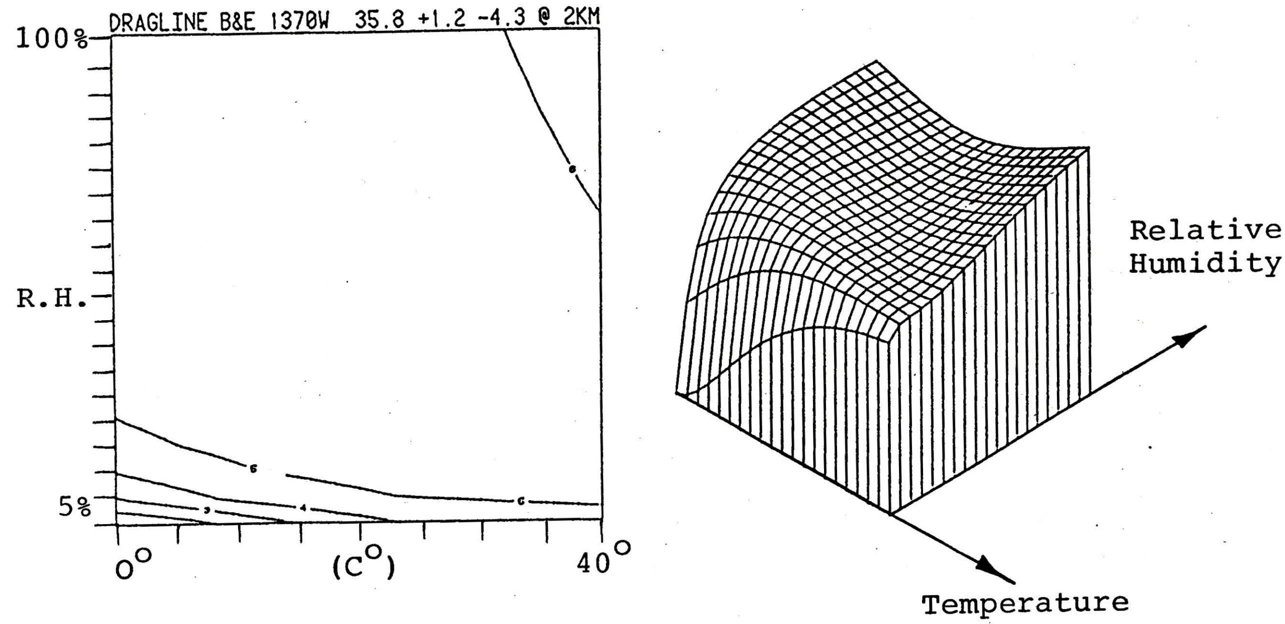

Fig. 3 - Predicted Levels at 2km for a Dragline

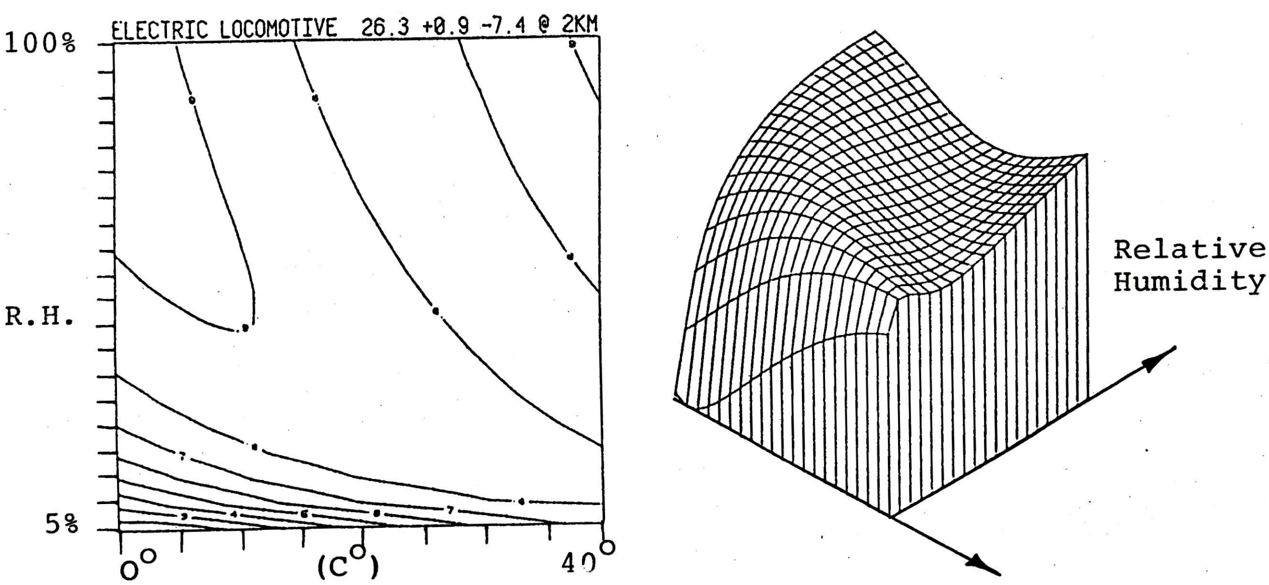

Fig.4 - Predicted Levels at 2km for an Electric Locomotive

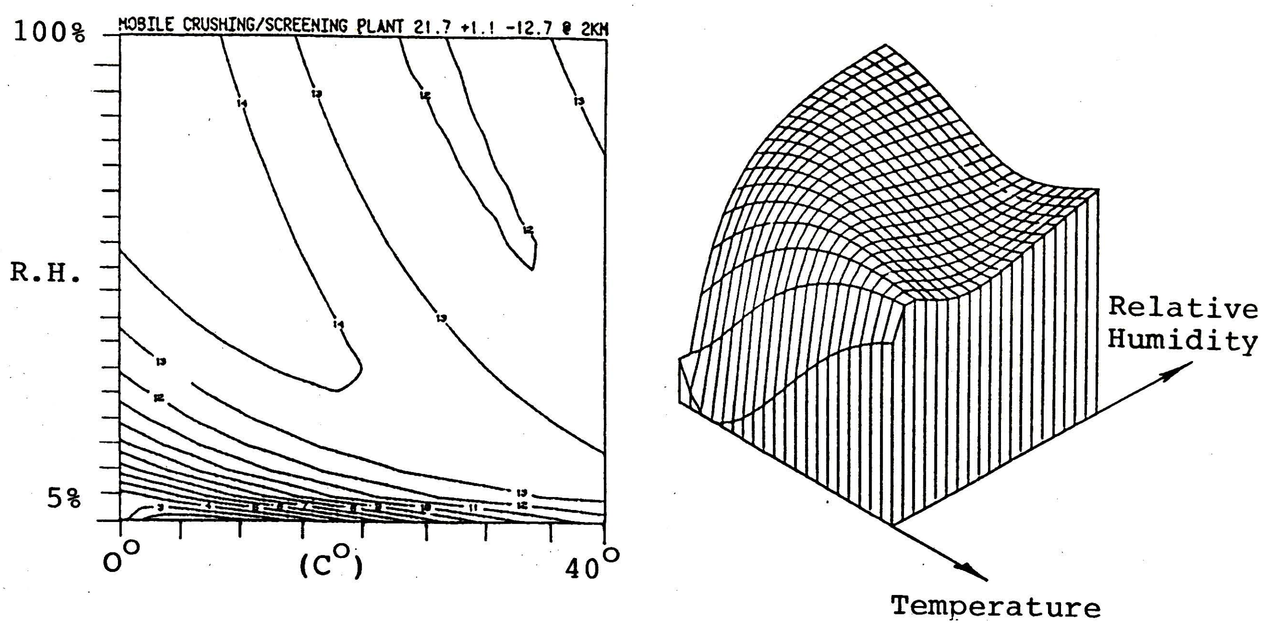

Fig. 5 - Predicted Levels at 2km of a Mobile Crushing and Screening Plant

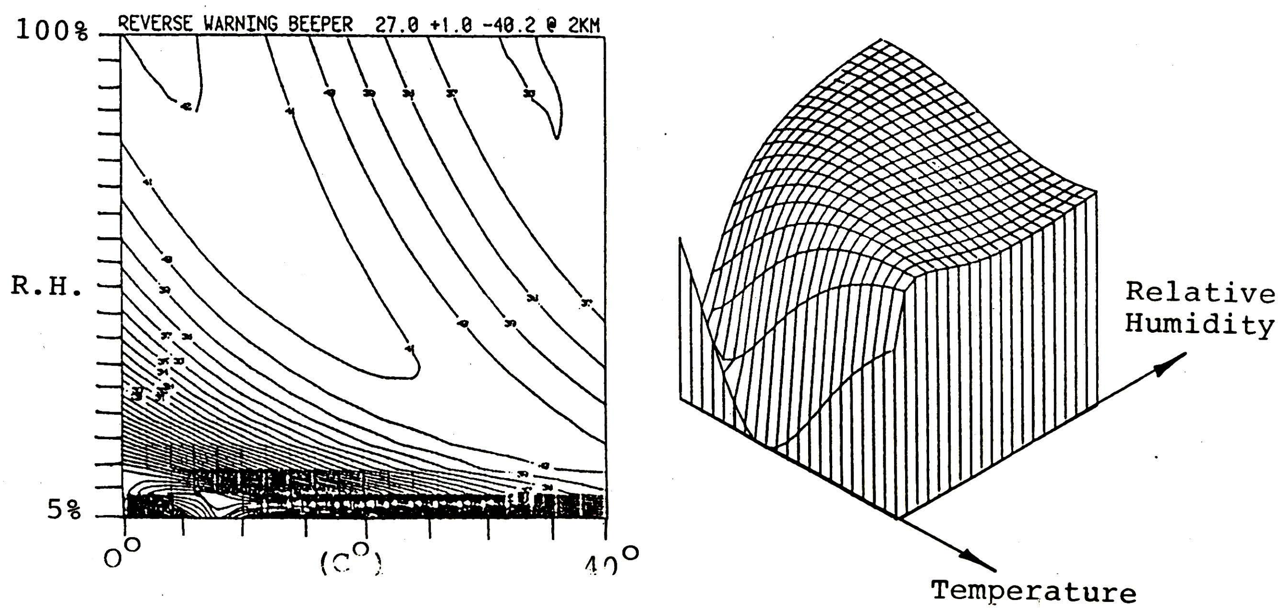

Fig. 6 - Predicted Levels at 2km for a Reverse Warning Beeper

DISTANCE ATTENUATION OF MINE VENTILATION FAN SOUND EMISSIONS

C. Tickell

DISTANCE ATTENUATION OF MINE VENTILATION FAN SOUND EMISSIONS

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ABSTRACT

BHP Engineering has prepared noise impact studies for developments at various BHP Colliery operations in recent years. The characteristics of distance attenuation of sound for the sites were determined for predictions of sound levels in the surrounding community from proposed developments. Mine ventilation fans on the sites were used as the sources for these measurements.

Sound levels were measured at regular distances up to 500 m from the fans. Results indicated that distance attenuation for the dominant frequencies of interest were between 4 and 5dB reduction for every doubling of distance rather than 6dB plus air attenuation as may be expected. This paper presents the results and comments on their implications for predictions of community sound levels from large sound sources with associated high gas flow rates (for example fans and gas-turbines).

1. INTRODUCTION

When predicting sound levels of proposed noise sources, similar sources at other locations may be used to provide typical sound level information, or manufacturers' data used, where available. However such information usually does not include measurements at distances greater than about 100 metres whilst it is common to seek predicted sound levels at much greater distances.

From the information available, predictions can be made using various methods, from 'published' computer models and their derivatives, one's own developed models, or standard rules of thumb, such as reduction of 6dB for every doubling of distance from the source.

Measurements of large sound source units (mine ventilation fans) have been obtained for prediction purposes over the past few years. The results indicate that for some configurations of source and measurement position, the 6dB reduction rule of thumb was an over estimation. In general the results indicated that the distance between source and measurement position were significant in selecting the correct 'rule of thumb' attenuation rate.

The results are presented with analyses of attenuation rates for various source-measurement position distance, and some discussion of the implications for prediction of sound levels at distances greater than 100 metres.

2. REASONS FOR PREDICTION AND SIMPLE METHODS

Prediction of sound levels at relatively large distances (greater than 100 metres) from noise sources are required in environmental assessment of new projects, either by regulatory authorities or by the prudent developer.

Information on source sound levels may be available from manufacturers, in which case for large sources the quoted sound levels are usually very close to the source (less than 10m) and in the near field, or from measurements of similar installations.

For measurements of similar installations, measurements of sound level are not usually taken at distances greater than 100 metres - reasons are typically, availability of distances greater than 100m (another property or the influence of other sources) or at these distances the source may be difficult to detect above ambient.

Mine ventilation fans for collieries in New South wales are large sources, typically with exhaust areas greater than 10m^2, and are often sited in areas many hundreds of metres from the nearest other major source, (although some are as close as 30m). These provide suitable sources for evaluation of distance attenuation of sound in the atmosphere, especially as they often have major energy components at low frequency where ambient sound levels are low.

A simple method of prediction of sound levels for these sources is to measure sound levels at increasing distances and determine the rate of attenuation at distance of greater than 200 metres, by plotting sound level against distance.

For a point source the attenuation rate is a reduction in sound level of 6dB for every doubling of distance from the source, from geometrical spreading of the sound wave. Other factors adding to this are atmospheric absorption of sound at distances greater than about 100m, ground surface absorption and impedance effects, and meteorological factors. For sources other than 'points' the geometrical spreading attenuation rate is less than 6dB, depending on the relative size of the source and the distance at which the sound is measured. This is all basic practical acoustics. But the comparison of actual measured results with theoretical attenuation rates is not often available, hence their presentation here.

3. RESULTS FOR A HORIZONTAL DISCHARGE FAN

3.1 THE SOURCE

Mine ventilation fans are used to provide airflow through underground workings to maintain a safe working environment. In order to move air through the large roadway cross-section for distances of several kilometres between inlet and exhaust, the fans are large to provide high volume flows with relatively high pressure.

Typical airflow characteristics would be a volumetric airflow range of 100 to 200 m^3/s per unit with a static pressure range of 150 to 250 mm. watergauge. Centrifugal impellor arrangements are common, with impellor diameters of 2 to 3 metres. The fans are usually sited at the exhaust end of the air circuit in pairs with discharge to the atmosphere. Volume control can be by either speed control of the impellor or variable inlet vane control with a constant speed impellor.

For the horizontal discharge fans measured, the volumetric flow rate was 150 m^3/s, when the eight bladed variable speed impellor was running at 416rpm. The exhaust discharge area was 11.7m^2.

Measurements of sound pressure level were made at distance of between 1m and 400m directly in front of the fans, at 1.5m above ground level. Each fan in the twin fan installation was measured individually.

When sound pressure levels are plotted against distance, the expected curve shapes are obtained, given in Figure 1. The measurement location had a relatively high ambient level of 40 to 45dB(A) at distances greater than 400m from the fans. One fan had a slightly higher sound level but the shape of the S.P.L.-Distance curves are virtually the same.

The information available can be manipulated and presented in different formats. One arrangement is to plot the reduction in sound pressure level from a measuring location close to the source, against distance. For example taking the sound level at 1m as the datum level and plotting the reduction in sound level which occurs with increasing distance from this point. Theoretical curves of reduction in sound level for various rates of distance doubling attenuation (eg. 3, 4, 5, 6dB reduction for every doubling of distance) can also be presented on the same graph. Figure 2 gives a typical result. Comparison between the actual and theoretical attenuation rates then allow selection of the most appropriate attenuation rate for prediction purposes. Curves could be presented with distance as either linear scale or logarithmic scale depending on personal preference and accuracy required. Again, this is all simple, straightforward practical acoustics.

3.2 ANALYSES OF RESULTS

Using the methods describe above, the datum distance was varied in calculation from 1m to 50m from the source. The theoretical attenuation rate curves changed slightly with different datum points, having lower gradients for increasing distance between datum point and source. Comparisons were made for different datum points, of theoretical attenuation rates against A-weighted, linear and octave band sound pressure levels.

As would be expected, as the datum distance increased from 1m to 50m, the actual attenuation rate increases.

With a datum distance of 1m, the attenuation rate at 200 to 400m varied between 3dB and 6dB, averaging around 4 to 4.5dB. At a distance of 10m, the attenuation rate was around 5 to 6dB, whilst with a datum distance of 20m and above, the attenuation rates were greater than 6dB, averaging between 7 and 9dB.

When the sound levels were corrected for the contribution of background sound levels, the analyses were similar to uncorrected curves.

4. VERTICAL DISCHARGE FANS

4.1 THE INSTALLATION

Another mine ventilation facility which has been measured has vertical discharge fans, with the discharge being at around 8 metres above ground level.

Measurements of sound pressure levels around this type of fan have the added consideration of directivity, but this causes little variation at distances greater than about 70m from the fans.

The particular installation was a triple fan installation with only two on at any time. It was located in a grassed pasture, allowing measurements in different directions from the fan axis. These were taken at distances of up to 300m from the fans, under different cross wind conditions.

4.2 ANALYSIS

Again the analysis was done for comparison of actual reduction in sound level against theoretical curves. On this occasion variations in datum distances were not investigated. The attenuation rate for each weighting and octave band were tabulated for conditions of upwind, across wind and down wind, and are given in Table 1. (3.9) Datum distances were 5m for the northern direction, 25m for the western direction, 90m for the southern direction, 25m for the north west direction, and 60m for the eastern direction.

The least attenuation rate over most frequency bands occurred for the condition of measuring West of the source with a South to South East wind of 1 to 4 m/s, with attenuation rates between 3 and 4.4dB. With low to calm wind speeds, the attenuation rates were higher for both across wind and downwind conditions. High attenuation rates over all frequency bands were noted for across wind and down wind conditions with attenuation rates up to 9dB(A). The 250 Hz octave band attenuation rate was generally the highest for each condition. The A-weighted attenuation rate varied between 4 and 7dB.

The attenuation rate appeared to be higher where the sound travelled across flat, open pasture to the measurement location rather than more direct through the air to a hillside location. This suggests that over flat, open, grass pastures there is a reasonable amount of surface absorption, although it has not been quantified.

5. IMPLICATIONS OF THE ANALYSES

When predicting sound levels from proposed installations, if an operating source of the same type is not available for distance attenuation measurements, it is normal practice to assume a sound level at a particular distance from the source (usually given by the supplier) and then determine sound levels at various locations distance from the source by using a geometric spreading attenuation rate. Air absorption and ground surface absorption are also included if the distances are greater than about 100 metres. If the source can be treated as a point relative to the distance at which the sound level is to be predicted, the attenuation rate is taken to be 6dB per doubling of distance in the far field. If the source is regarded as a line, the attenuation rate is taken to be 3dB.

Once the source generated sound level at a location has been determined, this can be compared to the ambient sound levels at the location to determine if there will be an increase and if there is an increase in sound level, what the potential for noise annoyance will be. This can then be related back to the proposed source and at the design stage and modifications considered to reduce the sound emissions and annoyance potential if required.

For this study the measured attenuation rates have been dependent to some extent on wind conditions, the relative direction from the source to the measurement location, and the datum distance selected.

6. CONCLUSION

When making predictions for sound levels at distances greater than 200m from large sound sources such as mine ventilation fans, a set of attenuation curves have been obtained. These allow more accurate prediction of variations in sound levels expected and indicate how conservative estimates or predictions can be, depending upon the datum distances used or available.

The analyses could obviously go further, and when time is available this will be done. In the meantime, the results reaffirm that when measuring sound levels for prediction purposes, the distance of the measurement location from the source should be in the far field.

TABLE 1

ATTENUATION RATES PER DOUBLING OF
DISTANCE FOR VARIOUS WIND CONDITIONS

DOUBLING ATTENUATION: dB per doubling of Distance

CONDITION	OCTAVE BAND CENTRE FREQUENCY - Hz										WEIGHTING		
	31.5	63	125	250	500	1000	2000	4000	8000	16000	A	C	LIN
1. UPWIND - West of source with a West wind, 4 m/s	4.9	4.5	4.	7	6	5	5				4		4
2. ACROSS WIND													
i) North of source with a a Westwind 4 m/s	4.5	4	5	6	5	4	5	5			5		4.5
ii) North-west of source with a West wind 1 m/s	4.4	5.3	5.6	8	6.5	6.5	7	5.6	6.1	4.5	5.6	5	5
iii) West of source with a S to S.E wind 1 to 4 m/s	3.5	3	3.2	4.4	4.4	3.7	3.2	3			4.2		1.5
iv) South of source with a West wind 2 to 4 m/s	4	4	4	6.5	3.6	4.5	5.3				4.5 to 7		4.5
v) East of source with a S-S.E wind 1 to 4 m/s	3.3	4.5	3.2	4.5	6.3	5.7	5.1	4			6.3		3.[illegible]
3. DOWNWIND													
i) North of source with wind 1 m/s S	3.5	3.5	4.8	6	5	3.5	4.6		5.5		4.9	3.8	4
ii) East of source with wind 1 m/s West	4.2	4.2	4.2	7.3	7	6.2	5.4	1	4.2	3	5.4	3.7	4.[illegible]
iii) North-East of source with wind 1-4 m/s South to South-East	5	5.5	6	9	6.5	6.5	6	7			7		7

Figure 1

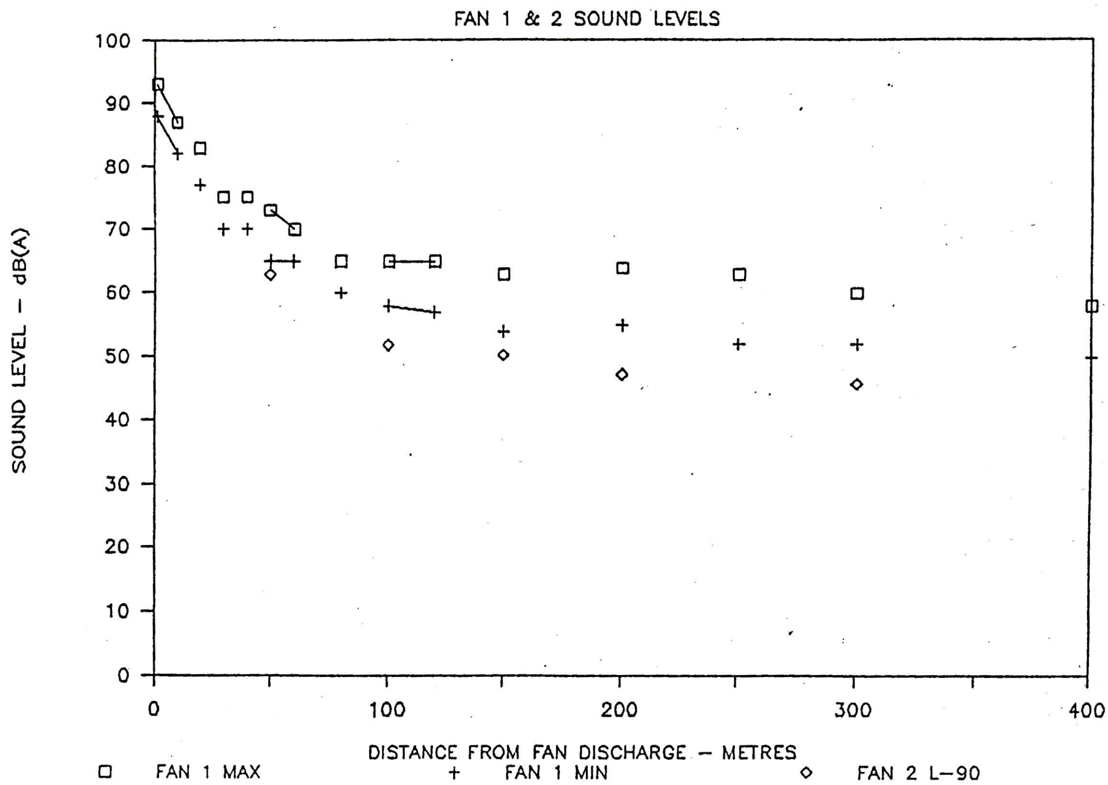
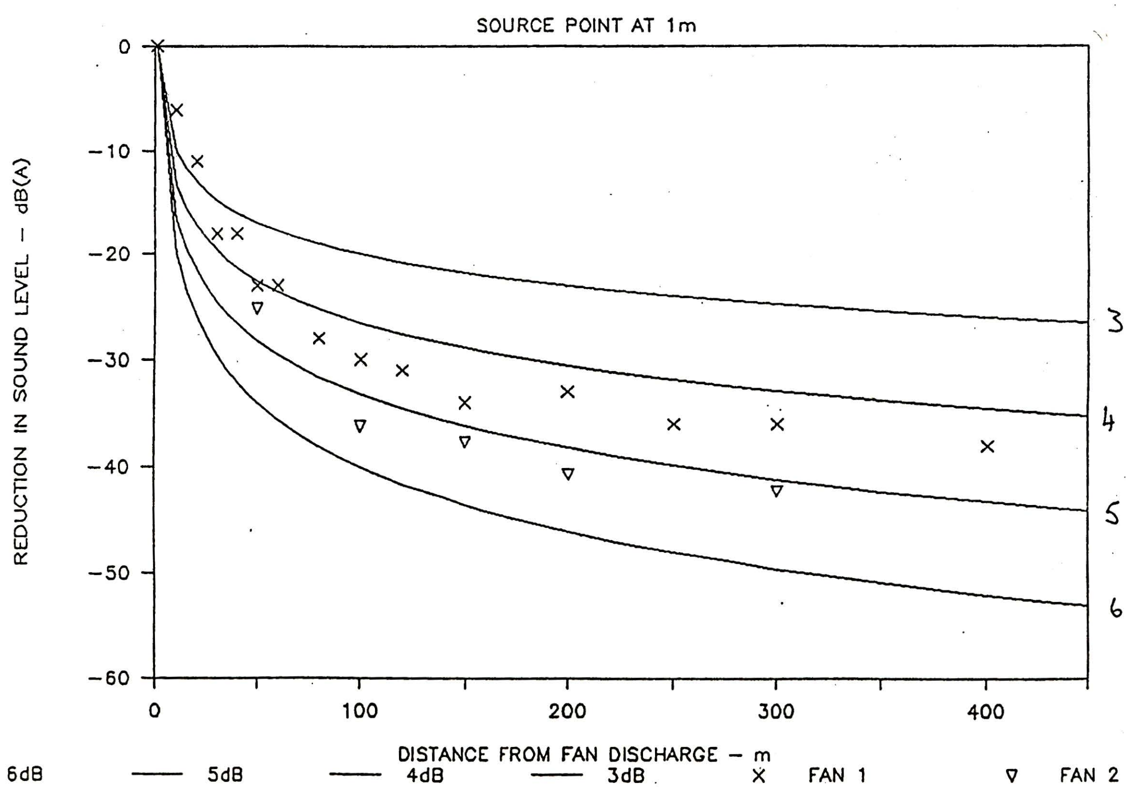


Figure 2



Acoustics in the Eighties

Session 2C: Acoustic Properties

ESTIMATION OF THE PHYSICAL AND ACOUSTICAL PROPERTIES OF A DUCT USING MEASURED EIGENFREQUENCIES

Wu Qunli and F. Fricke

ESTIMATION OF THE PHYSICAL AND ACOUSTICAL PROPERTIES OF A DUCT USING MEASURED EIGENFREQUENCIES

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ABSTRACT

The work presented in this paper, on the estimation of physical and acoustical properties of ducts, is based on existing eigenfrequency theories. The work starts with the estimation of duct length using measured eigenfrequencies. It is extended to include the detection of a partial blockage in a duct. The experimental results show that the eigenfrequency method is capable of estimating the physical properties of a duct. The use of measured eigenfrequencies to estimate the acoustical properties of materials is also discussed.

ACKNOWLEDGEMENT: We would like to gratefully acknowledge the financial support given by Vipac for this project.

1. INTRODUCTION

Much work has been undertaken on the eigenfrequencies of an enclosure or space, starting with the classical study of the rectangular room with rigid boundary conditions. The eigenfrequencies for some special shaped enclosures were studied by introducing mathematical functions (Morse 1968). Based on these studies, statistical analyses of eigenfrequencies were made during 1940's (Maa 1939;Bolt 1945). This work led to many applications of eigenmode theory to room acoustics and acoustical measurements. In recent years, researchers turned their attentions to the search for eigenfrequencies of complex enclosures, such as an enclosure containing blockage or rigid objects (El-Raheb et al 1982;Leung et al 1982). The development of the finite element methods for acoustics makes it possible to solve for the eigenfrequencies of complex shaped cavities with non-rigid boundaries (Petyt 1976).

The previous studies showed that the eigenfrequencies are related to the physical and acoustical properties of the enclosure. Eigenfrequency theory has been applied to the understanding of the behaviour of sound in enclosures but the use of eigenfrequencies to estimate the size, shape and position of objects in an enclosure has attracted very little attention (Domes 1979).

The present work is based on the eigenfrequency theories which are available in the references already cited. The study emphasizes the application of these theories to estimating the physical properties of the enclosure. The study starts with the estimation of the duct length using the measured eigenfrequencies. An extension is made to detect the size and locations of a blockage in the duct. The experimental results show the eigenfrequency method is capable of estimating the physical properties of the duct.

2. ESTIMATION OF PHYSICAL PROPERTIES OF A DUCT USING MEASURED EIGENFREQUENCIES

2.1 ESTIMATION OF THE LENGTH OF A DUCT

Consider a one dimensional duct with rigid ends. The solution of eigenfrequencies of the duct is

$$f_n = nc/2l \qquad n = 1,2,3,... \tag{1}$$

where c is the speed of the sound in the medium, l is the length of the duct and, f_n is the nth eigenfrequency.

Eq. (1) shows the relationship between the duct length and the eigenfrequencies. Eq. (1) is not good for estimating the duct length because the order of the eigenfrequency is difficult to measure, in many cases, especially when the duct has very low eigenfrequencies. A better estimate of the duct length can be obtained if Eq. (1) is written in the following form:

$$l = c/2(f_{n+1}-f_n) \qquad n = 1,2,3, \; ... \tag{2}$$

Eq. (2) shows that the length of the duct can be calculated by measuring any two adjacent eigenfrequencies. This makes it much easy to estimate the length of the duct by measured eigenfrequencies.

The correction according to the duct terminal impedance should be made if the duct ends cannot assumed as rigid (Kuttruff 1973).

2.2 DETECTION OF BLOCKAGE IN A DUCT

Consider the one dimensional model of a duct with a blockage as shown in Figure 1.

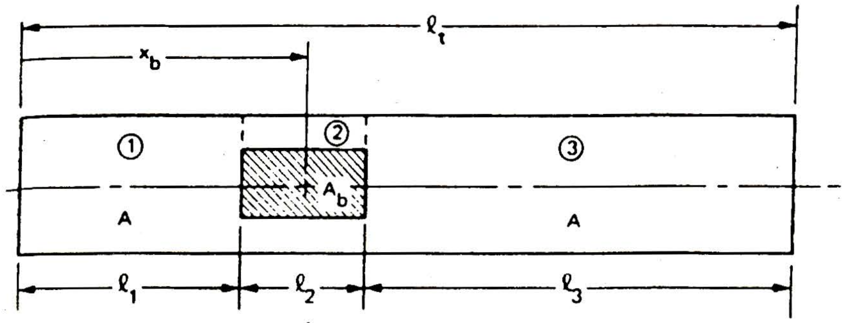


Figure 1 The duct with a blockage

The duct and blockage cross section area are defined by A and A_b respectively. The blockage partitions the duct into three segments with length l_1, l_2, and l_3, where l_2 is the blockage length.

Assuming the duct ends are rigid, the eigenvalues of the duct, derived from the wave equation and the boundary conditions, are (El-Raheb et al 1982)

$$\sin(k_f l_1)\,[(1-\beta)\cos(k_f l_2)\cos(k_f l_3)-\sin(k_f l_2)\sin(k_f l_3)]$$
$$+(1-\beta)\cos(k_f l_1)\,[(1-\beta)\sin(k_f l_2)\cos(k_f l_3)+\cos(k_f l_2)\sin(k_f l_3)] = 0 \quad (3)$$

where $k_f = 2\pi f_n/c$ is the eigenvalue of the duct, f_n is the eigenfrequency, $\beta = A_b/A$, is the cross section area ratio of the blockage and duct. In the limiting case of null blockage when $\beta=0$, Eq.(3) admits the expected elementary solution for the empty straight duct, $f_n = nc/2l_t$, ($l_t = l_1 + l_2 + l_3$, $n=1,2,3,\ldots..$). When $\beta=1$, which means total blockage, Eq.(3) becomes,

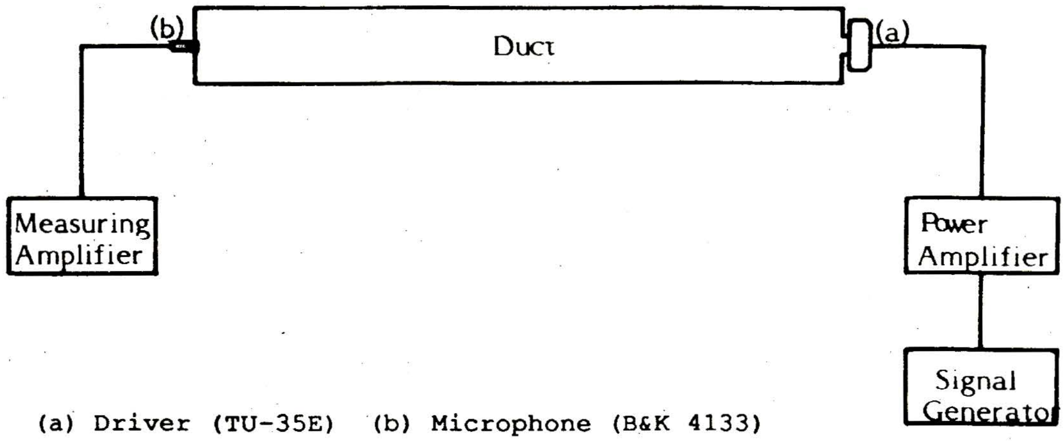
$$\sin(k_f l_2)\cos(k_f l_1)\cos(k_f l_3) = 0$$
$$\Rightarrow \cos(k_f l_1) = 0 \quad \text{or} \quad \cos(k_f l_3) = 0 \qquad (4)$$

which corresponds to the equations for straight ducts with rigid ends.

3. EXPERIMENTAL RESULTS

3.1 EXPERIMENTAL SET-UP

The experimental study was conducted in a 104 mm diameter, 2 metre long duct. The duct cut-off frequency is about 2000 Hz. The duct ends are closed with 13 mm thick plastic glass with a 1/2 inch microphone and a 1 inch driver in the centres. The experimental set-up is shown in Figure 2.



(a) Driver (TU-35E) (b) Microphone (B&K 4133)
(c) Signal generator (Philips PM 5133)
(d) Measuring Amplifier (B&K 2606).

Figure 2 Experimental set-up for the measurements of eigenfrequency

The eigenfrequencies of the duct were measured by slowly increasing the frequency of the signal.

3.2 THE RESULTS OF ESTIMATION OF THE DUCT LENGTH

The experimental results of the duct length were calculated according to the Eq. (1). The speed of sound in the air is estimated by,

$$c = 331.45\sqrt{(1 + t/373.16)} \qquad (5)$$

where t is the temperature in degree Celsius. The experimental results are expressed as a function of eigenfrequency shown in figure 3.

Fig. 3 shows the calculated duct length (measured length = 1988 mm) varies with the eigenfrequencies and that the calculated length has a systematic error. This is because of the assumption of the rigid duct ends.

Consider the non-rigid duct ends, the eigenfrequencies of the duct are complex quantities (Kuttruff 1973),

$$f_n' = f_n + i(\delta_n / 2\pi) \qquad (n = 1,2,3, \ldots\ldots) \qquad (6)$$

where f_n' and f_n are the eigenfrequencies of non-rigid and rigid end duct respectively, δ_n are the damping constants of the duct.

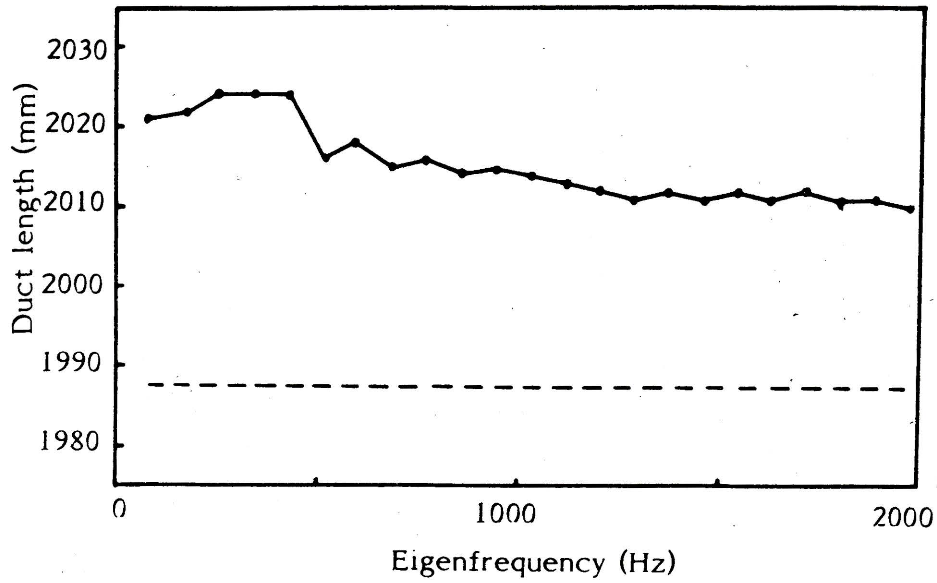


Fig. 3 Duct length estimated by measured eigenfrequencies

Because of the damping constants, the resonance frequencies of the duct, which are used in the calculation of the duct length, will shift to

$$f_{nre} = \sqrt{(f_n^2 - 2(\delta_n/2\pi)^2)} \qquad (n = 1,2,3, \ldots\ldots) \qquad (7)$$

where f_{nre} are the resonance frequencies of the duct. Eq. (7) shows that the resonance frequencies are always smaller than the eigenfrequencies. This causes the systematic error in the results presented in Fig. (3). As the damping constants are a function of frequency the calculated duct length varing with eigenfrequencies is expected. A better equation for calculation the duct length is,

$$l = nc / \{2\sqrt{(f_{nre}^2 + 2(\delta_n/2\pi)^2)}\} \qquad (n = 1,2,3, \ldots.) \qquad (8)$$

The damping constants δ_n can be estimated by measuring the half-widths of resonance curves.

$$\delta_n = \pi\ (\Delta f)_n \qquad (n = 1,2,3, \ldots\ldots) \qquad (9)$$

where $(\Delta f)_n$ are the half widths of resonance curves.

3.3 EIGENFREQUENCY SHIFT DEPENDS ON THE BLOCKAGE

The effect of a blockage on eigenfrequencies is measured up to 1900 Hz. The blockage used in the experiments is made from hard wood, with cross section area A_b = 38.72 cm^2 and length l_2 = 20.4 cm. The eigenfrequency shift is defined as the eigenfrequency difference of the duct with and without the blockage. Figures 4 and 5 show the eigenfrequency shift depends on the blockage locations. The results are expressed as a function of the measured eigenfrequencies of the empty duct.

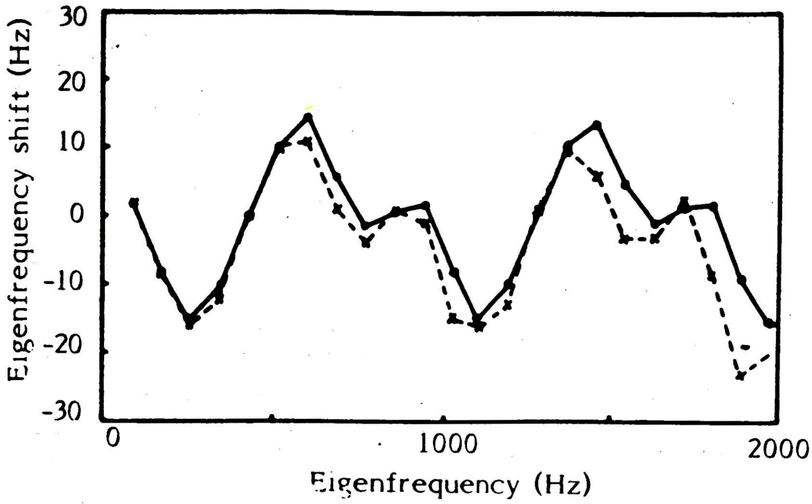


Figure 4 Eigenfrequency shift depends on the blockage.
(l_1 = 20 cm, l_2 = 20.4 cm, l_3 = 160 cm, β = 0.46)
------ experimental results, ------ theory of Eq. (4)

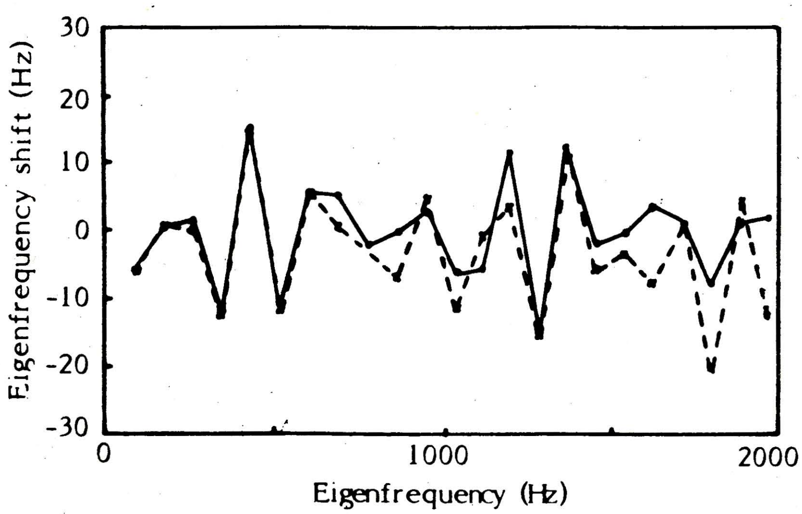


Figure 5 Eigenfrequency shift depends on the blockage.
(l_1 = 70 cm, l_2 = 20.4 cm, l_3 = 110 cm, β = 0.46)
------ experimental results. ------ theory of Eq.(4).

Figs. (4) and (5) show that the agreement between the measurement results and the theory is excellent at the frequencies below 500 Hz, the error increases with an increase of the frequency. This may be because the one-dimensional model in Eq. (3) becomes invalid at high frequencies.

Fig. (4) shows that the eigenfrequency shifts are a periodic function of the frequency. The period is 858 Hz, corresponding to the half wavelength, 19.4 cm , which is very close to the blockage length. Although Fig. (5) does not show the same property as Fig. (4), it shows the eigenfrequency shift is an asymmetrical function of the frequency at 858 Hz, which is the period of Fig. (4). The figures also show that the maximum eigenfrequency shift is a constant with the same blockage. These two properties may lead to ways of determining the size of the blockage.

3.4 ESTIMATION OF THE SIZE AND LOCATION OF THE BLOCKAGE

Eq.(3) shows the eigenfrequency depends on the blockage size and location in the duct. Eq. (3) also can be expressed in a general form,

$$Y_n\ (f_n,\ l_1,\ l_2,\ l_3,\ \beta) = 0 \qquad n = 1,2,3,\ldots.. \tag{10}$$

Eq. (10) is a five variable nonlinear function. The eigenfrequencies can be calculated when the size and location of the blockage are known. On the other hand, the size and the location of the blockage can be estimated by measuring any four eigenfrequencies. A computer program using the Newton-Raphson method (Carnahan et al 1969) is employed to solve Eq. (10). As Eq. (10) has multi-solutions the starting values and the accuracy should be choosen carefully, otherwise the program may not produce reasonable solutions. Table 1 shows the calculation results based on four measured eigenfrequencies.

Table 1. Computed values of the blockage size and location

Measured eigenfrequencie used in the computation	Computed blockage dimensions (actual values given in parentheses)			
	l_1 = (20 cm)	l_2 = (20.4 cm)	l_3 = (160 cm)	β = (0.46)
Mode 1,2,3,4	25.1	12.3	163.3	0.57
Mode 2,3,12,22	15.6	25.5	158	0.53
Mode 20,21,22,23	19.8	21.8	160.0	0.67

Another method which may more suitable to solve Eq. (10) is Gradent Method. Using this method the ill posed solutions can be got rid of. The Gradent Method is expressed as Eq. (11),

$$\text{Minimum } \{\ \Sigma\ Y_n^2\ (f_n,\ l_1,\ l_2,\ l_3,\ \beta)\ \}$$
$$l_i \geq 0 \quad i = 1,2,3 \quad \text{and} \quad 1 \geq \beta \geq 0 \tag{11}$$

This method will allow more than four eigenfrequencies in the equation. The accuracy of the dimension predictions will be expected to increase using this method.

4. CONCLUSIONS AND DISCUSSION

The data presented in Section 3 show that the physical parameters of the duct can be estimated using the measured eigenfrequencies. Further studies are needed in order to achieve higher accuracy of the estimation and to investigate the application of the method to more complex geometries, including two and three dimensional situations.

Another application of the eigenfrequency method to be considered is to estimate the acoustical properties of the material in the duct end or ends. In the experimental study, the authors found the eigenfrequency shifts and bandwidths are related to the acoustical properties of the duct end. This study may lead to a new method for the measurement of acoustical impedance of materials.

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Acoustics in the Eighties

THE USE OF WAVE IMPEDANCES IN CALCULATING THE ACOUSTIC PROPERTIES OF PLANE CONSTRUCTIONS

K.P. Byrne

THE USE OF WAVE IMPEDANCES IN CALCULATING THE ACOUSTIC PROPERTIES OF PLANE CONSTRUCTIONS

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ABSTRACT

Acoustic lagging treatments, which are used to attenuate the noise radiated from vibrating surfaces, and fabric structures, which are being used more frequently in architectural applications, are examples of acoustic constructions whose acoustic properties can be found by using wave impedance principles. These properties include quantities such as insertion loss, diffuse field sound absorption coefficient and diffuse field sound reduction index. The constructions can be modelled with a limited number of elements, for example, layers which may or may not contain porous materials and porous or impervious sheets which may or may not be flexurally stiff. Formulae are developed which relate the wave impedances across these elements. These formulae, along with formulae which relate acoustic pressures across the elements, can be used to compute the quantities of interest. A comparison is presented which relates computed and measured results for an acoustic lagging and an architectural fabric sheet.

1. INTRODUCTION

Many systems, the acoustic performance of which is of interest, are nominally plane and uniform. Acoustic laggings, which are applied to vibrating, sound radiating surfaces to inhibit the transmission of the radiated sound, are an example. An acoustic lagging is usually composed of one or more layers of porous materials, such as rockwool or fibreglass, one or more impervious barriers such as loaded plastic sheets or metal cladding sheets and airspaces. Fabric sheet constructions, which are now extensively used in air or cable supported architectural fabric structures are another example. Fabric sheet constructions can involve one or more fabric sheets and layers of porous materials and air spaces. Usually, the acoustic performance of an acoustic lagging is described by the insertion loss produced by the lagging when it is applied to a particular surface. In the case of an architectural fabric sheet construction, the sound absorption coefficient and the sound reduction index may be of interest. Generally, these performance measures are expressed in frequency bands such as one-third octave bands. Usually the nature of the sound field is specified. It would be usual, for example, in the case of the architectural fabric constructions to wish to know the sound reduction index and the sound absorbtion coefficient when the incident sound field is diffuse.

A procedure is described for computing the appropriate acoustic performance measures for a wide variety of nominally plane and uniform systems, two of which have been mentioned above. The procedure is based on the successive application of sets of transmission line formulae which relate the wave impedances across the various elements of the system. Sometimes formulae which relate the pressures across the elements must also be used. These formulae sets can be used to study the transmission and reflection of an oblique plane wave through and by the system. The oblique plane wave results then can be used to determine diffuse field quantities if these are required.

2. THE CONCEPT OF WAVE IMPEDANCE

Suppose that a sinusoidally varying force $F\cos(\omega t+\phi_F)$ is applied to a point on a compliant structure and that the resulting displacement of this point in the direction of the force is $X\cos(\omega t+\phi_X)$. The result $\exp j\theta = \cos\theta + j\sin\theta$ can be used to interpret $F\cos(\omega t+\phi_F)$ as the real part, denoted Re, of $\boldsymbol{F}\exp(j\omega t)$ where $\boldsymbol{F} = F\exp(j\phi_F)$. Similarly, the displacement can be given a complex representation $\boldsymbol{X}\exp(j\omega t)$ and so the velocity has the complex representation $j\omega\boldsymbol{X}\exp(j\omega t)$. The mechanical impedance at the driving point is the complex ratio of the complex representations of the force and the velocity. The modulus of this complex quantity specifies the magnitude of the force needed to produce a unit amplitude velocity and the argument of this complex quantity defines the phase shift between the force and the velocity.

The analogous quantity which can be applied to extended surfaces is the wave impedance. Suppose that the compliant surface shown in Fig. 1 is subject to a propagating sinusoidal pressure wave $P\cos(\omega t - k_x x + \phi_P)$. This wave is propagating in the the positive x direction with a velocity $c_x = \omega/k_x$. The complex representation of this wave is $\boldsymbol{P}\exp[j(\omega t - k_x x)]$. The surface will deflect under the influence of the applied pressure and a propagating sinusoidal displacement wave whose complex representation is $\boldsymbol{W}\exp[j(\omega t - k_x x)]$ will be

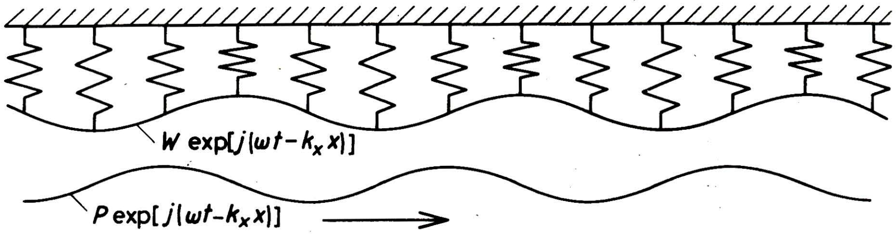


Figure 1: Infinite Compliant Surface Subject to Travelling Pressure Wave

induced on the surface. The complex representation of the corresponding sinusoidal velocity wave which travels along the surface is then $j\omega \boldsymbol{W} \exp[j(\omega t - k_x x)]$. The wave impedance at the surface shown in Fig. 1 is the complex ratio of the complex representations of the pressure and the velocity, that is , $\boldsymbol{P}/j\omega\boldsymbol{X}$. The modulus of this complex quantity specifies the amplitude of the propagating pressure wave which when applied to the surface produces a propagating unit amplitude velocity wave on the surface. The argument of this complex quantity specifies the phase shift at any position between the pressure and the velocity.

3. THE USE OF WAVE IMPEDANCES

Systems of the type of interest here are made of elements which have different acoustic properties as shown in the arbitrary system of Fig. 2. The interaction of an oblique plane wave with such a system, which is assumed to extend infinitely, can be analysed by noting firstly that the trace wavelengths on all of the elements are the same and secondly that the pressures and the normal velocities on elements on either side of an interface surface are equal. This first point means that all elements have the same wave number in the plane of the system and the second point means that the wave impedances on the surfaces of elements on either side of an interface surface are equal. Although, for some purposes, such a system can be analysed if only formulae which relate the wave impedances across the elements can be devised, it is also usually neccessary to use formulae which relate the pressures across the elements. This point is illustrated with regard to the system shown in Fig. 2. The absorption of and transmission of sound by the system is of interest. If the sound velocities in the media on both sides of the system are the same, the incident and transmitted oblique plane waves will both be at the same angle θ to the normal to the system. The wave impedance on Surface 1 $\boldsymbol{z}_1$, can be seen to be $\rho c/\cos\theta$, where ρc is the characteristic impedance of the medium in which the transmitted wave is travelling. This result arises since the component of the particle velocity of the transmitted wave normal to the surface is $(p/\rho c)\cos\theta$, where p is the amplitude of the pressure of the transmitted wave. If an appropiate "transmission line" formula is available, the wave impedance on Surface 2 $\boldsymbol{z}_2$, can be found. The wave impedance on Surface 1 $\boldsymbol{z}_1$, can be considered to be the terminating impedance, and the wave impedance on Surface 2 $\boldsymbol{z}_2$, can be considered to be the input impedance. The wave impedance on Surface 2 $\boldsymbol{z}_2$, can be used as the terminating impedance for the next element. Thus by successive application of the appropriate form of impedance "transmission line" formula, the wave impedance on the terminating surface, which is Surface N for the system shown in Fig. 2 can be found. The relationship between the incident and reflected waves on Surface N can then be established. If however it is necessary to relate the incident and the transmitted waves, it is neccessary to step back through the structure from Surface N to Surface 1 to find the complex representation of the acoustic pressure on Surface 1. Many systems of the type of interest here incorporate impervious sheets which may be modelled as

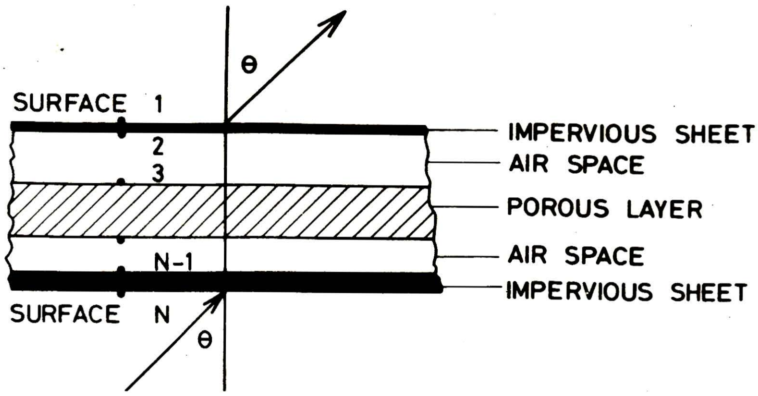


Figure 2: General Plane Construction

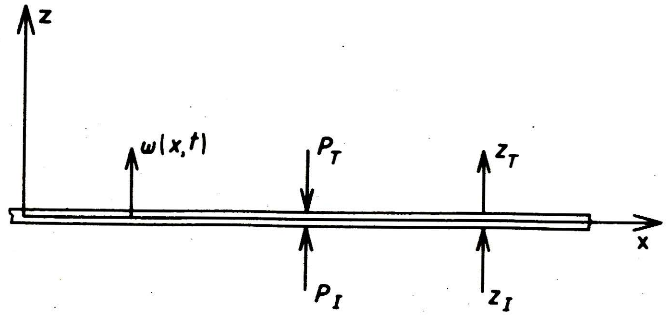


Figure 3: Impervious Barrier Model

elastic plates and porous layers or air spaces. The formulae which relate the wave impedances and the pressures across these elements are developed in the following section. It is possible to consider other types of elements by the same procedures. Other possible elements include porous tensioned fabric sheets and stiffened metal cladding sheets. These elements render the system acoustically orthotropic. However, the same basic principles already outlined can still be used in the analysis of systems which incorporate such elements.

4. THE TRANSMISSION LINE FORMULAE

4.1 TRANSMISSION LINE FORMULAE FOR AN ELASTIC PLATE

A thin elastic plate of infinite extent which is characterised by a mass per unit area m, and an elastic plate bending constant D, lies in the x-y plane as shown in Fig. 3. A time dependent nett pressure $p = p(x,t)$ on the plate induces a displacement $w = w(x,t)$ of the plate. These quantities are related by the well known elastic plate equation (Timoshenko, 1940) which here becomes

$$D\frac{\partial^4 w}{\partial x^4} + m\frac{\partial^2 w}{\partial t^2} = p. \tag{1}$$

Suppose now that the lower side of the plate is subject to a propagating sinusoidal pressure wave which is described by $P_I \cos(\omega t - k_x x + \phi_I)$ and on the upper side of the plate the pressure wave can be written as $P_T \cos(\omega t - k_x x + \phi_T)$. The difference between these two pressures constitutes the nett pressure $p(x,t)$. It is convenient to use the complex representation already described and so eqn (1) can be written as

$$(\boldsymbol{D}k_x^4 - m\omega^2)\boldsymbol{W} = \boldsymbol{P}_I - \boldsymbol{P}_T. \tag{2}$$

$\boldsymbol{D}$ is the complex plate stiffness given by $\boldsymbol{D} = D(1 + j\eta)$ where η is the flexural loss factor. The wave impedance on the lower or input side of the plate, $\boldsymbol{z}_I = \boldsymbol{P}_I/j\omega\boldsymbol{W}$, can be written in terms of the wave impedance on the upper or terminating side of the plate, $\boldsymbol{z}_T = \boldsymbol{P}_T/j\omega\boldsymbol{W}$, by rearranging eqn (2) to give

$$\boldsymbol{z}_I = \boldsymbol{z}_T + j(\omega m - k_x^4\boldsymbol{D}/\omega). \tag{3}$$

Equation (3) is the impedance formula for an impervious barrier. Equations (2) and (3) can be manipulated to give the pressure formula for an impervious barrier

$$\boldsymbol{P}_T = \boldsymbol{P}_I[\boldsymbol{z}_T/\boldsymbol{z}_I]. \tag{4}$$

4.2 TRANSMISSION LINE FORMULAE FOR A POROUS LAYER OR AN AIR SPACE

The simplest model which can be used to consider the propagation of acoustic waves in a porous material assumes that the structure of the porous material is rigid, thermally non-conductive and occupies a neglegible fraction of the total volume. Acoustically, such a material can be completely characterised by its flow resistivity R_1. The propagation of acoustic waves in such a material is governed by eqn (5), which is written in terms of the acoustic pressure $p(x, y, z, t)$. More elaborate models such as those described by Cremer and Müller (1982) could be used if desired.

$$\frac{\partial^2 p}{\partial x^2} + \frac{\partial^2 p}{\partial y^2} + \frac{\partial^2 p}{\partial z^2} = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} + \frac{R_1}{\rho c^2}\frac{\partial p}{\partial t} \tag{5}$$

ρ is the density of the gas and c is the velocity of sound in it. The relationship between the pressure gradient in the x direction and the particle velocity $u(x, y, z, t)$ in that direction is

$$-\frac{\partial p}{\partial x} = \rho\frac{\partial u}{\partial t} + R_1 u. \tag{6}$$

Consider now a plane wave which is travelling in the x direction. Equation (5) becomes simpler due to the absence of the y and z variables. It can be readily shown from this simplified equation that the complex representation of the acoustic pressure associated with a plane sinusoidal wave which is travelling in the positive x direction is $\boldsymbol{P}_+ \exp[j(\omega t - \boldsymbol{k}_x x)]$ where the complex wave number $\boldsymbol{k}_x = k(1-jR_1/\rho\omega)^{1/2}$. The imaginary part of $\boldsymbol{k}_x$ defines the spatial decay rate of the wave. The complex representation of the particle velocity associated with this wave can be found by use of eqn (6). The characteristic impedance $\boldsymbol{Z}_0$, the complex ratio of the complex representations of the acoustic pressure and the particle velocity, can then be shown to be given by $\rho c(1 - jR_1/\rho\omega)^{1/2}$. Consider now the layer shown in Fig. 4. The wave propagation direction is in the x-z plane. Equation (5) is then independent of y. Sinusoidal waves are propagating without decay in the positive x direction and the wave number is k_x. The complex representations of these waves are $\boldsymbol{P}_+ \exp[j(\omega t - k_x x - \boldsymbol{k}_z z)]$ and $\boldsymbol{P}_- \exp[j(\omega t - k_x x + \boldsymbol{k}_z z)]$. The first of these expressions represents a wave which is travelling in the positive x and positive z directions while the second expression represents a wave which is travelling in the positive x and negative z directions. Substitution of either of these expressions into eqn (5) shows that k_x and $\boldsymbol{k}_z$ must be related to the complex wave number $\boldsymbol{k}$ by eqn (7).

$$k_x^2 + \boldsymbol{k}_z^2 = \boldsymbol{k}^2 \tag{7}$$

The analogous form of eqn (6) enables the particle velocity in the z direction associated with each wave to be found. The acoustic pressure and z direction particle velocity at any point

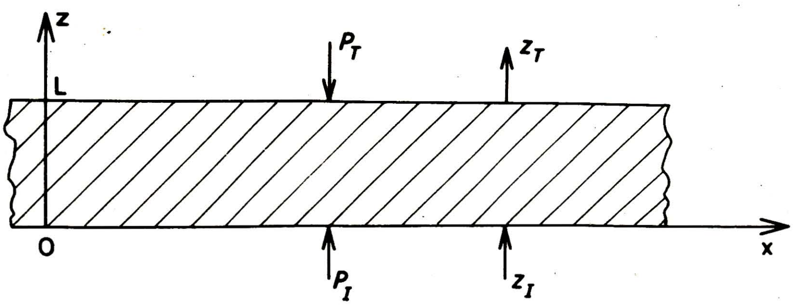


Figure 4: Porous Layer or Air Space Model

can then be found by superposition of the appropriate waves. Thus the acoustic pressures and z direction particle velocities on the surfaces $z = 0$ and $z = L$ can be found in terms of $\boldsymbol{P}_+$ and $\boldsymbol{P}_-$. These expressions can be manipulated to give the wave impedance on the lower or input side of the layer $\boldsymbol{z}_I$, in terms of that on the upper or terminating side of the layer $\boldsymbol{z}_T$. Similarly, a formula can be derived relating the acoustic pressures on the input and terminating sides of the layer. The impedance and pressure formulae are

$$\boldsymbol{z}_I = \frac{\boldsymbol{Z}_0\boldsymbol{k}}{\boldsymbol{k}_z}\left[\frac{[1+\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_T\boldsymbol{k}_z]\exp(j\boldsymbol{k}_zL)+[1-\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_T\boldsymbol{k}_z]\exp(-j\boldsymbol{k}_zL)}{[1+\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_T\boldsymbol{k}_z]\exp(j\boldsymbol{k}_zL)-[1-\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_T\boldsymbol{k}_z]\exp(-j\boldsymbol{k}_zL)}\right] \tag{8}$$

and

$$\boldsymbol{P}_T = \frac{\boldsymbol{P}_I}{2}\left[[1+\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_I\boldsymbol{k}_z]\exp(-j\boldsymbol{k}_zL)+[1-\boldsymbol{Z}_0\boldsymbol{k}/\boldsymbol{z}_I\boldsymbol{k}_z]\exp(j\boldsymbol{k}_zL)\right]. \tag{9}$$

If the layer does not contain a porous material, that is, it is an air space, the preceding formulae are used with $R_1 = 0$ and so $\boldsymbol{k} = k$ and $\boldsymbol{Z}_0 = \rho c$.

5. APPLICATION OF THE TRANSMISSION LINE FORMULAE

The procedure described in the previous section enables the interaction between the system and an oblique plane wave, the propagation direction of which is at an angle θ to the normal to the system, to be considered. Application of the wave impedance formulae, eqns (2) and (8) along with the facts that, on Surface 1 the wave impedance is $\rho c/\cos\theta$ and $k_x = k\cos\theta$ in all the elements of the system, enables the wave impedance on Surface N to be found. It can be readily shown that if the incident wave on Surface N has a complex representation of unity, the complex representation of the acoustic pressure on Surface N, $\boldsymbol{P}_N$ and the complex representation of the reflected wave $\boldsymbol{P}_R$, are

$$\boldsymbol{P}_N = (\boldsymbol{\alpha}-1)/(\boldsymbol{\alpha}+1), \tag{10}$$

and

$$\boldsymbol{P}_R = 2\boldsymbol{\alpha}/(\boldsymbol{\alpha}+1), \tag{11}$$

where

$$\boldsymbol{\alpha} = \boldsymbol{z}_N\cos\theta/\rho c. \tag{12}$$

The sound reflection coefficient and hence the sound absorbtion coefficient can be found from $\boldsymbol{P}_R$. By use of the pressure formulae, eqns (4) and (9), the complex representation of the pressure of the transmitted wave can be found. The sound transmission coefficient and hence the sound reduction index for the system then can be determined. These frequency and angle dependent coefficients $\tau(f,\theta)$, can be used to derive the corresponding coefficients when a particular distribution of the incident sound field is assumed. It is common, for example,

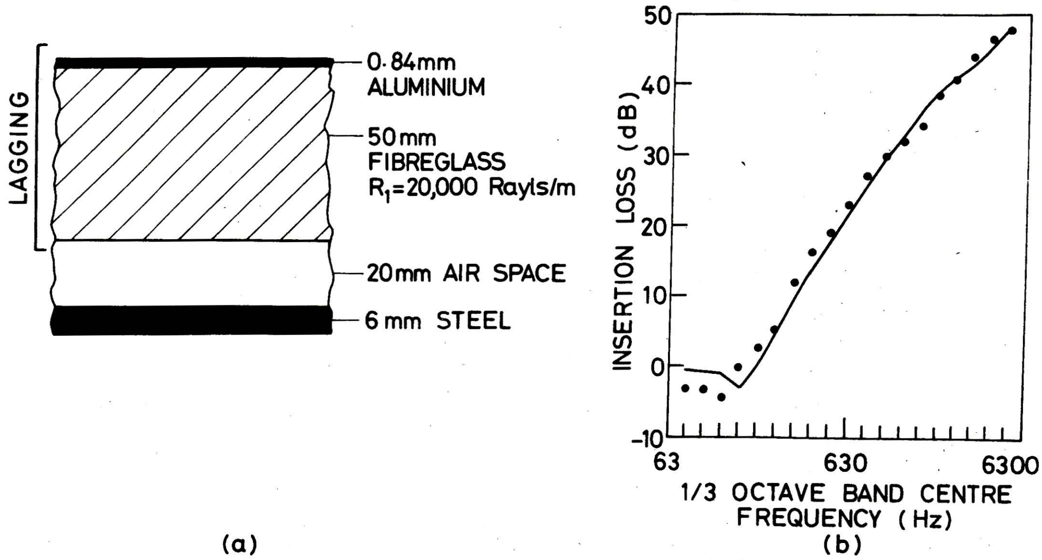


Figure 5: (a) System and (b) Predicted (—) and Measured (•) One-Third Octave Band Insertion Losses

to assume that the incident field is diffuse. The corresponding diffuse field coefficients $\tau(f)$ can then be found from the well known result

$$\tau(f) = \int_0^{\pi/2} \tau(f,\theta)\sin 2\theta\, d\theta. \tag{13}$$

The band average quantities $\tau(\Delta f)$ can be readily found from $\tau(f)$.

6. EXAMPLES OF APPLICATION OF THE PROCEDURE

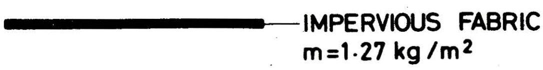
Two examples which illustrate how the previously described procedure can be used are given in this section. The results predicted by the procedure are compared with measured results. The first example relates to calculation of one-third octave band insertion losses produced by a simple industrial acoustic lagging and the second example relates to the calculation of the one-third octave band diffuse field sound absorption coefficients of an architectural fabric. Both examples involve first computing, for a number of values of f and θ, $\tau(f,\theta)$ by use of the transmission line formulae. These values are then numerically integrated to give $\tau(f)$ and then $\tau(\Delta f)$. Here ten values of f were used in each one-third octave band and the computations were made at 50 values of θ in the range 0 to $\pi/2$. Thus although considerable computation is required, the coding is simple, particularly if a language which incorporates complex variables is used as the coding of the transmission line formulae is facilitated.

6.1 THE INSERTION LOSS OF A LAGGING

The system is shown in Fig. 5(a). The sound field incident on the lagged plate is assumed to be diffuse. The insertion loss is calculated by first finding the one-third octave band sound reduction indices for the system with and without the lagging applied. The one-third octave band insertion losses can then be found. The predicted results are shown in Fig. 5(b) along with results measured by Au and Byrne (1987). It can be seen that the agreement is excellent.

6.2 THE ABSORPTION COEFFICIENT OF A FABRIC SHEET

The system is shown in Fig. 6(a). The complex representation of the reflected wave can be found from eqns (11) and (12) and so only the wave impedance on the incident surface of the sheet need be found. This can be done simply by use of eqn (3). The fabric sheet is assumed to be limp and so the flexural stiffness $D = 0$. The sound reflection coefficient can be found from the complex representation of the reflected wave and so the sound absorption coefficient can be determined. The predicted results are shown in Fig. 6(b) along with measured values given by Wyerman (1981). The measured results were obtained from octave band reverberation time measurements in a fabric roofed sports arena. The agreement between predicted and measured results is seen to be excellent.



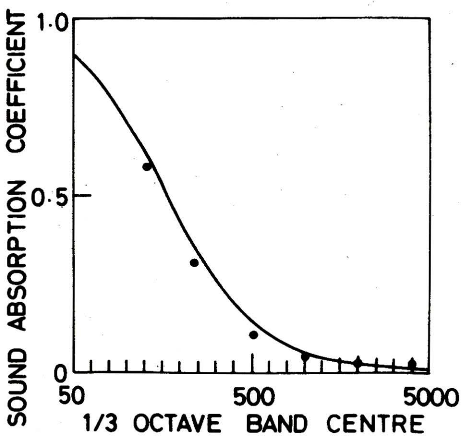


Figure 6: (a) System and (b) Predicted One-Third Octave Band (—) and Measured Octave Band (•) Diffuse Field Sound Absorbtion Coefficients

7. CONCLUDING REMARKS

A method for computing the acoustic properties of nominally uniform plane constructions has been described in this paper. The constructions are considered to be composed of discrete elements such as impervious barriers, porous layers and air spaces The computation procedure is based on formulae which describe the acoustic performance of each of these element types. The good agreement between the predicted and measured results for the two examples indicates the value of the procedure.

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Acoustics in the Eighties

Sessions 3A and 4A Workshop

Acoustics in the Eighties

SCENARIOS FOR AUSTRALIAN ACOUSTICS IN THE 1990s

SCENARIOS FOR AUSTRALIAN ACOUSTICS IN THE 1990s

ABSTRACT

The trend in all branches of Science and Education is towards more accountability. This tends to mean more applied research and greater attention to the needs of industry and commerce. As part of this shift in emphasis ASTEC has set up a committee to look at acoustics research in Australia. The findings of this committee are likely to have far reaching consequences on Australian Acoustics. The workshop has been organised to inform the Australian Acoustics Community about the ASTEC Committee's findings, to discuss the consequences of them and to look at ways of improving co-operation between individuals and organisations involved in acoustics research, teaching and testing. Several invited speakers, including the Chairman of the ASTEC Committee, Professor Roger Tanner, Dr Neville Fletcher, Director of the CSIRO Institute of Physical Sciences, Marshall Smither, Director of the National Acoustic Laboratories, Dr Bob Hooker of the University of Queensland and Melbourne Consultant Jim Watson will participate in the Workshop.

Acoustics in the Eighties

Session 3B: Measurement Technology I

Acoustics in the Eighties

A DIVER OPERATED ACOUSTIC SURVEYING SYSTEM

A.J. Duncan, J.D. Penrose, J. Green

A DIVER OPERATED ACOUSTIC SURVEYING SYSTEM

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ABSTRACT

This paper describes a wreck site surveying system developed for the Western Australian Maritime Museum. The system is diver operated and uses acoustic distance measurements to determine the location of points of interest on a wreck site relative to acoustic transponders located at known points on the sea bed. Measured distances are stored in semiconductor memory together with pressure measurements which are used to determine the vertical coordinates of the points of interest. Initial data processing by a microprocessor in the diver unit includes evaluation and averaging of acoustic transit times and weighted averaging of pressure signals to offset wave effects. All data are transferred to a small ship-board computer for processing and permanent storage when the diver surfaces. The system has been used in a major field program in the Abrolhos Islands off the coast of Western Australia, yielding an extensive set of results.

ACKNOWLEDGEMENT: The development of this system was funded by a grant from the Australian Research Grants Committee.

1. INTRODUCTION

In recent years the potential of shipwrecks to provide a valuable insight into the lives of our forebears has begun to be realised. Legislation has been passed in Australia to protect wreck sites of historic importance and maritime archaeology departments have been set up in most States to carry out systematic studies of important sites.

Once a wreck site has been located and its position accurately charted one of the first tasks of the maritime archaeologist is to carry out a pre-disturbance survey of the site. The object of this survey is to accurately map the relative positions of all exposed artefacts and parts of the wreck itself. Once excavation has commenced it is necessary to have some means of recording the locations of artefacts as they are found so that as accurate as possible a picture of the original ship and its contents may be built up.

Traditionally the surveys are carried out using techniques adapted from land surveying, utilising underwater theodolites and magnetic compasses for angle measurement, and tape measures for distance measurement. Photogrammetry has also been applied with considerable success to relatively small scale surveys. Unfortunately these methods are of limited usefulness when the visibility on a site is poor, and it is not uncommon for divers to be working in visibilities of less than 1 m. Under these conditions optical surveying methods must be abandoned and it is necessary to rely on tape measured distances to obtain the necessary information. In poor visibility the problems associated with using tape measures underwater (snagging of the tape, errors due to bottom currents and problems associated with co-ordinating the actions of two divers who are out of sight of one another) make such surveys very time consuming and prone to error.

In an attempt to improve the efficiency and accuracy of underwater surveying, a surveying system has been developed which uses acoustic distance measurements to obtain horizontal positions, and pressure measurements to obtain vertical positions. Electronic logging of the measured values relieves the diver of the task of recording measurements manually and removes the possibility of transcription errors.

The design parameters for the system were that it should give position measurements to an accuracy of +/- 10 mm in three dimensions, that it should operate over distances of up to 50 metres, and that it should be as simple as possible for the diver to operate.

2. SYSTEM DESCRIPTION

2.1 PRINCIPLE OF OPERATION

Fig. 1 shows the configuration of a typical system deployment.

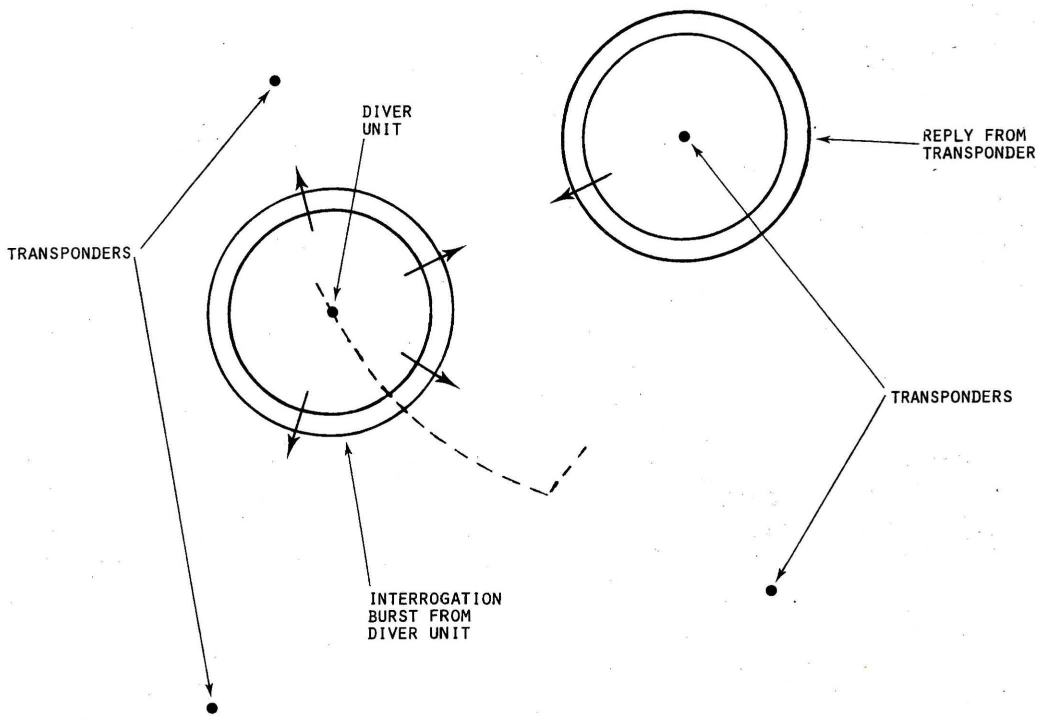

FIGURE 1.
Plan view of survey site for typical system deployment.

Two or more (to a maximum of eight) transponders are located at known positions around the survey site. The diver locates a measurement probe containing an acoustic transducer and a pressure transducer on the point to be surveyed. The probe is connected by cable to an electronics unit carried by the diver, which controls the measurement sequence and stores the results. The diver initiates a measurement sequence by depressing a switch on the electronics unit. A coded acoustic burst is sent to each transponder in turn, which then replies with another coded burst. Given an accurate knowledge of the sound velocity, the time between transmitting the interrogation burst and receiving a reply yields the distance to each transponder. A number of pressure readings are made and averaged to provide depth information, the results are stored in semiconductor memory, and then the diver moves on to the next location where the process is repeated. More than 1600 position measurements can be made before it is necessary to download the data to a computer or terminal via a standard RS232 interface.

2.2 ACOUSTIC CONSIDERATIONS

Sound Velocity

The speed of sound in sea water is about 1500 m/sec. The actual value depends on temperature, salinity and depth, and to a good approximation can be calculated from the following formula due to Medwin (1975):

$$c = 1449.2 + 4.6\ T - 0.055\ T^2 + 0.00029\ T^3 + (1.34 - 0.010\ T)(S - 35) + 0.016\ D$$

Where:

c is the sound speed (m/sec)

T is the temperature (C)

S is the salinity (parts per thousand)

D is the depth (m)

Taking first differentials and evaluating for typical values of T = 15 degrees, S = 35 ppt and D = 10 metres yields the following values for the variation of sound velocity with temperature, salinity and depth:

$$\partial c/\partial T = 3.2 \text{ m/sec per degree C}$$

$$\partial c/\partial S = 1.2 \text{ m/sec per ppt}$$

$$\partial c/\partial D = 0.016 \text{ m/sec per metre}$$

The desired measurement accuracy of +/- 10 mm in 50 metres necessitates a knowledge of the sound velocity to +/- 0.02 % or +/- 0.3 m/sec. If the sound velocity is to be inferred from measurements of temperature and salinity this requires a knowledge of temperature to better than +/- 0.1 degrees and of salinity to better than +/- 0.25 ppt.

These are rather stringent requirements, and a more practical way of obtaining the sound velocity is to use redundant distance measurements, in other words, to use more than the bare minimum of two transponders, and to use a least squares adjustment procedure to find the best-fit sound velocity. This procedure assumes that the sound velocity is constant over the survey site, which is likely to be a reasonable assumption provided that the site is essentially horizontal. Velocity changes in the vertical are generally more significant than those in the horizontal, especially when the medium is stratified. In the present work, transponder separations and hence coordinates in a local reference frame are established by the acoustic/pressure system so that absolute length scales depend on at least one temperature and salinity measurement sequence, or alternatively on one tape-measured distance.

Choice of Frequency

The desired distance measurement accuracy of +/- 10 mm puts a lower limit on the the acoustic frequency adopted, as the uncertainty in the distance measurement will be of the order of a wavelength. Attenuation increases with frequency and thus provides an upper limit for the frequency. A frequency of 300 kHz was chosen for the system, yielding a wavelength of 5 mm, and a logarithmic absorption coefficient, calculated using the formula of Fisher and Simmons (1977), of 0.081 dB/m at 15 degrees C.

Coding

The underwater environment is highly reflective to sound, and it is the interference of multipath signals reflected off the sea surface, the sea bed, and any other objects present (including divers) that provides the main source of signal degradation. The adopted coding technique must therefore be as insensitive as possible to variations in signal amplitude caused by multipath interference, which rules out any form of amplitude modulation. Pulse-width and pulse-length modulation are also susceptible to multipath interference, and so some form of frequency modulation is commonly used for transmitting information underwater.

The digital form of frequency modulation, known as frequency shift keying (FSK), was adopted for this system. The transmitted burst is illustrated in Figure 2.

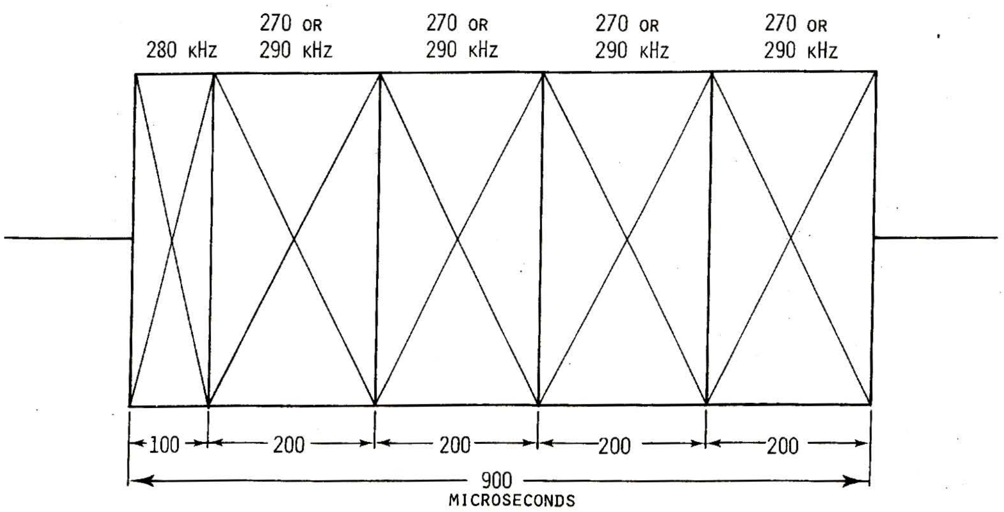

FIGURE 2.
Acoustic burst format.

The initial 100 microsecond burst at 310 kHz is used to trigger the timing circuits, after which there are four one-bit periods, each of 200 microseconds duration, and during each of which the transmitted frequency may be either 290 kHz (logic 0) or 330 kHz (logic 1). A total of sixteen unique codes are thus possible, and as each transponder has a different receive and transmit code, this makes it possible to use up to eight transponders simultaneously.

2.3 ELECTRONICS

Transponder

A block diagram of the transponder electronics is given in Figure 3. Sound energy is converted into an electrical signal by a cylindrical piezo-ceramic transducer. The signal is amplified and detected, and its code is compared to the expected code by the receiver logic circuit. If a match is found, the transmitter logic circuit generates a coded reply burst, which is amplified by the power amp and transmitted by the transducer. Power is provided by eight D size NiCd batteries, sufficient to allow the transponders to operate continuously in listening mode for approximately one week.

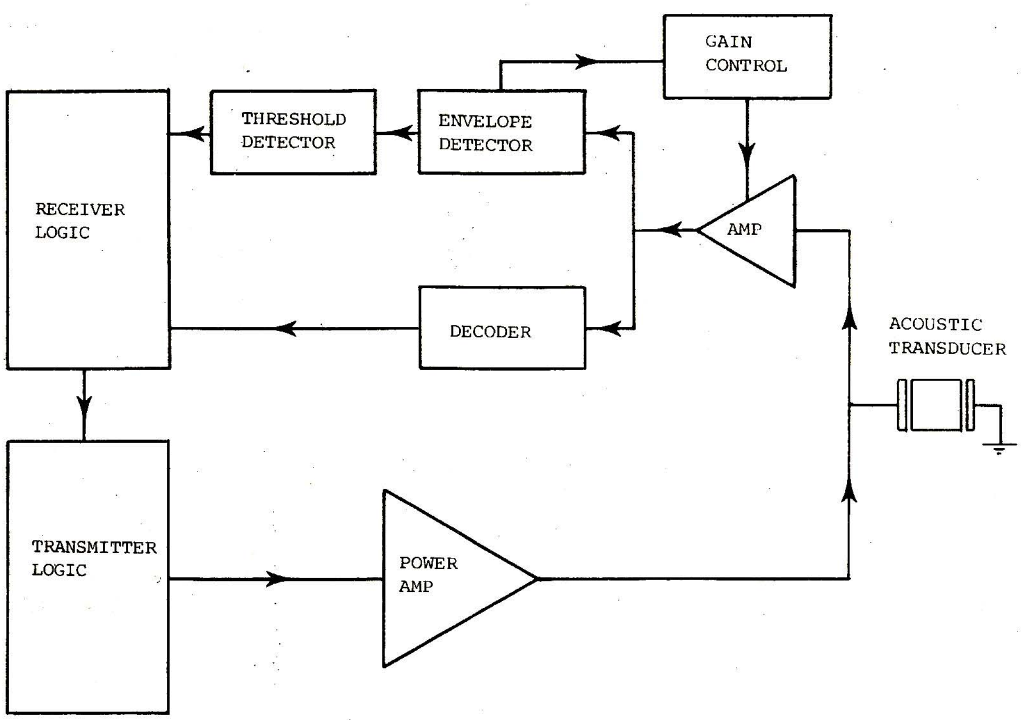

FIGURE 3.
Transponder electronics block diagram.

Diver Unit

The diver unit electronics is shown in outline in Figure 4. The receiver and transmitter circuits are identical to those in the transponder, with the exceptions that the transmitter is triggered by the microprocessor, the receiver signals the microprocessor when a correctly coded burst has been received, and the transmit and receive codes are set up by the microprocessor rather than by switches.

The NSC800 microprocessor controls the operation of the system, executing a program stored in read only memory (ROM), and storing the results in 56k bytes of random access memory (RAM). An interface to the diver is provided by way of a single magnetic reed switch, a 4 digit liquid crystal display and nine light emmitting diodes (LEDs). The output of the pressure transducer is measured by an analogue to digital converter and there is an RS232 interface for data read-out and for setting up system parameters. Power is provided by twelve D size rechargeable NiCd batteries.

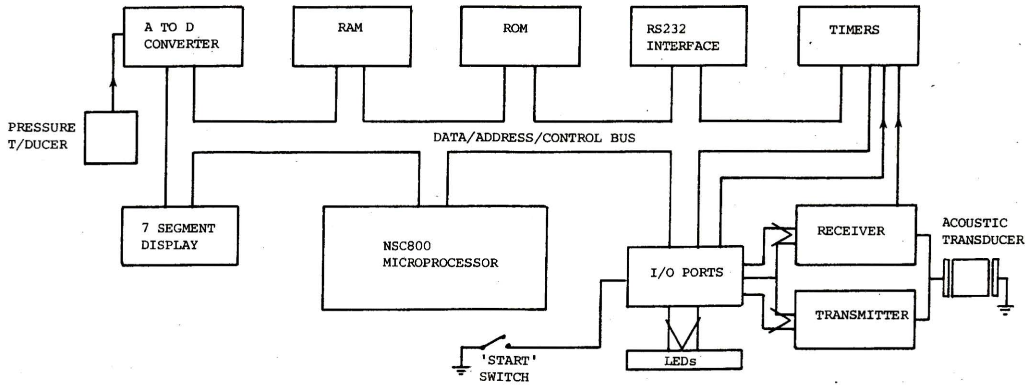

FIGURE 4.
Diver unit electronics block diagram.

2.4 MECHANICAL ARRANGEMENT

Transponder

The transponder housings are made from 90 mm o.d. PVC pipe with end-caps machined from solid PVC and sealed with double O-ring piston seals. A magnetic reed switch in the base of the unit allows the power to be switched off underwater, and a screw plug in the base provides easy access to a connector for battery charging. Protection for the transducer, mounted on top of the housing, is provided by thin stainless steel guards, and by a screw-on cap which can be removed when the transponder is in use. The overall length of the transponder is 790 mm.

Diver Unit

The transducer probe contains the pressure and acoustic transducers and has an overall length of 320 mm. It has a built-in spirit level and has been designed so that it can be set up vertically on almost any terrain. This is necessary to ensure that the centre of the acoustic transducer is directly above the point to be measured.

The housing for the electronics unit is made from welded marine grade aluminium alloy and incorporates an easily removable rear port which gives access to a power switch, a connector for the RS232 interface, and a connector for battery charging. The display is visible through a perspex view-port in the top of the housing. The overall dimensions of the housing are: height 120 mm, width 270 mm and depth 370 mm.

3. FIELD TESTING

The first full scale field test of the acoustic surveying system was carried out in October 1986 in the Abrolhos Islands, 60 nautical miles northwest of Geraldton, Western Australia. The time and place were chosen to take advantage of a maritime archaeology student field trip and the consequent availability of divers and logistics support. Six transponders and a wire-wrap prototype of the diver electronics unit were assembled for the test.

Four transponders were deployed at the corners of a forty metre square in a sheltered location within easy reach of the base camp. Transponder separations were measured using three techniques:

(i) By direct tape measurements. The site was chosen to allow tape measures to be utilised conveniently. In particular the site was protected from any wave or current influences.

(ii) By placing the transducer probe on top of each transponder in turn and measuring the distances to every other transponder acoustically.

(iii) By moving the transducer probe across the line joining each pair of transponders and taking the minimum sum of the acoustic ranges measured to the two transponders as the inter-transponder distance. This technique is conventionally known as baseline crossing.

All measured distances were corrected to a horizontal plane by using the output of the pressure transducer to determine the vertical co-ordinates of the transponders. It was found necessary to adjust one of the tape-measured distances by 1.000 m to make the set of tape-measured transponder separations self-consistent.

The root mean square (RMS) difference between the tape-measured distances (after adjustment) and the distances obtained from direct acoustic measurements was 120 mm, and the RMS difference between the tape-measured distances and the baseline crossing distances was 64 mm. This in itself does not indicate which technique is the most accurate as the tape-measured distances themselves could be in error. Each set of measurements was used in a least squares adjustment program to calculate the transponder co-ordinates. The program calculates the residual for each distance measurement, in other words, the difference between the measured distance between two transponders and the distance between the calculated transponder co-ordinates. Small residuals indicate accurate distance measurements. The RMS residuals for the three different measurement techniques were: 7.5 mm for the tape-measured distances, 37 mm for the direct acoustic measurements and 23 mm for the base-line crossing measurements.

4. CONCLUSIONS

The results of the Abrolhos Islands field test indicate that the desired measurement accuracy for the acoustic system is being approached but has not yet been obtained. Since the field test, effort has centred on building a printed circuit board version of the diver electronics unit together with a purpose designed housing. The housing design in particular has benefited greatly from the practical experience gained in using the equipment in the field. A number of modifications have been made to the electronics to improve measurement accuracy and reliability and several static trials have been undertaken in Fremantle Fishing-boat Harbour. A full scale field test of the new equipment will proceed in the near future.

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Acoustics in the Eighties

COMPARISONS OF VARIOUS METHODS FOR DETERMINATION OF SOUND POWER LEVELS OF NOISE SOURCES

J.C.S. Lai and A. Dombek

COMPARISONS OF VARIOUS METHODS FOR DETERMINATION OF SOUND POWER LEVELS OF NOISE SOURCES

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ABSTRACT

Accurate determination of sound power levels for noise sources is important in prediction of installation noise levels, comparison of noise emissions between different machines and in devising noise control strategies. Three different methods have been used to measure the sound power of a noise source. The first method involves sound pressure measurements inside an anechoic chamber built to ISO standard while the second method employs a known reference sound power source to establish the sound power of the noise source. The third method applies the sound intensity technique to determine the sound power and sound intensity mappings will also be presented. Comparisons of the results obtained by these three methods will be made and the results discussed. Conclusions regarding the limitations and applicability of each method will be drawn.

1. INTRODUCTION

A basic measure of the acoustical output of a sound source operating in a particular medium is its sound power W which is independent of the acoustic environment. It is defined by eqn (1) as follows:

$$W = \int_S \tilde{I} \cdot d\tilde{S} = \int_S I_r \, dS = \int_S \overline{pu_r} \, dS \qquad (1)$$

where $\tilde{I}$ is the intensity vector (rate of energy flow per unit area)
S is a hypothetical surface enclosing the source
I_r is the component of the intensity vector normal to S
p is the sound pressure
u_r is the particle velocity normal to S
and overbar denotes time-average.

On the other hand, sound pressure, which is a scalar quantity often measured, gives an indication of the human response to noise. Unlike sound power, the level of sound pressure measured for a given sound source is dependent on the acoustic environment such as the amount of absorption and the characteristics of other noise sources existing in the operating environment. Hence it is important to be able to determine the sound power of a source as this data can be used to predict the sound pressure level produced by the source in different operating environments or to compare its noise-producing capacity with other sources. There are various methods in determining the sound power of a noise source and these methods can basically be classified into two types, namely, indirect and direct methods.

In the indirect methods, the sound power of a noise source is determined by measuring the sound pressure level in some particular acoustic environment such as in anechoic or reverberant rooms. The international standards ISO 3741-3746 cover methods in this category. With the advent of digital signal processing technique, the sound power of a source can now be determined directly by measuring the component of sound intensity normal to a given hypothetical control surface area enclosing the source. However, as the technique is relatively new, there is not yet an international standard that deals with measurements of this type.

The objectives of this paper were to study two indirect methods for determining sound power, namely, through measuring sound pressure level in an anechoic room and comparison of sound pressure level with a known sound power source, and the direct method of measuring sound intensity. In particular, the advantages and disadvantages of each method will be discussed.

2. INDIRECT METHODS

2.1 DETERMINATION OF SOUND POWER IN AN ANECHOIC ROOM

The sound power of a source can be determined by measuring the averaged sound pressure level over a hypothetical closed surface in an anechoic room since the intensity I is related to the root-mean-square pressure p in a free field by eqn (2) so that the sound intensity level L_I is equal to the sound pressure level L_p.

$$I^2 = p^2/\rho c \qquad (2)$$

where ρc is the acoustic impedance of the medium

The anechoic room at the Australian Defence Force Academy is a double-shelled construction. The outer shell consists of 190 mm thick reinforced concrete walls and the inner shell is a 106 mm thick Sonex-N reinforced enclosure lined with fibreglass slabs which are 540 mm deep with a taper angle of 14° and have been installed in staggered layers. The resulting free-field dimensions of the room are 3.5m x 3.5m x 3.5m and the lower cut-off frequency, in compliance with ISO 3745 standard, is 150 Hz.

Two different sources were used in this experiment, namely, a Bruel & Kjaer (hereinafter referred to as B&K) 4205 reference sound power source and a small Bosch hand drill. The B&K 4205 sound power source was used to produce octave band noises (from 125 Hz to 8 kHz) and wide band noise (covering 100 Hz to 10 kHz). Sound pressure level measurements in octave bands were performed over a hypothetical hemisphere of radius 1 m using a B&K 2231 modular sound level meter (with filter set 1625) inside the anechoic room with the sound power source placed on a reflecting plane made of 3.5mm thick aluminium sheet. The sound power level was set to 80 dB for all the octave band noises and for the wide band noise. Measurements were made for the wide band noise first with ten microphone positions (hereafter referred to as 10-point measurements) and then with 20 microphone positions (hereafter referred to as 20-point measurements) as shown schematically in Fig.1 (a) and (b) respectively. The sound power was calculated from eqn (3) according to ISO 3745 and and the sound power spectrum for the wide band noise is displayed in Fig. 2.

$$L_W = \overline{L_p} + 10 \log_{10}(2\pi r^2) \tag{3}$$

where $\overline{L_p} = 10 \log_{10} \frac{1}{N} \sum_{i=1}^{N} 10^{0.1 L_{pi}}$

$\overline{L_p}$ is the surface sound pressure level in dB

L_{pi} is the band pressure level resulting from the i^{th} measurement

and N is the number of measurements

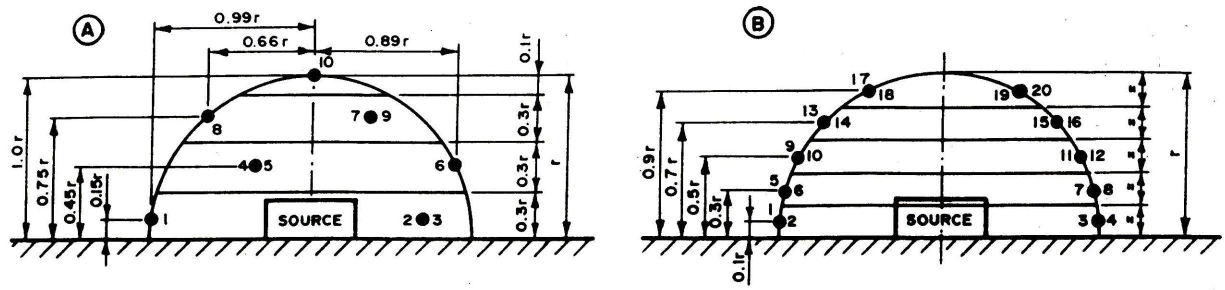

Figure 1. Microphone positions for sound power measurements in a free field over a reflecting plane.

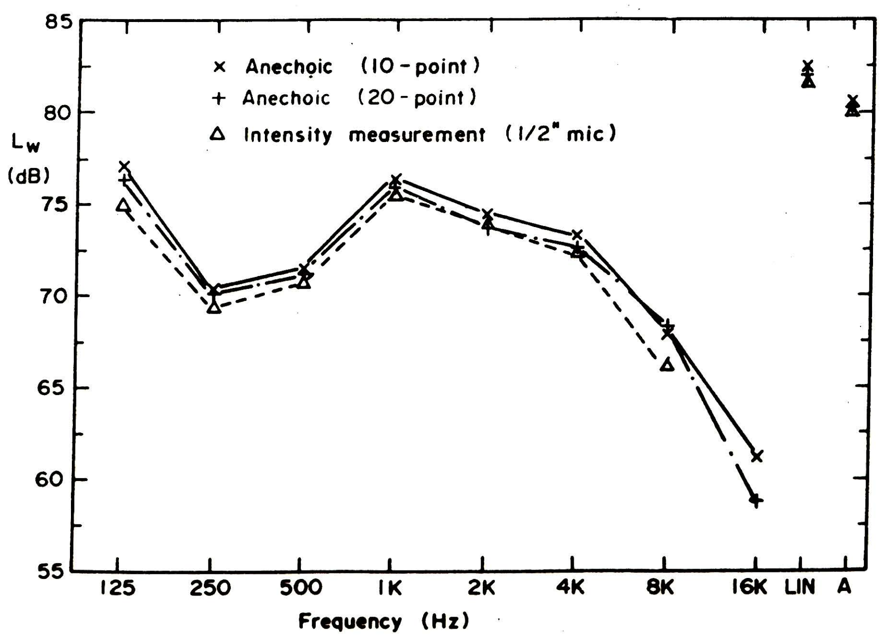

Figure 2. Comparisons of sound power spectra obtained by anechoic and sound intensity measurements for reference sound source.

The uncertainty in determining sound power levels of sound sources over a reflecting plane in a free field is ± 1.5 dB for 125 to 500 Hz, ± 1dB for 800 to 5000 Hz and ± 1.5 dB for 6300 to 10000 Hz. It must be pointed out that for the 10-point measurements, the difference in decibels between the highest and lowest sound pressure levels measured in some octave bands exceeds the criterion (in this case 5 dB) set by the ISO 3745 standard, thereby overestimating the sound power for the source. The 20-point measurements yield a A-weighted sound power level to within 0.5dB of the set level. Furthermore, it agrees with the value obtained by A-weighting the sound power spectrum in Fig. 2 to within 0.2 dB.

Fig. 3 shows the deviation ΔL_w from the set sound power level of 80 dB for the octave band noises and wide band noise. All the measurements in Fig.3 were made with the 20-point arrangements. Also indicated in Fig. 3 are the limits of accuracy of the sound power level of the B&K 4205 source. The measured deviation ΔL_w is within ± 0.5 dB, well within the accuracy limits, thus establishing the use of the 4205 as a reference sound power source for determining the sound power of other sources using the justaposition method described in section 2.2.

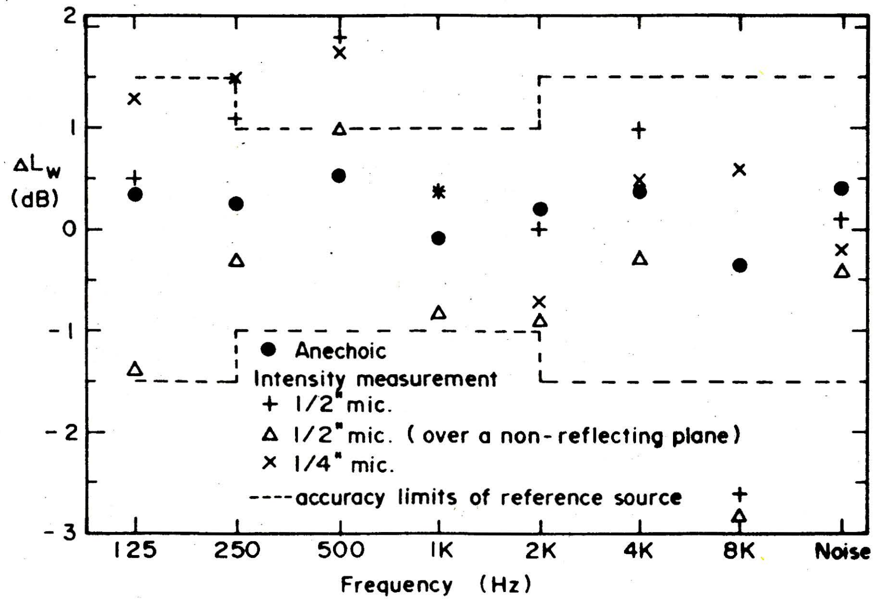

Figure 3. Deviation ΔL_w of measured sound power from actual value for reference octave band and wide band noise.

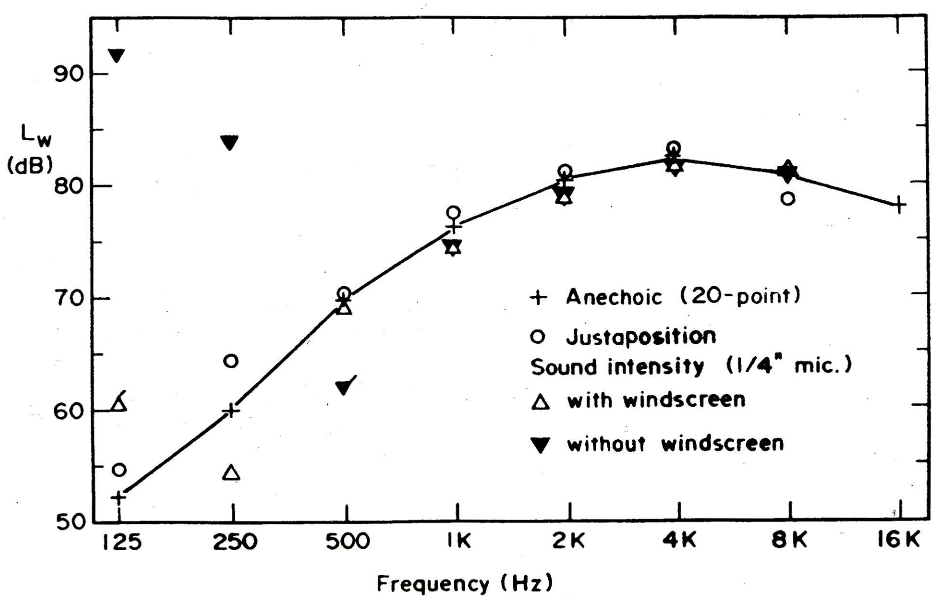

Figure 4. Comparisons of sound power spectra obtained by anechoic, justaposition and sound intensity measurements for a small hand drill.

The sound power spectrum for the drill obtained by 20-point measurements over a reflecting plane in the anechoic room is shown in Fig. 4.

2.2 DETERMINATION OF SOUND POWER USING JUSTAPOSITION PRINCIPLE WITH A REFERENCE SOUND POWER SOURCE

In this method, the sound power level L_w of a noise source is determined by the measuring the sound pressure levels produced by the noise source and by the reference source using eqn (4).

$$L_w = L_{wr} + L_p - L_{pr} \qquad (4)$$

where L_w = sound power level of noise source under test
L_{wr} = sound power level of reference source
L_p = sound pressure level due to noise source
L_{pr} = sound pressure level due to reference source.

The noise source was the Bosch hand drill and the B&K 4205 reference source was placed beside the hand drill in the anechoic room on an aluminium sheet. Four microphone positions spaced 90^o apart at a radius of 1m from the drill/4205 and at an elevation of 0.9 m from the reflecting plane were used. At each position, with the drill switched on, the sound pressure level in octave bands was measured with the B&K 2231 sound level meter and the 1625 filter set ; the 4205 reference source was then switched on and the sound power output from the 4205 was increased until the sound level meter reading was 3 dB higher than that due to the drill alone. The sound power output from the machine is then equal to that from the 4205. The sound power spectrum of the drill obtained by averaging the results over the four microphone positions is shown in Fig. 4 and the A-weighted sound power level, as shown in Table I, agrees to within 0.2 dB of that obtained according to ISO 3745 standard.

Table I Comparison of sound power levels for drill (in dB)

	Anechoic	Justaposition		Intensity	
		Averaged	One position	With windscreen	Without windscreen
Linear	86.95	86.24	83.91	85.7	93
A-weighted	86.98	86.79	84.82	86.1	86.5

3. DIRECT METHOD

3.1 SOUND INTENSITY TECHNIQUE

Although the first device for sound intensity measurements was patented by Olson in 1932, it was only until 1977 when Fahy (1977) and Chung (1978) applied digital signal processing techniques to sound intensity theory that commercial devices became available. In order to measure sound intensity directly, the sound pressure p and particle velocity u_r must be measured simultaneously. In the absence of a mean flow in the medium, the particle velocity u_r in direction r is related to the pressure gradient $\partial p/\partial r$ by Euler's eqn (5).

$$\rho\, \partial u_r/\partial t = - \partial p/\partial r \qquad (5)$$

where ρ is the density in the medium
and t is time.

Eqn (5) can be approximated by measuring the pressures, p_A and p_B detected by two microphones closely spaced at a distance Δr along the direction r as given in eqn (6).

$$u_r = -\frac{1}{\rho \Delta r} \int (p_B - p_A)\, dt \tag{6}$$

It has been shown (see, for example, by Gade 1982) that the intensity L_I is related to the cross-spectrum G_{AB} between the two microphone signals or to the phase gradient $\partial\Phi/\partial r$ of the sound field as follows:

$$I_r = -\frac{1}{\omega \rho \Delta r} \operatorname{Im} G_{AB} = -\frac{p^2}{\rho c k} \frac{\partial \Phi}{\partial r} \tag{7}$$

where ω is frequency in radians/sec.
and k is the wavenumber.

A practical implementation of the sound intensity theory for measurements as given in eqn (7) would consist of a dual channel signal analyser and a sound intensity probe comprising of two closely spaced microphones. Since the intensity is related to the phase gradient of the sound field, the two microphones have to be closely matched in phase in order to provide reasonable accuracy for low frequency measurements. Possible sources of error in applying the sound intensity technique have been given by Gade (1985). Basically, they are related to the phase-mismatch of the microphones which imposes a limit on low frequency measurements and the finite difference approximations in eqn (6) which impose a limit on high frequency measurements.

All the sound intensity measurements reported here were made with the B&K 2032 dual channel analyser and the B&K 3519 sound intensity probe (with two microphones mounted face to face) in a room with a volume of 85 m^3 and a reverberation time of 1.32 s at 500 Hz. For a measurement accuracy of ± 1 dB, the frequency range for a pair of 1/2 inch microphones separated at 12 mm is 125 Hz - 5 kHz while that for a pair of 1/4 inch microphones separated at 6 mm is 250 Hz - 10 kHz. The hypothetical measurement closed surface used was a rectangular parllelopiped with six faces as shown schematically in Fig. 5. However, only faces 1-5 are measurement faces as the bottom surface coincides with the plane on which the noise source is placed. Unless otherwise stated, the intensity probe was swept over each measurement face to obtain a spatial-averaged intensity for that face.

Table II Apparent sound power measured with no source inside measurement volume

Trial	1	2	3
Face 1	75.7	77.4	76.1
Face 2	-84.0	-81.0	-81.4
Face 3	75.7	75.5	75.2
Face 4	78.4	78.6	76.7
Face 5	75.8	73.6	74.8
L_{wm}	-78.4	77.8	71

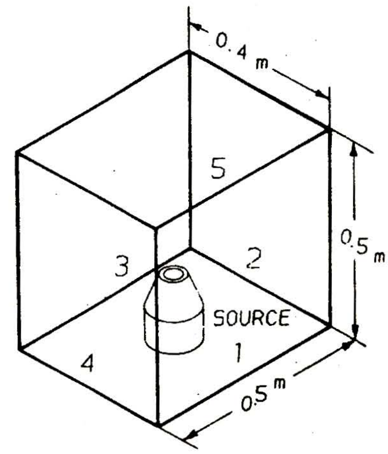

Figure 5. Hypothetical closed surface

3.2 SUPPRESSION OF EXTERNAL NOISE

According to Gauss theorem, if there is no source within a measurement volume, then W as given in eqn (1) is zero. This property offers a significant advantage in using sound intensity technique for sound power determination as external noise will not contribute to the sound power of the noise source under test provided that the external noise is stationary and there is no absorption within the measurement volume. Thus 'in situ' sound power measurements with high accuracy can theoretically be achieved. In order to check the capability of the system in suppressing external noise, a loudspeaker at 0.5 m from measurement face 2 was used to produce a pure tone of 1kHz with a sound power level of 89 dB outside the measurement surface. The results for three different trials with different number of spectrum averages (the minimum being 64) are tabulated in Table II. Contrary to the prediction of Gauss theorem, the apparent power measured L_{wm} is not zero, as has also been reported in literature (for example, Stirnemann et al (1985)). Although the results indicate a suppression of at least 10.6 dB in power for external noise in this particular case, the apparent sound power level measured ranges from -78.4 dB to 77.8 dB, where negative and positive power indicates respectively sound power being absorbed and emitted within the measurement volume. As there is no source within the measurement volume, a positive value for apparent sound power is not a possible physical reality. Based on the results in Table II, a simplified model has been developed as follows to examine the sensitivity of apparent power measured L_{wm} to measurement error associated with intensity measurement at face 2.

Let the power level L_{we} of the external noise source be a dB. Assume the surface area of each measurement face be 0.2 m^2 and the apparent averaged sound intensity level associated with faces 1,2,3,4, and 5 be a, -(a+y), a, (a-x) and a dB respectively. Further assume that y = $(x \pm \varepsilon)$ dB, where ε is the measurement error. Then the external noise suppression in sound power L_{ws} is given by eqn (8).

$$L_{ws} = L_{we} - L_{wm} = (-1)^m \, 10 \log_{10} |f| \qquad (8)$$

$$\text{where} \quad f = 0.6 + 0.2 \left[10^{-0.1x} - 10^{0.1(x \pm \varepsilon)} \right]$$

$$\text{and} \quad m = \begin{cases} 1 & \text{for} \quad f \geq 0 \\ 0 & \text{for} \quad f < 0 \end{cases}$$

A plot of L_{ws} versus ε for x = 5.18876 dB is given in Fig. 6 and clearly indicates the high sensitivity of the measurement for low ε, particularly for ε less than $\pm$ 0.05 dB. Here negative value of L_{ws} implies that the noise source is outside the measurement volume. Note that with the noise source outside the measurement volume, it is possible to record a positive L_{ws} for $\varepsilon < 0$. For ε =0 dB, the suppression L_{ws} is more than 50 dB; however, with $\varepsilon = \pm 0.05$ dB, the suppression has been reduced to just over 20 dB. In most cases, one does not expect an accuracy better than $\pm$ 0.5dB, in which case the suppression is of the order of 10dB, which is compatible with most measurements reported to-date for background noise source close to the measurement volume.

In order to assess systematically the effects of background noise on sound power measurements, the B&K4205 was used as the source under test inside the measurement volume with a loudspeaker at 0.5 m from measurement face 2 producing external noise. Measurements had been made with the B&K 4205 producing 250Hz, 500 Hz, 1 kHz and 2 kHz octave band noise, and wide band noise, each of which was set at three different sound power levels, namely, 65, 75 and 85dB. Corresponding to these noises under test, the loudspeaker was made to produce pure tones of 250 Hz, 500 Hz, 1 kHz and 2 kHz, and white noise respectively, each of which was set at three different power levels, namely, approximately 70, 80 and 90dB. Both sound pressure and intensity measurements were made. The deviation ΔL_w of the measured sound power level from the set level for the B&K4205, has been plotted in Fig. 7 against the difference $L_p - L_I$ between the averaged sound pressure level L_p and sound intensity level L_I. In Fig. 7, flagged symbols are simply negative values plotted in the positive quadrant. Although there is considerable scatter in the data especially for large $(L_p - L_I)$, a curve of best fit subject to the constraint that $\Delta L_w = 0$ for $L_p - L_I = 0$ has been obtained:

$$\Delta L_w = 10 \log_{10} \{0.929 + 10^{0.1[(L_p - L_I) - A]}\} \qquad (9)$$

where A = 11.49 dB

Here A can be interpreted as the external noise sound power suppression level. Considering an accuracy of $\pm$ 1 dB in setting the sound power level with the B&K4205, the data in Fig. 7 indicates that for measurements with an unknown source using this system, the sound power measured would be accurate to within $\pm$ 1 dB for $L_p - L_I$ less than 7 dB. It must be pointed out that an attempt to use eqn (9) as an universal correction curve could be erroneous; it merely indicates the range of $L_p - L_I$ for which one can establish the confidence limits for a particular measurement. This is particularly obvious as the scatter at large $L_p - L_I$ is considerable, the reason being that the external source is dominant; hence the same comments for Fig. 6 applies here.

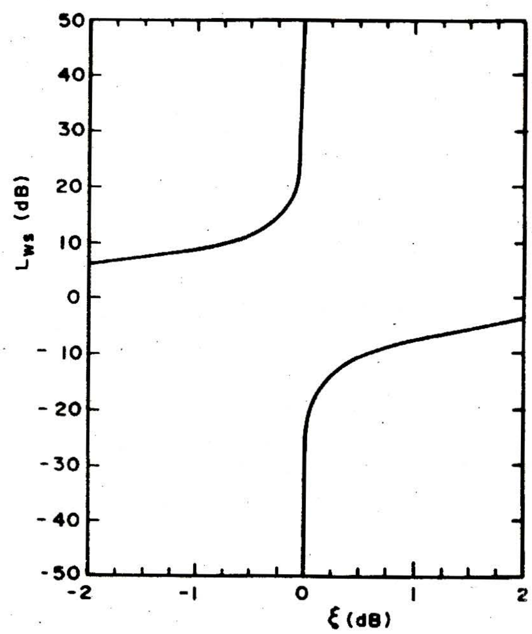

Figure 6. Variation of sound power suppression L_{ws} with ε

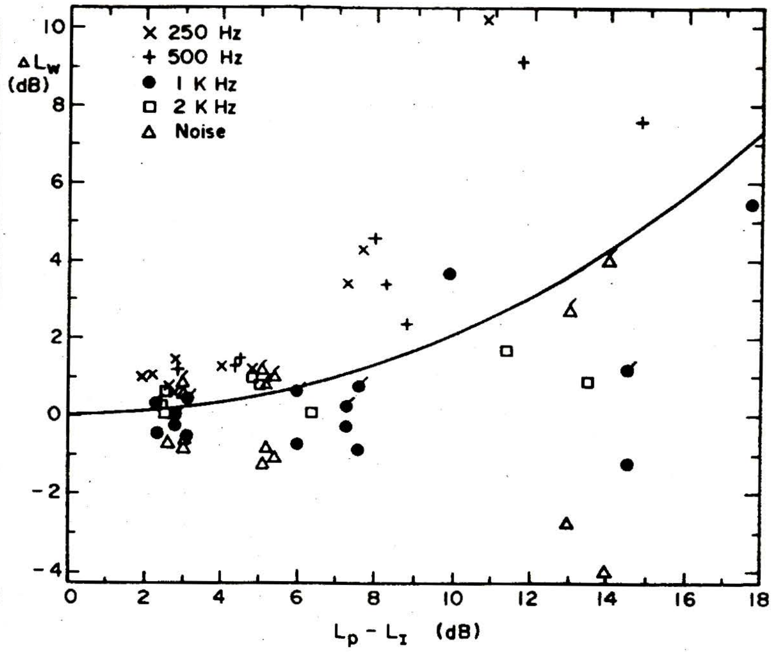

Figure 7. Variation of ΔL with $(L_p - L_I)$

3.3 SOUND POWER MEASUREMENTS

Fig. 3 shows the deviation ΔL_w in measured sound power level from the set power level of 80 dB for the B&K 4205 for various octave band and wide band noise using respectively a pair of 1/2 in. microphones with separation 12 mm and a pair of 1/4 " microphones with separation 6 mm. The 4205 was placed on a reflecting plane. The apparent background noise sound power level was only 55 dB. With the exception of the 500 Hz octave band noise and within the frequency range of application of the microphone pairs, the results all fall within the accuracy limits of the 4205 as in the case of anechoic measurements, thus supporting the use of the sound intensity measurements under non-anechoic conditions. Another set of measurements was performed with the 4205 placed on a mat of 25 mm thick fibre glass (density 50 kg/m^3) to increase the amount of absorption inside the measurement volume. The results indicate minimal effect on the sound power levels measured as the background noise level is low.

Sound power level measurements were then obtained for the Bosch hand drill with a pair of 1/4" microphones (separation 6 mm) with and without a windscreen and the resulting sound power spectra were shown in Fig. 4 for comparison with the anechoic and justaposition results. As the motor in the drill set up an air circulation, the measurements without the use of a windscreen give rise to very high values at low frequencies as expected whereas the use of a windscreen does seem to suppress the effect of a mean flow on the measurements. The linear and A-weighted sound power levels measured with different methods are tabulated in Table I and confirm the accuracy of the use of sound intensity measurements. An intensity map for the drill is included in Fig. 8 .

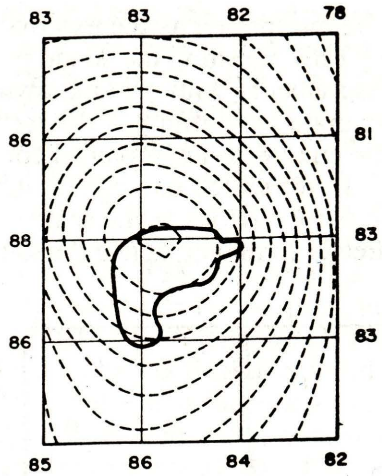

Figure 8. Intensity map for a hand drill

4. CONCLUSIONS

Two indirect methods (through sound pressure level measurements) and the direct method of measuring sound intensity have been presented for sound power determination. In the anechoic method, the facility is very costly and the procedure is long and tedious. For instance, it took two hours to complete the measurements for the drill. On the other hand, the cost of a reference sound source is negligible compared with an anechoic facility. The measurements performed using the justaposition principle indicate very good accuracy in a free-field environment and there is potential in using this method for 'in situ' measurements. However, even under such ideal condition, the results in Table I clearly indicate that the directionality of the source under test can cause substantial errors unless careful consideration has been given to proper spatial average measurements. Furthermore, there are various factors that affect the use of the justaposition principle for 'in situ' measurements. For example, the reverberation time in the environment should be independent of frequency and the sound power spectrum of the source under test should be similar to that of the reference source.

The sound intensity system, though more costly than the reference sound source, is considerably cheaper than the anechoic facility. The system has been demonstrated to yield sound power results with an accuracy to within $\pm$ 1dB in an non-anechoic environment. Furthermore, intensity map can be obtained and localisation of source is possible. The system is easy to use and the testing time is nomally quite short; for example, it took less than five minutes to complete the sound power measurements for the drill. However, in practice, results presented here indicate that background noise effects can become significant when the difference between the averaged sound pressure level and intensity level exceeds 7dB.

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THE MOVING OVERLAPPING FAST FOURIER TRANSFORM AS AN AID TO PHONOCARDIOLOGY

R.J. Alfredson

THE MOVING OVERLAPPING FAST FOURIER TRANSFORM AS AN AID TO PHONOCARDIOLOGY

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ABSTRACT

This study has examined the merits of using the moving overlapping Fast Fourier Transform for converting heart sounds into a visual format which can be used to identify heart abnormalities. It includes a brief description of the functioning of the heart and an account of the origin of normal and abnormal heart sounds. Several different heart sounds were analysed and the resulting time dependent spectra were shown to be able to identify easily normal heart sounds and abnormal heart sounds such as click and murmur. The significance of some of the smaller components in the spectra was less obvious. Work is continuing with a wider range of heart sounds and the importance of the infrasound associated with the heart is being investigated.

ACKNOWLEDGMENTS: The support of Professor Carl Wood of the Queen Victoria Medical Centre, and the medical supervision provided during the measurement program is gratefully acknowledged. Similar thanks go to the patients who willingly participated in the program.

1. INTRODUCTION

Ausculation is widely used as a first means of identifying heart condition. It relies on practicioners being competent in recognising several different types of heart sounds. The fact that ausculation has had a long history and is still in current use implies that audible pattern recognition is useful and can be highly developed.

Visual pattern recognition provides a complementary approach for detecting and diagnosing heart condition. These visual patterns can easily be related to the audible ones due to the developments that have occurred in the past decade in the digital processing of signals. Thus the use of moving, overlapping fast Fourier transform permits the generation of time dependent spectra which display frequency content and amplitude in much the same way as the ear interprets sound. This allows a permanent record to be obtained which can display several cycles of the heart beat. In addition a reasonable signal to noise ratio is possible and if desired frequencies below those of the audio frequency range can be included. Frequency resolution is good.

Some tests have been carried out on heart sounds in order to gain an appreciation of the usefulness of the time dependent spectra method for identifying different types of heart sounds. The initial results were very satisfactory and are described below. However as a preliminary to discussing these it will be worthwhile to outline briefly the relevant physiological features of the heart and to indicate the origin of the sources of noise.

2. THE HEART

The heart is in reality two pumps working synchronously but independently. It is shown in Fig. 1 from Lamb (1970). There are four chambers. The two upper chambers, or artria, act essentially as reservoirs while the two lower chambers, the ventricles, provide the pumping action. Thus each pump consists of an atrium and a ventricle and the pumps are arranged on the left and right side of the heart. The right side receives blood low in oxygen and high in carbon dioxide and sends it through the pulmonary circulation where the carbon dioxide is extracted and oxygen is taken up. The pump on the left side receives the blood from the lungs and sends it through the rest of the body. The greater distances involved in this latter process compared with the pulmonary circuit requires that the left side operates at a substantially higher pressure than the right side.

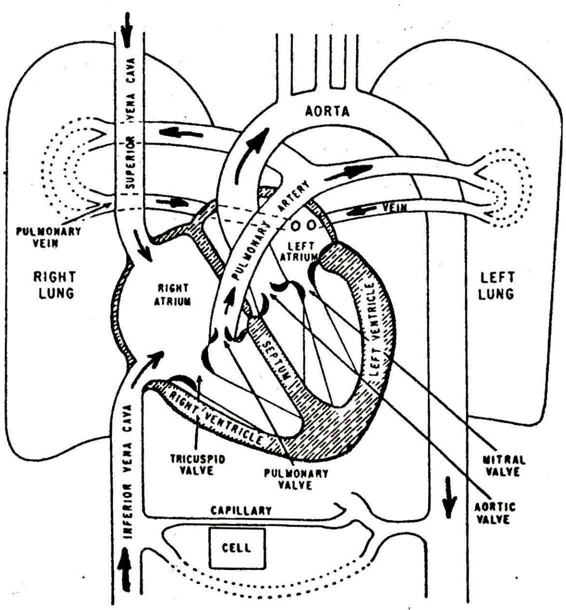


Fig. 1 The Configuration of the Heart

The operation of the heart is dependent upon the contraction and relaxation of the various chambers and on the functioning of two sets of cardiac valves. The atrioventricular valves separate the ventricles from their respective atria. The left atrioventricular valve is called the mitral valve and the right is the tricuspid valve.

Two semilunar valves separate the ventricles from their arterial trunks. The aortic valve separates the left ventricle from the aorta and the pulmonary valve separates the right ventricle from the pulmonary artery.

3. THE CARDIAC CYCLE

The cardiac cycle is traditionally divided into two parts, systole and diastole. The onset of systole occurs when the atrioventricular valves are wide open so that the atria and ventricles function as single chambers on each side of the heart. The contraction of the ventricles then raises the pressure, promptly shutting these valves. Further increase in pressure occurs until the respective ventricle pressures exceed those of the aorta and pulmonary arteries. The semilunar valves are forced open and blood is ejected. Initially the rate of ejection is high with half of the transfer occupying the first quarter of the ejection period.

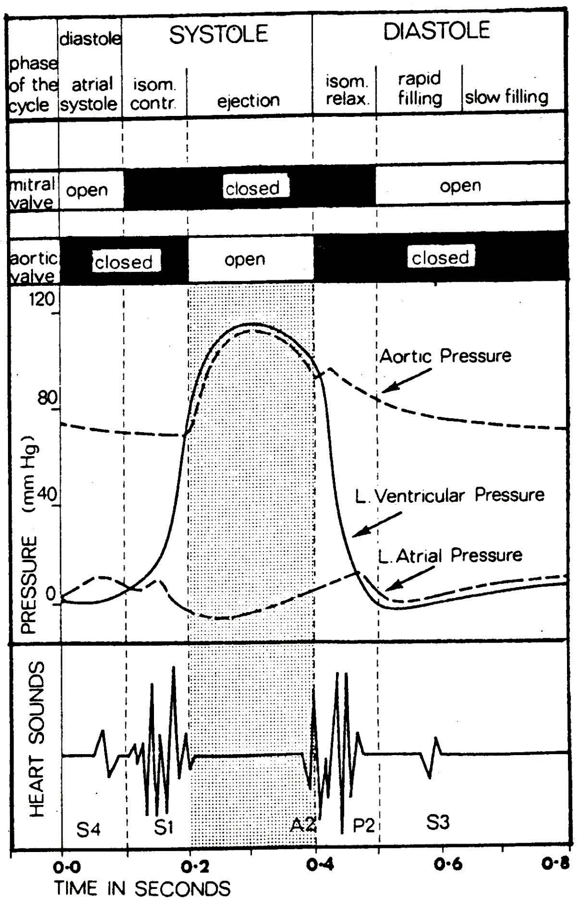


Fig. 2 Pressure Variation and Heart Sounds During a Cardiac Cycle

Ventricular relaxation, or disastole, begins causing the closure of the semilunar valves. Further relaxation results in continued lowering of the ventricular pressures until they fall below those of the atria at which stage the autrioventricular valves open. Blood is drawn into the ventricles, initially very rapidly. The final stage of diastole involves an atrial contraction phase when the remaining blood to be transferred is forced into the ventricles. The above process repeats each cycle. The process for the left side is shown in Fig. 2 from Wartak (1972). That of the right is similar except that the peak ventricular pressure is about 35mm. None of the chambers empties completely during the cardiac cycle. The ventricles eject about 60% of their contents during systole while the atrial variation is less.

4. ORIGIN OF HEART SOUNDS

Heart sounds are of two essential types. The first type is of a transient nature and short duration. It is associated with the stopping and starting of the blood flow as a result of the opening and closing of valves. The second type is associated with the turbulent flow of blood in the heart and large vessels. Sounds of this type are referred to as murmurs. Each of these will be briefly discussed below.

4.1 TRANSIENT HEART SOUNDS

There are traditionally four heart sounds in one cardiac cycle. Of these the first and second are normally audible while the third and fourth are normally inaudible.

The first heart sound is produced by the sudden closure of the mitral and tricuspid valves associated with the ventricular contraction, the rapid rise in pressure, and the opening of the pulmonary and aortic valves. The closure of the atrioventricular valves and the opening of the semilunar valves is separated by only 10 to 30 milliseconds and the two components merge to give a single sound. The sound is usually described as 'lubb'.

The second heart sound results from the abrupt cutting off of the blood flow from the heart caused by the closure of the semilunar valves. Since the aortic valve closure occurs marginally earlier than the pulmonary valve closure, the sound consists of two components. These components can sometimes be distinguished because of the splitting that occurs during respiration. Thus during inspiration the two sounds may be recognised as a 'dlup' sound. During expiration only a single 'dup' sound is heard.

The third and fourth heart sounds result respectively from rapid ventricular emptying, and the sudden inflow of blood into the ventricles. Neither of these is normally heard except occasionally in a younger person with a vigorous heart.

4.2 MURMURS

Murmurs are the result of turbulent blood flow. This can be caused by obstructions to the flow, sudden enlargement in the cross sectional area of flow, pathologic communication in the cardiovascular system, or ruptured intracardiac structures. Occasionally murmurs are of no clinical significance as for example when they occur after vigorous exercise. Innocent murmurs occur in a large proporation of children of school age. Murmurs may occur during systole or diastole or be continuous.

With this background, it is now possible to proceed to analyse and understand the heart sounds.

5. ANALYSIS OF HEART SOUNDS

Two sources of heart sounds were analysed. The first source was based on a training tape produced for medical practicioners. This tape contained 12 different heart sounds which had been generated artificially in order to high-light the various components of the heart sound. Analysis was made initially of these signals so that an appreciation of the different visual formats of the time dependent spectra could be obtained. The second source was based on measurements made at Monash University using a Bruel and Kjaer type 4146 microphone and associated amplifiers.

Analyses were carried out for each signal on the HP21MX computer having sampled a few seconds of data at 1000Hz. Some 200 FFT's were calculated for each heart sound using 128 words for each FFT. The frequency resolution was thus 7.8Hz with the duration of each signal being 128 milliseconds. Each subsequent FFT was separated by 10 milliseconds from the previous one. A sine squared window was used in the analysis. Signal to noise ratio was about 50dB.

Fig. 3(a) shows two cycles of the simulated normal heart sound. The first and second sounds are easily identified in each cycle. In the second cycle there is a small amount of splitting and, as noted previously, this is normal and indicative of inspiration.

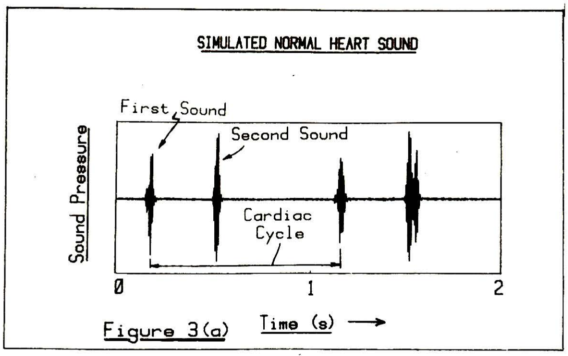


Figure 3(a)

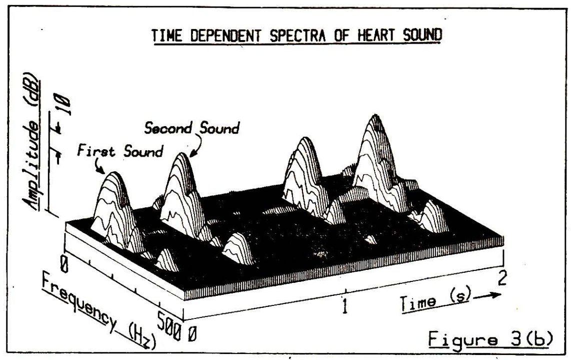


Figure 3(b)

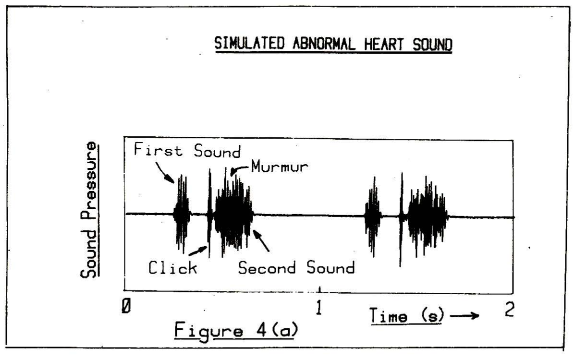
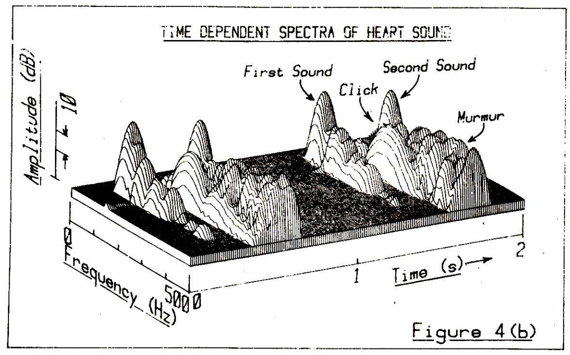


Figure 4(a)



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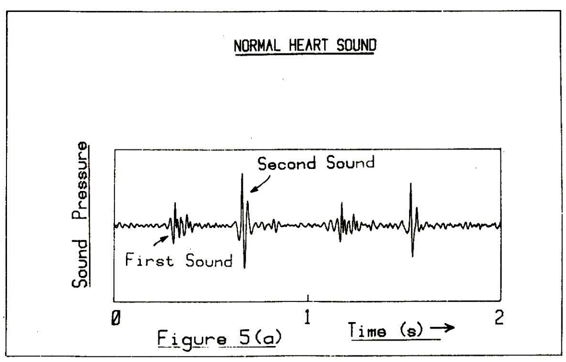

Figure 5(a)

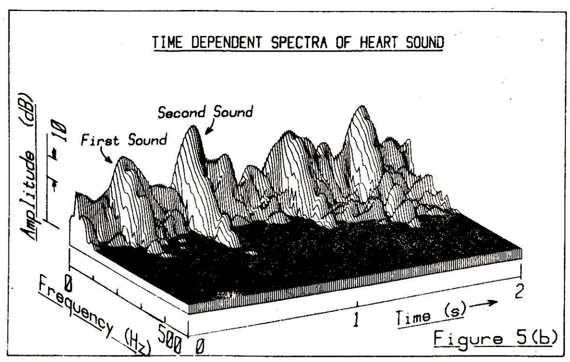

Figure 5(b)

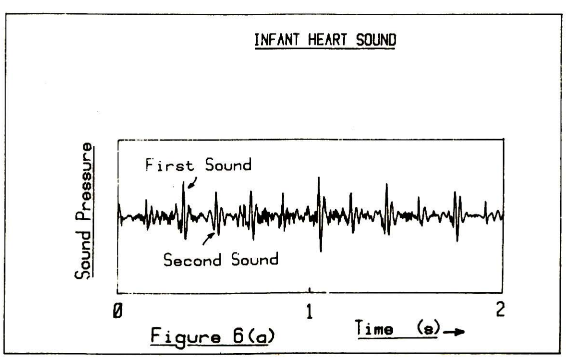

Figure 6(a)

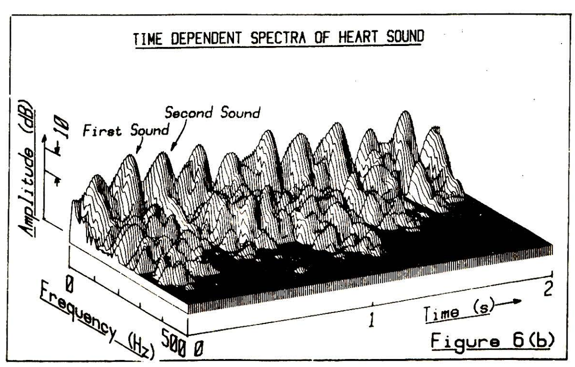

Figure 6(b)

The frequency content and the time dependent amplitude of this signal is given in Fig. 3(b). The first and second sounds are obviously of low frequency with peaks occurring at a little below 100Hz. There is little in the spectrum above 200Hz. The repetitive 'lubb-dup' sound is clearly visible above the surrounding terrain.

A simulated abnormal heart sound is given in Fig. 4(a). There are four elements to each cycle. The first sound is followed by a mid systolic click and then by a late systolic murmur which virtually conceals the second heart sound. The corresponding time dependent spectra is given in Fig. 4(b). The first and second sounds are conspicuous with their low frequency peak. The mid systolic click is also apparent with somewhat higher frequencies - peaks occur around 200Hz. The murmur is characterised by a continuous spectrum which no doubt extends in this case beyond 500Hz. (Antialiasing filters set at 500Hz were used during data collection.) This plot easily allows the components of the time domain signals to be clearly identified.

The first of the (natural) heart sounds is given in Fig. 5(a). For comparison with the previous signals the low frequency components, those below 20Hz, have been filtered off but they obviously contain much information which is outside the audible range. (That is the subject of a separate study.) The present signal is recognisably similar to that of Fig. 3(a) but apparently the first and second sounds are of a lower frequency. Note the high damping present particularly in the second sound. The time dependent spectra of Fig. 5(b) confirm the lower frequency content where peaks occur at around 50Hz. There is significantly more activity during the entire cycle at the low frequency end of the spectrum. Possibly it arises from the 3rd and 4th heart sound as discussed in Section 4.1. Their origin is being further investigated.

The final heart sound is that of a six week old baby, Fig. 6(a). As expected the heart rate is much faster but is recognisably similar to the previous normal heart sounds. The first and second heart sounds are clearly visible although the apparent signal to noise ratio is smaller. The three dimensional spectra, Fig. 6(b), indicates that this higher frequency component has some periodicity with possibly a component of murmur which as noted previously is not unusual in young vigorous hearts. There is obviously plenty of scope for further investigation.

6. CONCLUDING COMMENTS

This study is regarded as a first attempt at analysing heart sounds using time dependent spectral analysis techniques. Only a few have been included in this paper but they are sufficiently illuminating to encourage a more systematic study. In particular the ability to recognise visually the different types of heart sounds, even when the time domain signal is apparently confusing, is exciting and complements the normal ausculatory process. It is recognised of course that the frequency-amplitude-time dependence is present in the original signal but its perception is not as clear as it is with these time dependent spectra. Further work is continuing, particularly in understanding the significance of the infrasound which is present in the heart sound.

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Acoustics in the Eighties

THE EFFECT OF MODAL PATTERNS OF VIBRATION ON NEAR FIELD ACOUSTIC INTENSITY

R.J. Alfredson and I.J. Phelan

THE EFFECT OF MODAL PATTERNS OF VIBRATION ON NEAR FIELD ACOUSTIC INTENSITY

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ABSTRACT

This study reports the first stage of a systematic investigation aimed at understanding the limitations and the validity of using near field acoustic intensity measurements as a means of identifying and ranking noise sources in a multi-source situation. Several different modal patterns of vibration of a baffle mounted circular piston were generated theoretically and the acoustic intensity calculated using numerical integration procedures. The results indicated that regions of significant vibrational activity were generally associated with higher near field intensities. Regions of low near field intensity were usually related to small amplitudes of vibration and associated anti-phase surface motion. There were nevertheless some unexpected patterns which could rise to misleading conclusions regarding the ranking of noise sources.

1. INTRODUCTION

The direct measurement of acoustic intensity in the near field of vibrating surfaces has gained acceptance as a method of identifying and ranking noise sources in a multi-noise situation, e.g. Alfredson (1977), Rion (1984) and Krishnappa (1981). The implicit assumption is that high levels of intensity are an indication that surfaces in the immediate neighbourhood are significant sources of noise.

A recent study however, Phelan (1986) has shown that regions of negative intensity can occur close to surfaces that are being mechanically excited. This study involved both the prediction of the near field intensity (based on an analysis of the vibration patterns) as well as the direct measurement of acoustic intensity using a conventional two microphone intensity probe. Both approaches conclusively indicated the region of negative intensity. Others too have reported such findings, e.g. Gade (1982) but have apparently not pursued the matter further indicating simply that acoustic energy is being absorbed.

The present study was thus undertaken to begin to understand under what circumstances negative intensity was likely to occur near mechanically driven vibrating surfaces, whether high levels of vibration necessarily give rise to high values of near field intensity, and to start cataloguing near field intensities adjacent to surfaces whose vibrational patterns were known. The results, as discussed below, indicated that in fact near field measurements are of use in identifying sources at least for the simpler mode shapes examined. Regions of small intensity indicated that surfaces in the immediate neighbourhood were fairly quiet from a vibrational point of view and that other surfaces nearby were vibrating in antiphase. None of the mode patterns examined produce negative intensity.

2. THE PREDICTION OF ACOUSTIC INTENSITY

The prediction of acoustic intensity involves averaging the instantaneous product of acoustic pressure and particle velocity. The direction of this latter quantity, being a vector, must be specified as must acoustic intensity.

For a sinusoidal excitation, intensity, I, is given by,

$$I(x,y,z) = \frac{1}{2}\,\mathrm{Re}\{p'(x,y,z,t)\; u'(x,y,z,t)^*\} \qquad (1),$$

where p' represents, in complex notation, the fluctuating pressure,
u' represents, in complex notation, the particle velocity.
Conjugation is indicated by the asterisk.

The surface subject to vibration was a circular piston mounted in an infinite baffle. Following the customary approach, it was treated as a source consisting of an infinite series of point monopoles configured so that they covered the piston surface. The fluctuating pressure and particle velocity at any point was found by integrating the effects of each point source over the piston area - see Fig. 1.

Adopting the approach of e.g. Reynolds (1981), the fluctuating pressure, p', at position $N(x,y,z)$ is given by,

$$p'(x,y,z,t) = \frac{j\rho c k e^{j\omega t}}{2\pi} \int_o^a \sigma \int_o^{2\pi} \frac{u_x(\sigma,\psi)}{r(x,y,z,\sigma,\psi)}\, e^{j(\theta(\sigma,\psi)-kr)}\, d\psi\, d\sigma \qquad (2),$$

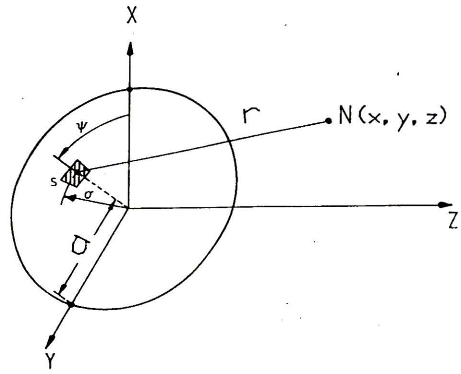

Fig. 1 Circular Piston Configuration

where

ρc is characteristic impedance,
k is the wave number
u_s is the velocity of the surface area $\sigma d\psi d\sigma$ which has associated with it a phase angle θ,
r is the distance from the elemental area to the point at which pressure is required,
ω represents frequency, and t time.

Resolving particle velocity into the Cartesian co-ordinate directions namely u'_x, u'_y, u'_z results in the following expressions,

$$u'_x(x,y,z,t) = \frac{e^{j\omega t}}{2\pi} \int_o^a \sigma \int_o^{2\pi} (1 + jkr) \frac{u_s}{r^2} e^{j(\theta - kr)} \cdot \left(\frac{x - \sigma\cos\psi}{r}\right) d\sigma d\psi \, ,$$

$$u'_y(x,y,z,t) = \frac{e^{k\omega t}}{2\pi} \int_o^a \sigma \int_o^{2\pi} (1 + jkr) \frac{u_s}{r^2} e^{j(\theta - kr)} \cdot \left(\frac{y - \sigma\sin\psi}{r}\right) d\sigma d\psi \, ,$$

$$u'_z(x,y,z,t) = \frac{e^{j\omega t}}{2\pi} \int_o^a \sigma \int_o^{2\pi} (1 + jkr) \frac{u_s}{r^2} e^{j(\theta - kr)} \cdot \left(\frac{z}{r}\right) d\sigma d\psi \, . \qquad (3)$$

These three equations together with equations (1) and (2) allow intensity to be calculated in any direction and at any position.

The double integrations present a challenge except for the on axis and the far field position and even then only for in-phase constant amplitude motion of the piston. The challenge was met by resorting to conventional numerical integration approaches using some 800 elemental areas on the piston surface. Extensive checking of the routines was carried out and where possible compared with known analytical solutions. The errors that occurred were minimal except immediately adjacent to the surface of the piston - typically less than a few millimetres. This was of no practical consequence since normally the intensity probe size prevents the sensing elements approaching closer than about 20mm.

3. PISTON MOTION

Five different resonant modes of the piston motion were examined and for each the intensity on axis and at 50mm from the piston surface were calculated. Both traverses referred to intensity in the axial direction. For comparison with earlier studies the piston frequency was set at 5000Hz and its diameter at 282mm. Setting the outer edge of the piston to zero displacement, allowed the following function to be used to describe the piston motion, namely

$$\phi(\sigma,\psi) = \cos(m\,\psi)\, J_m\left(\beta_{m,n} \frac{\sigma}{a}\right) \qquad (4),$$

where m and n are integers,

J_m is the Bessel function of the first kind of order m,
$\beta_{m,n}$ can assume particular values depending on the value of m and n,
σ refers to the radial direction,
ψ refers to the angular direction
a is the outer radius of the piston.

The following table sets out the mode shapes used.

Piston No.	Piston Function, ϕ
1	1.0 (uniform)
2	J_o (2.450 σ/a)
3	J_o (5.520 σ/a)
4	cos ψ J_1 (3.832 σ/a)
5	cos ψ J_1 (7.016 σ/a)

The mode shapes of pistons 2 to 5 are indicated in Fig. 2. Note that the various points on the surface are either in phase or in antiphase. Piston 1 has uniform motion and is not shown.

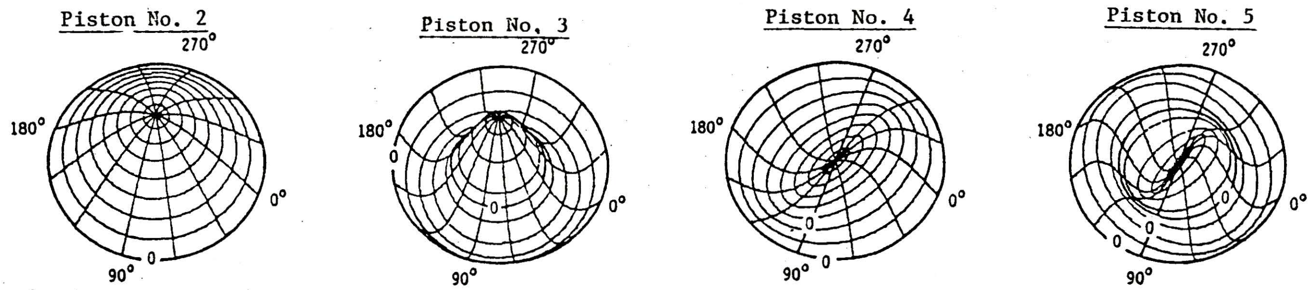

Fig. 2 Mode Shapes of Circular Piston

4. ACOUSTIC INTENSITY

As noted earlier acoustic intensity was calculated in the axial direction for each piston along two separate traverses. The first involved moving away from the piston along the piston centreline. The second involved a transverse traverse at 50mm from the piston surface. It corresponded to the XZ plane of Fig. 1 and to the plane normal to the piston and containing 0°-180° line in Fig. 2.

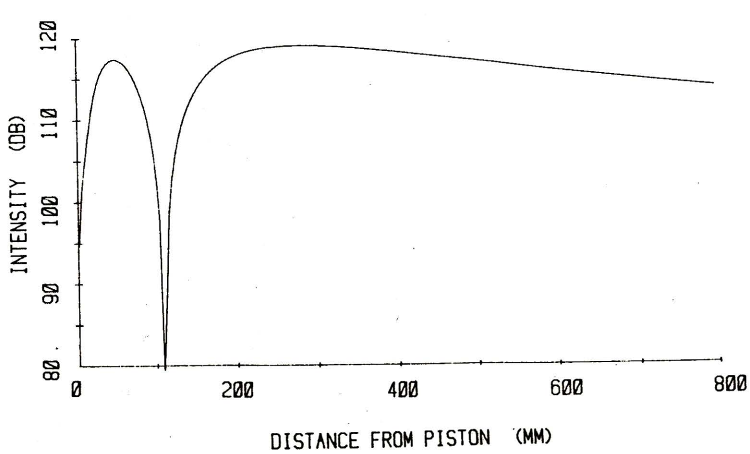

Fig. 3 On Axis Intensity of Piston No. 1

The on axis intensity for piston No. 1 is given in Fig. 3.

The dip at 110mm from the piston surface is due to the fact that the acoustic pressure at that position becomes very small as is demonstrated in many standard texts, e.g. Kinsler (1982). Note that near the surface acoustic intensity varies by 40dB or more which could greatly influence the importance attributed to the source area at the centre of the piston.

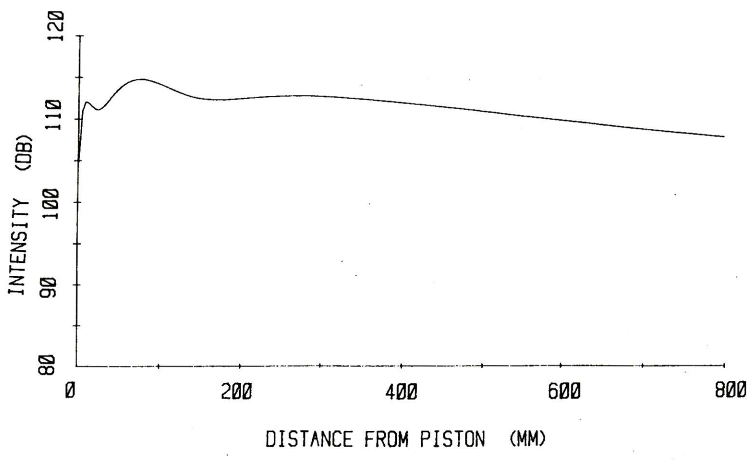

Fig. 4 On Axis Intensity of Piston No. 2

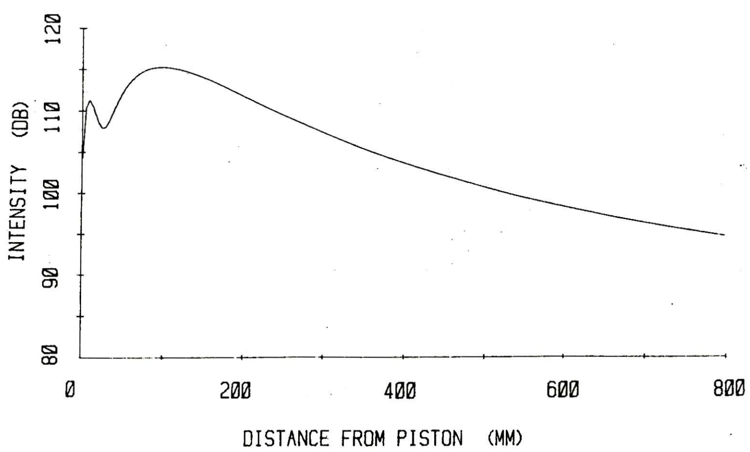

Fig. 5 On Axis Intensity of Piston No. 3

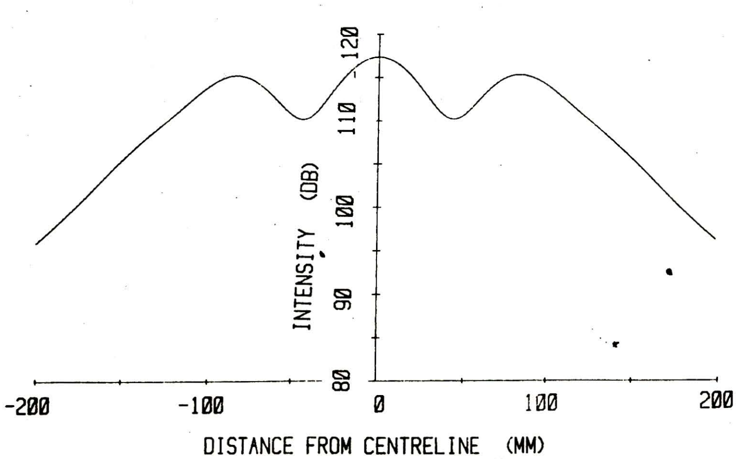

Fig. 6 Near Field Intensity of Piston No. 1

The on axis intensity for pistons 2 and 3 behave much more smoothly with near-field variations only of the order of about 5dB. The more rapid fall off in intensity with distance from the surface of piston No. 3 is not surprising in view of its expected lower radiation efficiency caused by the counterphase motion of various sections of the surface, of the piston.

The on axis intensity for pistons 4 and 5 are not given as both of these pistons have a dipole like motion producing zero intensity along the axis. It was a tribute to the numerical integration procedure that the calculated intensities were more than 120dB below those of piston Nos. 1 to 3 - i.e. the signal to noise ratio was in excess of 120dB.

The nearfield intensity in the z direction at 50mm from the surface of piston No. 1 is given in Fig. 6. There is about a 15dB variation in level across the piston surface even though the piston velocity is of uniform amplitude. The reason for the ripple in the central region is not obvious, although the fall off towards the outer edges of the piston would be expected.

The corresponding plot of the near field intensity for piston No. 2 is given in Fig. 7. This pattern contains no surprises in that higher levels of intensity occur adjacent to regions of larger amplitudes of vibration.

The nearfield intensity of piston No. 3 is also indicative of the nearby vibrational activity. It approaches a small value near the node in the piston

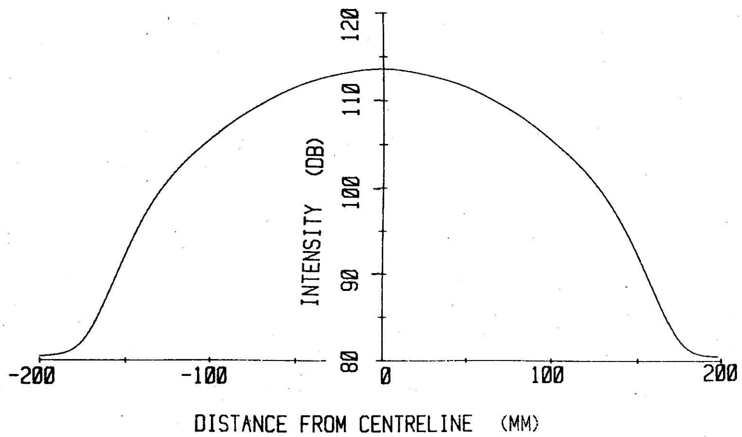

Fig. 7 Nearfield Intensity of Piston No. 2

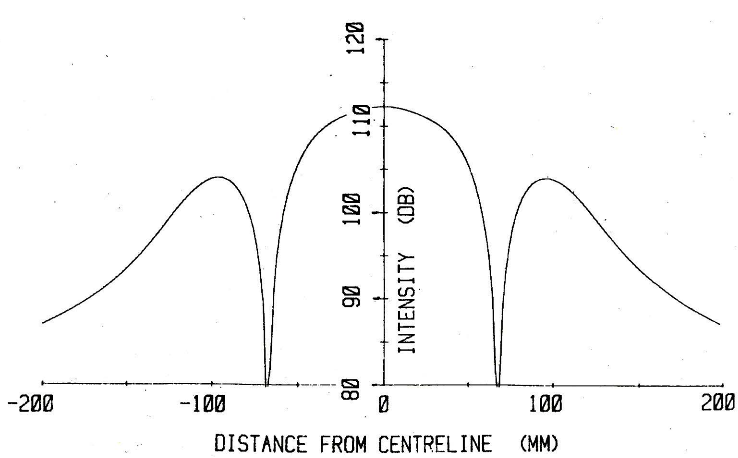

Fig. 8 Nearfield Intensity of Piston No. 3

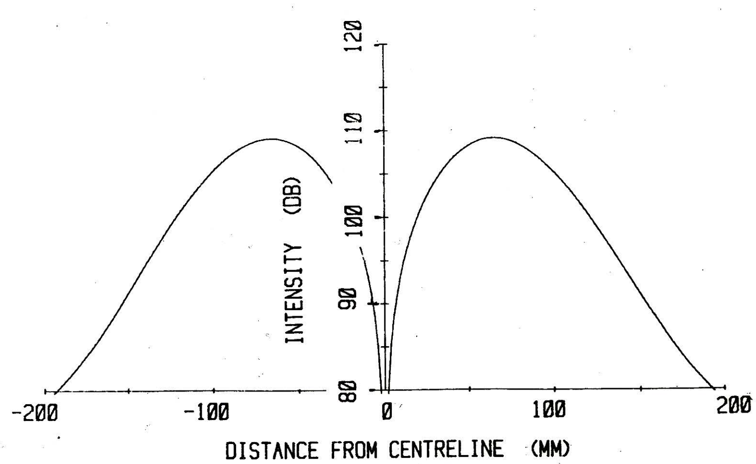

Fig. 9 Nearfield Intensity of Piston No. 4

vibration pattern, namely at about a radius of about 65mm. These smaller values are of course only partly due to the reduced activity in the vibrational field. The effect of adjacent areas operating in anti-phase would also contribute to the lowering in intensity levels.

The same general trend is observed in Figs. 9 and 10 which refer to the two remaining pistons. Specifically the regions of minimal piston motion and the effect of antiphase motion of surrounding surfaces produce minima in nearfield acoustic intensity. Similarly regions of greater piston motion tend to produce peaks in acoustic intensity.

It is clear from these examples that knowing in advance the modal patterns of vibration it is possible to make a fairly informed approximation to the expected nearfield intensity at least for the fairly simple single excitation patterns examined. It would however be rather more difficult to move in the opposite direction - i.e. to predict vibrational characteristics from the nearfield intensity measurements. It would also be very unwise to infer sound pressure levels from the intensity measurements without further detailed investigation. The fields are clearly reactive and the phase angle between pressure and velocity would need to be known.

5. CONCLUDING COMMENTS

The study has begun the process of cataloguing the nearfield acoustic intensity patterns produced by a baffle mounted circular piston vibrating in

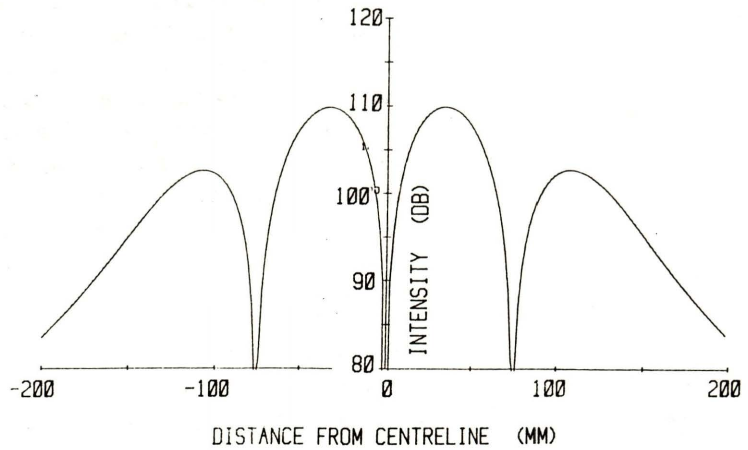

Fig. 10 Nearfield Intensity of Piston No. 5

its various resonant modes. Overall the results agree fairly well with those which would have been intuitively expected. In particular, regions of small vibrational activity and associated antiphase motion of the adjacent surfaces produced minima in the nearfield intensity. Higher levels of vibration generally resulted in increased levels of near-field intensity. However occasionally, e.g. on axis of piston No. 1, there were variations in intensity which were not intuitively obvious.

The piston motions examined so far have supported the notion that the measurement of the nearfield intensity is a reasonable approach to source identification. Further study is clearly needed and there has not been to date any light on the circumstances which lead to the generation of negative intensity.

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Acoustics in the Eighties

Session 3C: Noise Control

Acoustics in the Eighties

MASS DAMPERS FOR NOISE AND VIBRATION REDUCTION

L.L. Koss

MASS DAMPERS FOR NOISE AND VIBRATION REDUCTION

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ABSTRACT

Three basic elements are used to control high levels of noise and vibration. They are stiffness, mass, and damping. This paper describes the use of a new type of mass damper to control noise and vibration in impulsively operated machinery such as punch presses and flying shears. The trade name of this damper is called "Rodam".

The Rodam is placed between the moving slide and bed of a press to prevent the sudden unloading of the frame at the time of material fracture during a blanking operation. This action decreases the amplitude of the high frequency components of the force time history, thus reducing frame vibration amplitudes in comparison to the uncontrolled situation.

Measurements made on a flying shears decking mill with and without a Rodam in place have shown that the Rodam has provided an eight decibel vibration reduction along with a two dB(A) peak noise reduction. The paper will describe the technical operation of the Rodam and its use in the flying shears machine.

1. INTRODUCTION

Many impulsive noises and vibrations are associated with the sudden force unloading of a machine structure. A typical force time history of such an unloading is shown in Figure 1. Force build-up can take several milli-seconds whereas unloading can occur over a period of 100 microseconds. This severe rate of unloading of a machine frame causes the frame to vibrate with subsequent radiation of sound and transmission of vibration into the ground. The unloading is associated with work material fracture in processes such as blanking and piercing. Two types of machines which perform such operations are punch presses and flying shears.

Noise levels associated with blanking operations are given by Koss (1979) and Koss (1981). Other publications explain the phenomena of blanking noise, Evensen (1980) and show that the sound level and vibration are associated with the rate of change of force with time. In order to control the noise at the source the force acting between the bed and crown of a press has to be modified to allow for a gradual unloading. Several attempts at controlling the unloading using hydraulic shock absorbers between the slide and crown of presses have proven only marginally successful. The hydraulic shock absorbers had to be tuned to act at the point in time when force unloading commences. If the shock absorber acts prior to unloading the press may not perform its function due to an increase in load. If it acts too late, the shock absorber does not fulfill its function. The difficulty in tuning such a device in practice has made such a solution impracticable.

Secondly, a hydraulic shock absorber depends upon a velocity difference for its action. Once the unloading occurs and velocity build-up commences, impulse sound may have already been radiated. Thus a shock absorber device that depends upon acceleration for its action and does not have to be tuned would be ideal for control of noise and vibration of a press.

2. RODAM INERTIAL SHOCK ABSORBER

An inertial shock absorber was devised by Rosendal (1984) for the reduction of noise from punch press operations. The theory of its operation is based upon converting a linear motion into a rotary motion, a diagram of a Rodam shock absorber is shown in Figure 2. The top platten is attached to the slide of a press and the bottom is attached to the bed of the press. For a typical press operation the slide moves from its top dead centre position towards the work. This movement causes the centre spindle to move downwards which in turn generates motion of the planet gears and outside annulus and flywheel.

The effective linear mass of the rotating inertias is about 600 kg for a 160 kN capacity Rodam. So long as the press slide acceleration motion amplitudes are low and/or the stroke is short the Rodam will not significantly mass load the slide during the non-work portion of the operation.

Once a blanking operation is complete (fracture occurs) in normal operating mode the slide accelerates towards bottom dead centre. The effect of the Rodams is to present a "blocking mass" to this acceleration and thus prevent unloading of the press frame. This blocking mass action reduces frame accelerations wih subsequent reduction of sound radiation from the frame and transmission of vibration into the ground. Also, the Rodam does not have to be tuned to act at the position of material fracture. A spring washer clutch disengages the inertial action during the up stroke.

A Rodam absorber thus can reduce vibration and sound without having to be tuned. In the next section experimental results obtained from the use of a Rodam on a flying shears will be given.

3. FLYING SHEARS VIBRATION REDUCTION

A flying shears is used to cut metal sheet whilst the sheet is moving through a forming machine, metal decking and corrugated sheet are examples of items cut by a flying shears. The operation of a flying shears is similar to that of an ordinary metal shears except that the shears take up the velocity of the moving sheet in order to cut it on the run. A flying shears has a head which moves both vertically and horizontally during the cutting operation. The cutting operation is a source of vibration; this vibration is transmitted into the ground and can be a cause of disturbance to operators and neighbours.

A set of Rodams was used to reduce vibration from a decking mill (flying shears), this mill produces decking sheet. Results of vibration measured on the ground near the machine and in a neighbour's backyard are given in Table I. The use of Rodams reduces peak vibration by a factor of at least three and g_{rms} by a factor of at least two. A comparison of spectra for the in factory measurement is shown in Figure 3. The Rodams significantly reduce the vibration peak at approximately 18 Hz and for frequencies up to 60 Hz. New isolation pads placed under the machine (2 layers of super shearflex as supplied by G. Embleton and Co.) give a further reduction of the peak, and for other frequencies up to 100 Hz.

4. CONCLUSIONS

The use of Rodam shock absorbers in a Decking Mill flying shears machine has significantly reduced vibration on the ground near the machine and in a neighbour's yard. Ground vibration was reduced across a broad frequency range, and peak amplitude was reduced by a factor of three. Rodams can significantly reduce ground vibration levels for flying shears machines which "act like" punch presses.

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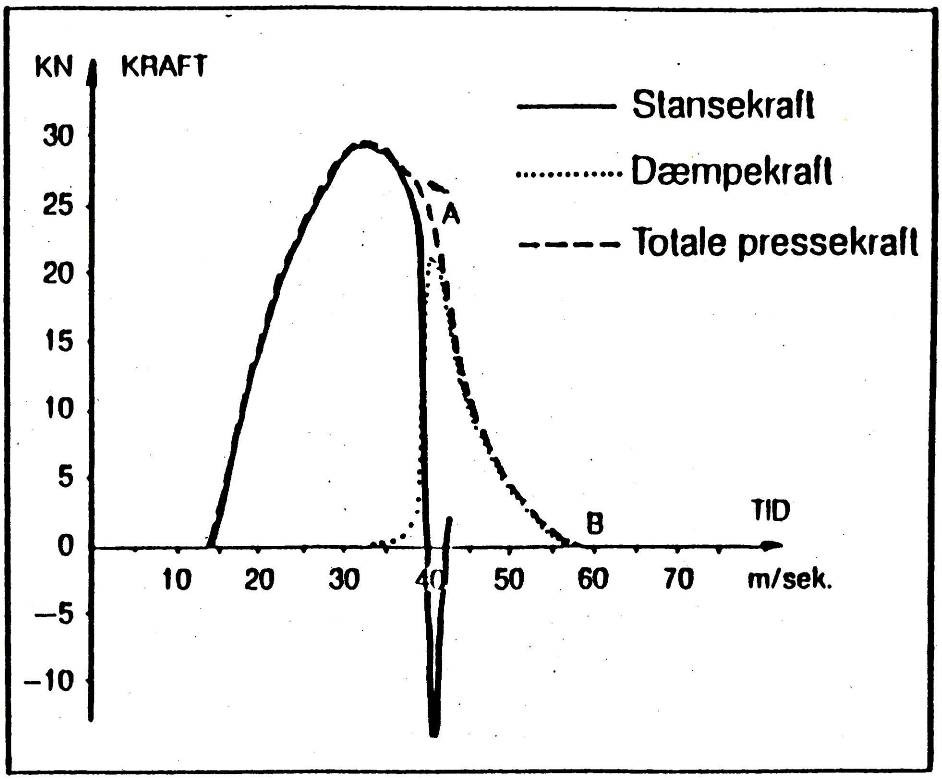

FIGURE 1. Force time history of blanking operation ——— original blanking force, ... damping force as provided by Rodam, ----- total force due to blanking and Rodam. Notice the complete change in character of the force unloading history due to the use of the Rodam.

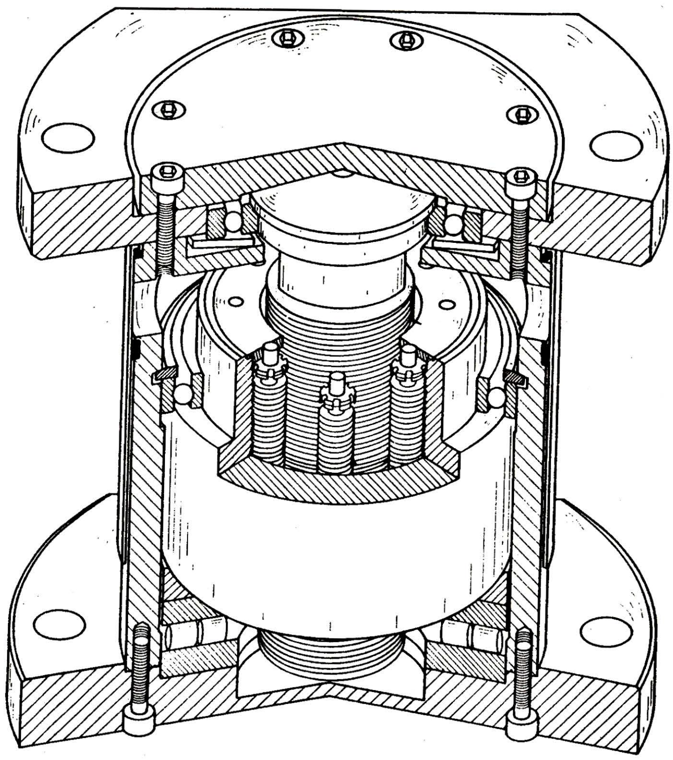

FIGURE 2(a) Exploded View of Rodam

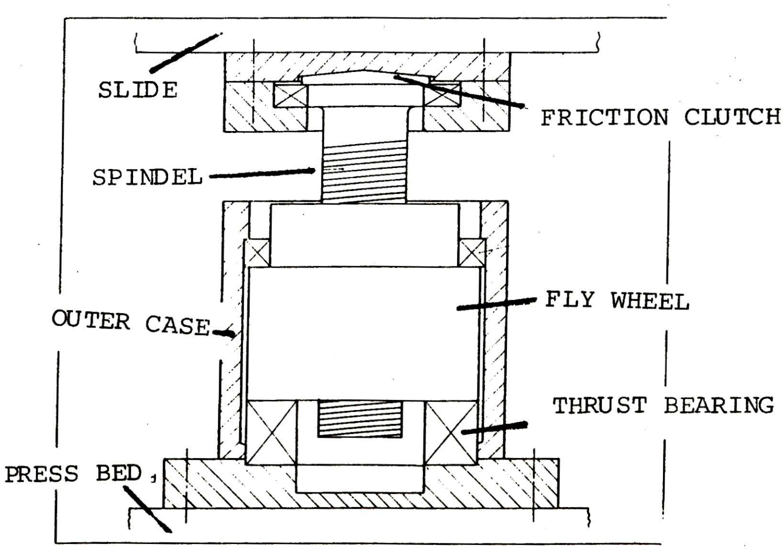

FIGURE 2(b) Schematic of Rodam showing important conceptual parts.

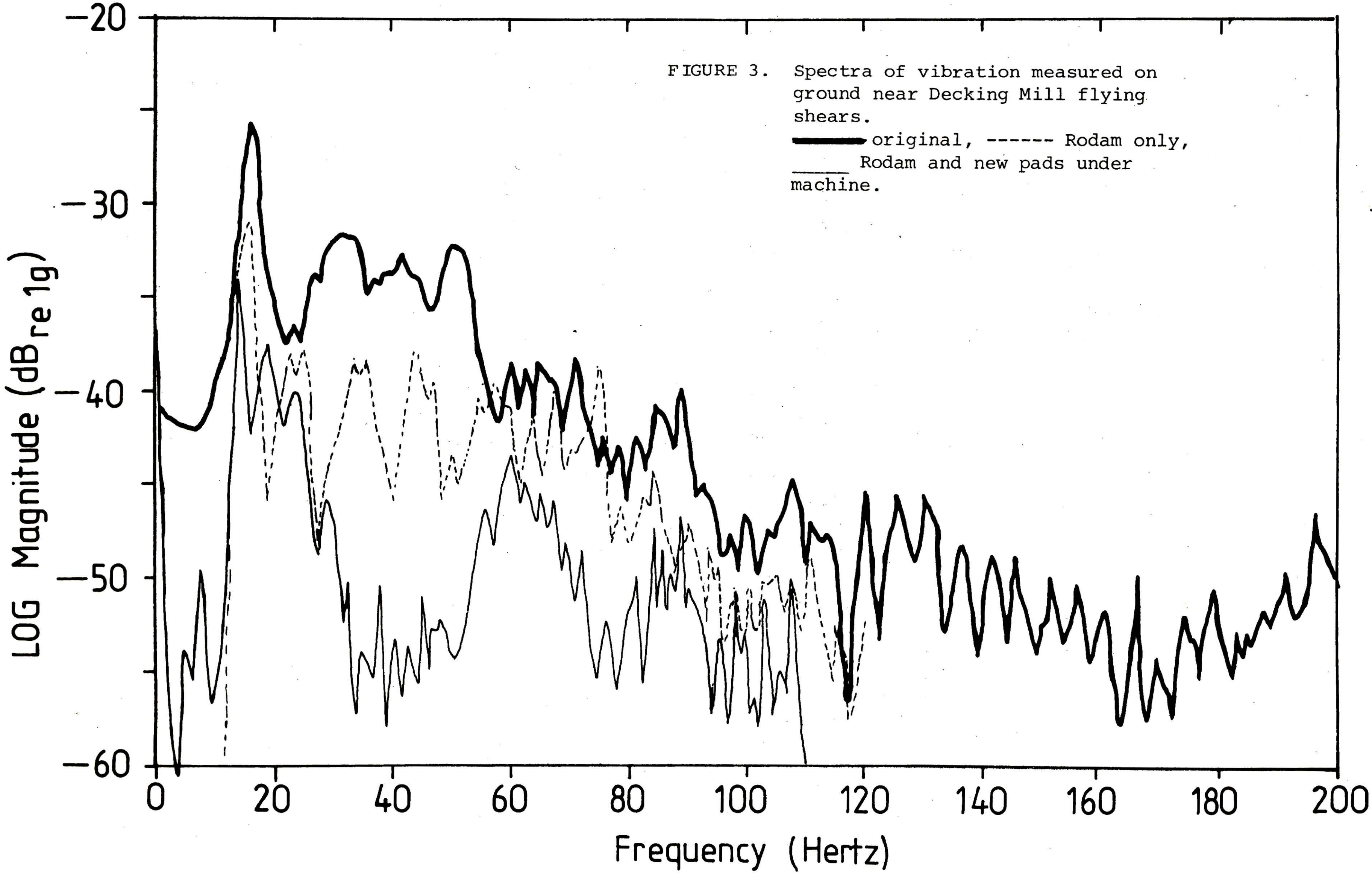

FIGURE 3. Spectra of vibration measured on ground near Decking Mill flying shears. ▬▬ original, ------ Rodam only, _____ Rodam and new pads under machine.

TABLE I

DECKING MILL VIBRATION RESULTS

A. In factory ground vibration measurements.

	Condition	g_{rms}	g_{peak}
1.	Original	0. 1	0.9
2.	Rodam only	0.05	0.3
3.	Rodam and new isolation pads	0.03	0.2

B. Neighbour's yard ground vibration measurements (vector result).

1.	New isolation pads below machine *	0.005	0.029
2.	Rodam and new isolation pads	0.001	0.006

* (Note: Original was not measured.)

Acoustics in the Eighties

TREATMENT OF CENTRIFUGAL FAN NOISE

R.F. LaFontaine, I.C. Shepherd and A. Cabelli

TREATMENT OF CENTRIFUGAL FAN TONAL NOISE

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ABSTRACT

A method of reducing the blade passing tones of centrifugal fans with a resonator mounted on the fan cutoff is discussed. Few guidelines are available for the design of these devices and no significant theoretical treatment has been published. Since the optimum configuration is not always a quarter-wave tube, the designer must experimentally determine resonator performance in every new application. A summary of the tests conducted on two dissimilar centrifugal fans demonstrates the complexities of implementation.

1. INTRODUCTION

Quarter-wave tube branch pipes and Helmholtz resonators are accepted methods of attenuating tonal noise in ducts, but are practical only if the offending sound propagates as plain waves. Where tones are present as higher modes in a centrifugal fan supply or discharge duct, a resonator located at the fan cutoff (Koopmann and Neise 1980, Neise 1982, Neise and Koopmann 1980) deserves consideration. Advantages of such silencers are that they treat the noise at the source and can attenuate the fundamental blade passing tone as well as the first few harmonics.

Design procedures for a resonator applied in this manner are non-existent or unpublished, and therefore experimentation is required to determine its potential. Furthermore, the arrangement which provides optimum insertion loss is not necessarily a quarter-wave tube, as envisaged in the original concept. In some cases, a tube length closer to a half wavelength is best.

The following summary of experimental work deals with two fan/resonator arrangements and illustrates several design factors. One arrangement, comprising a simple quarter-wave tube and a fan with a 12-blade rotor, realised 25 dB and 18 dB insertion loss for the blade passing tone and second harmonic respectively. The other arrangement, employing 8-blade and 16-blade rotors, needed two separately tuned tubes to achieve 13 dB and 7 dB insertion loss in the blade passing tone and its second harmonic respectively.

2. FAN NUMBER 1

2.1 EXPERIMENTAL APPARATUS

Fan No. 1 shown in Fig.1 operated at 1200 rpm and had a 12-blade, 400 mm diameter rotor and a normalised cutoff clearance $\delta/D=0.0125$, where δ is the cutoff clearance and D is the rotor diameter. The resonator comprised a 50x100 mm (cross-section) aluminium tube with a wooden piston at one end, and a perforated sheet metal cutoff of 40% porosity at the other. Bruel & Kjaer 4134 microphones were attached to the duct outlet flange (A), at the piston face (B) and to a rod which projected through the piston (C). Noise inside the supply duct was not considered. A Hewlett Packard 3582A spectrum analyzer provided spectra of the measured signals.

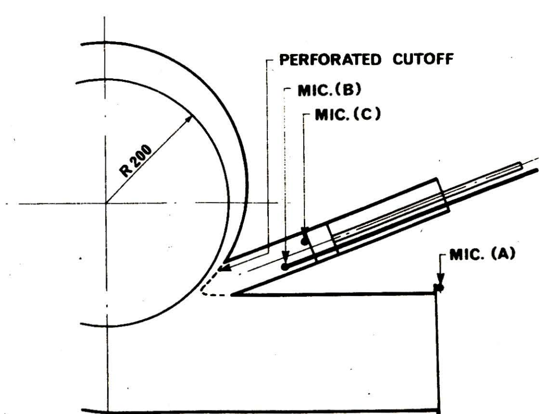

Fig.1 12-blade fan with single resonator tube. Microphones enable measurement of the tube phase gradients.

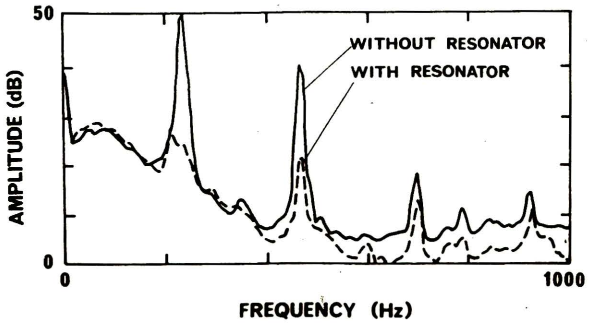

Fig.2 Noise spectra for Fan No.1;
——— without resonator;
– – – – with resonator.

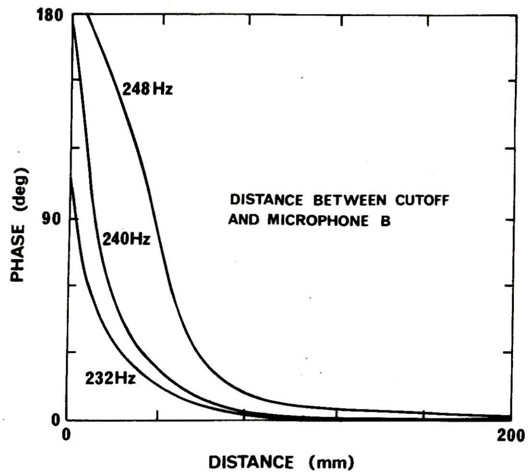

Fig.3 Phase of pressure at microphone (B) relative to that at microphone (C). Distance is measured between cutoff and microphone (B).

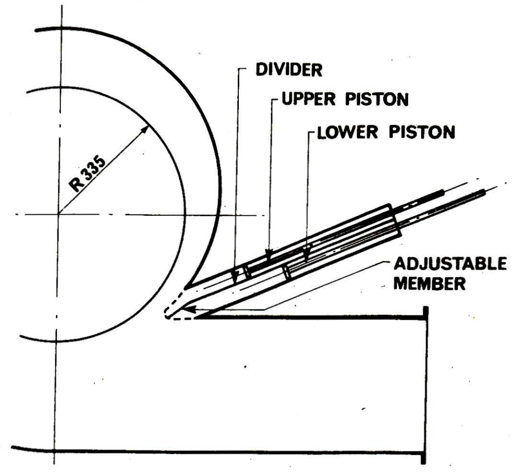

Fig.4 8- and 16-blade fan with resonator split to form two tubes.

2.2 RESONATOR PERFORMANCE

A noise benchmark was established with a solid cutoff using the microphone mounted on the duct outlet to measure the sound pressure level. With the resonator tuned as a quarter-wave tube, insertion losses of 25 dB and 18 dB were obtained at the blade passing frequency (240 Hz) and second harmonic respectively as shown in Fig. 2. Similar results were achieved with the resonator tuned for three-quarter wavelength operation. Effective silencing required the resonator to be well-sealed at the tube/cutoff junction and the resonator walls to be rigid.

Reduction of the second harmonic tone was unexpected since reflection from the piston was 'in-phase' with the wave entering the perforate. Tuning the resonator by moving the piston had little influence on the amplitude of the second harmonic, however the insertion loss at the blade passing frequency fell to 10 dB with the piston moved five percent of one wavelength from the optimum position.

Figure 3 shows the phase of the acoustic pressure along the tube for the blade passing frequencies 232, 240 and 248 Hz, with the resonator tuned to maximise insertion loss at 240 Hz in all cases. The phase of a roving microphone (B) relative to microphone (C) is shown for each of these frequencies, and typical quarter-wave tube behaviour is evident.

3. FAN NUMBER 2

3.1 EXPERIMENTAL APPARATUS

Fan No. 2 had a normalised cutoff clearance of 0.05 with optional 8-blade and 16-blade 670 mm diameter rotors. The fan speed was variable to 2040 rpm. The resonator, depicted in Fig.4, comprised two adjustable tubes of 33x186 mm cross-section. Pistons were 12 mm steel plate and the resonator walls were 12 mm plate aluminium. A neoprene band of circular cross-section contained in a groove around the edges of the pistons provided an acoustic seal at the tube walls. The divider which separated the upper and lower tubes had an adjustable member to alter the point of contact with the perforate. Cutoffs with surface porosities of 17%-50% were investigated.

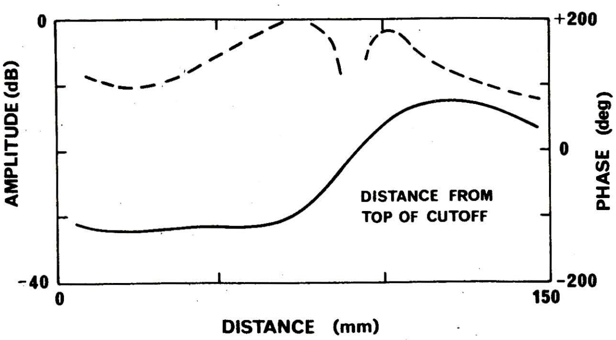

Fig.5 Acoustic pressure gradient around the solid cutoff of Fan No.2. Distance is measured from the top of the perforated cutoff. Phase ——— ; amplitude ---- .

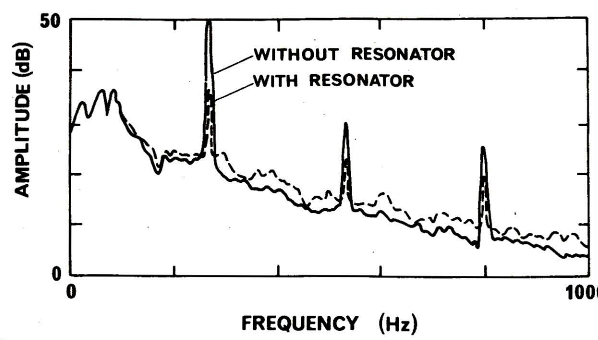

Fig.6 Noise spectra for Fan No.2; without resonator ——— ; with resonator ---- .

3.2 INFLUENCE OF DIVIDER POSITION

Preliminary tests were conducted using the 8-blade rotor, with the adjustable divider a straight line extension of the main member. The best combination of perforate porosity and resonator tuning produced only 6 dB insertion loss for the blade passing frequency of 272 Hz, obtained with the fan at full speed. To determine the optimum divider position, acoustic pressures were measured through ten orifices distributed around a solid cutoff and their spectra were obtained. Figure 5 shows the amplitude and phase of the 272 Hz fundamental tone, relative to the pressure at a screened microphone in the discharge duct. Zero phase was arbitrarily shifted to coincide with the position of minimum acoustic pressure amplitude, nevertheless it is apparent that at this point the pressure phase undergoes a rapid transition. Similar results were obtained at the second and third harmonic frequencies, although the pressure amplitudes were comparatively flat in the central regions. The distributions shown in Fig. 5 were typical over the range of blade passing frequencies tested, between 136 and 272 Hz.

Insertion loss improved noticeably when the divider was attached to the cutoff at the point of minimum acoustic pressure. Inferior performance resulted with the divider located elsewhere; only 3 dB insertion loss resulted at 272 Hz with the divider completely removed and the piston faces coupled by a steel plate to facilitate single tube operation.

3.3 EFFECT OF PERFORATE POROSITY

The best cutoff surface porosity appeared to be a reciprocal function of frequency, as suggested by Neise and Koopmann (1980). A 47% perforate produced the greatest insertion loss at 272 Hz for both the 8-blade and 16-blade rotors. A porosity of 40% reduced insertion loss by 3 dB at that frequency, but was more appropriate for a 544 Hz fundamental or harmonic tone.

Figure 6 shows the noise spectra for the 8-blade rotor, with solid and 47% porosity cutoffs installed. The blade passing frequency, second and third harmonic fell 13 dB, 7 dB and 5 dB respectively, while non-tonal noise increased slightly. The non-tonal noise amplitude became greater with increasing cutoff porosity.

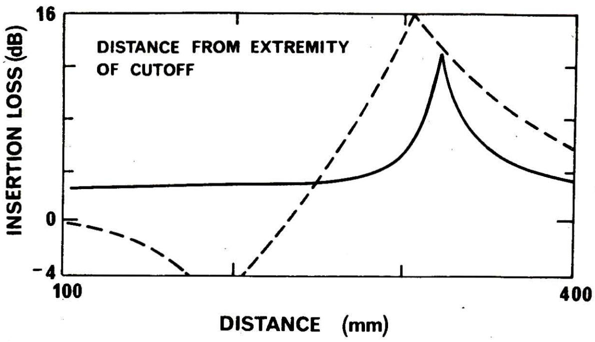

Fig.7 Insertion loss versus piston position measured from the of extremity of the perforated cutoff; 272 Hz ——— ; 544 Hz ----.

3.4 RESONATOR TUNING

Figure 7 shows insertion loss as a function of distance between the piston and the end of the cutoff. The 272 Hz curve refers to the 8-blade rotor and bottom piston with a 47% perforated cutoff, while the 544 Hz curve is that for the 16-blade rotor and top piston with a 40% perforate. At a given frequency, whether a fundamental or harmonic, the optimum piston positions were the same for both rotors. Tests with the 16-blade rotor revealed that the optimum piston positions for the 40% and 47% perforated cutoffs were within 2 mm of each other at 272 Hz.

An interesting but confounding aspect of this investigation concerns the tuning of the upper and lower resonator tubes for the two rotors. Maximum insertion loss at 272 Hz was achieved with the top piston positioned only 100 mm from the perforate. The bottom piston position approximated quarter-wave tuning. At 544 Hz, the bottom piston was best positioned about 100 mm from the cutoff, with the top piston located about one half-wavelength from the cutoff. Insertion loss was considerably less when the perforate holes of the shorter tube were blocked, indicating that the entire perforate surface contributed to noise attenuation. Insertion loss was relatively insensitive to the length of the shorter tube which could be varied plus and minus 30 mm with little effect.

Experiments with the 272 Hz blade passing frequency revealed that withdrawal of the top piston caused the second harmonic amplitude to decrease. However, the blade passing tone increased at the same time, to eventually exceed the level generated by the fan with a solid cutoff.

3.5 FAN PERFORMANCE

Fan volume flow was measured with a flush static pressure tap on a BS848 conical inlet and the fan inlet total pressure was measured using a total pressure tube. The motor power (DC supply) was determined from the input voltage and current. These characteristics were recorded with solid and 47% perforated cutoffs installed. Differences in fan performance due to the inclusion of the resonator could not be detected.

4. DISCUSSION

The performance and optimum resonator configuration varied considerably for the different fans. A simple quarter-wave tube proved effective for Fan No. 1, while Fan No. 2 required a split resonator with tubes adjusted to different lengths. We can speculate that the normalised cutoff clearance (δ) determined the most appropriate configuration. Fan No. 1 had a particularly small cutoff clearance, and the cutoff was therefore well within the region swept by the pressure field revolving with the rotor. Interaction between this field and the cutoff is accepted as the cause of sound at the blade passing frequency. Fan No. 2 had a normalised cutoff clearance four times greater than Fan No. 1. The noise mechanism is thought connected with flow vorticity shed by the rotor blades. To compensate, the resonator must present different impedances at the upper and lower surfaces of the cutoff.

The resonator of Fan No. 2 had microphones attached to the piston faces and the inside surface of the cutoff in both chambers (not shown in Fig.4). Acoustic pressures measured by the microphones verified that in fact the resonator operated as two independent, rigid and completely sealed tubes. Holes in the 47% and 50% perforates were rectangular-shaped with aspect ratios of about 1:6, whereas other perforates had circular holes. The longer axes of the 47% and 50% perforates were parallel with the curvature of the cutoff. It is thought that the shape of the perforations had little influence on performance; certainly with respect to resonator tuning, since Section 3.4 reports similar trends with the 47% (rectangular hole) and 40% (round hole) perforates.

Results depicted in Fig. 7 show that tuning at 272 Hz was more sensitive to piston position than at 544 Hz and further, imply that the 272 and 544 Hz tones required quarter- and half-wavelength tuning respectively. At least with respect to Fan No.2, it appears that the term 'quarter-wave' is inappropriate for this method of noise silencing.

5. CONCLUSION

At present there are no design procedures for this method of silencing centrifugal fan tonal noise. Such inconvenience may be offset by the advantages of these silencers, mainly;

(a) they are applicable at frequencies above cut-on of the higher order modes in the inlet and discharge ducts,

(b) they can sometimes reduce the blade passing tone and harmonic components at the same time.

The normalised cutoff clearance is thought to influence the optimum resonator configuration. This, and the fact that piston control may be unavoidable where the fan speed varies, could limit resonator use to large installations which can support the cost of experimentation.

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Acoustics in the Eighties

THE EFFECT OF CONFIGURATION OF SOUND INTENSITY MEASUREMENTS ON SOUND POWER DETERMINATION

N. Tandon

THE EFFECT OF CONFIGURATION OF SOUND INTENSITY MEASUREMENTS ON SOUND POWER DETERMINATION

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ABSTRACT

Sound intensity measurements have been performed on a calibrated sound power source in an ordinary room. The measurements were performed using a two microphone probe, dual channel real time analyzer and a microcomputer. The two measurement methods, namely scanning and point by point are compared. The effect of the distance of the control surface from the source for these two methods is investigated. Different configurations of points were used on the cubical box type control surface around the source to study their effect on sound power determination and to find optimum area per point for this surface for sound intensity measurements. Fairly accurate results are obtained at larger distances from the source. The measurement results indicate that if sufficient number of points are taken on the control surface then point by point method gives better results as compared to the scanning method.

1. INTRODUCTION

Sound intensity measurements are increasingly being used for the determination of sound power. The main advantage of using these measurements over the conventional method of sound power determination is that no special environment i.e. anechoic chamber or reverberation room is required for these measurements. Fairly accurate sound power levels can be obtained in an ordinary room. So the measurements can be made in situ. Sound intensity measurements can also be made in the presence of a steady background noise.

The acoustic intensity vector component I_r in the direction r using two closely spaced microphones, can be estimated from the imaginary part of the cross spectrum between the two microphone signals, based on a finite difference approximation

$$I_r = \frac{I_m \, G_{12}}{\omega \, \rho \, \Delta r} \tag{1}$$

where, G_{12}: The cross spectrum between the two microphone signals.

ω : angular frequency

ρ : density of air

Δr : distance between the two microphones

The acoustic power, W passing through the surface S is

$$W = \int_S \bar{I} . \overline{dS} \tag{2}$$

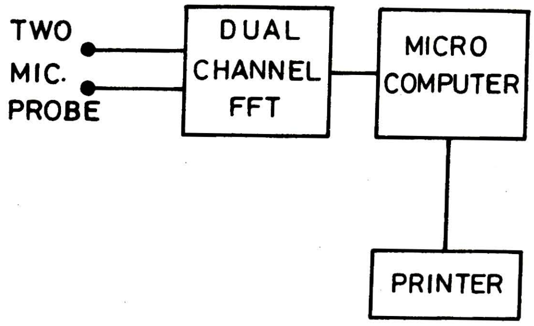
where $\bar{I}$ is the intensity measured on an element of area $\overline{dS}$ on surface S enclosing the sound source.

The purpose of the present work is to study the influence of measuring distance from the source and the number of measurement points, on the sound power determination. The two methods of sound intensity measurement namely point by point and scanning are also compared. A sound source producing known sound power of 80dB was used for these measurements. All the measurements were performed in an ordinary room of 96.5 m^3 volume (6.4x3.96x 3.81 m).

2. INSTRUMENTATION AND MEASURING TECHNIQUE

Sound intensity measurements have been performed with the help of a two microphone probe, dual channel real time analyzer and a desk top microcomputer to carry out final analysis and present sound power spectrum in one third octave bands (Fig. 1). The probe consists of two closely spaced phase matched microphones separated by a spacer (12 mm) having a flat amplitude response. The useful frequency range for the analysis of the data was taken as 125-5000 Hz.

The sound power of the source has been determined by fully enclosing it in a box shape control surface as shown in Fig. 2 and measuring the intensity component normal to this surface.



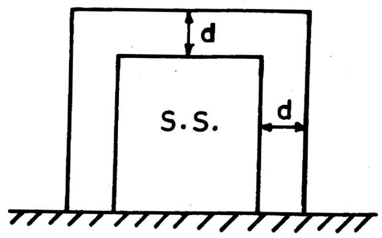


Fig.1 Block Diagram of measurement set-up for sound intensity

Fig.2 Control surface at distance d from sound source, S.S.

In the scanning method the probe was moved slowly on this surface with a uniform velocity. Each surface was divided into several equidistant points in the point by point method and the intensity was measured at each point. The average value of sound intensity level obtained was used to get the sound power level. The reactivity of the sound field, defined as the difference in sound pressure and intensity level at a point, influences the intensity measurements. So the average reactivity for the room at distancens of 5,10,25,35,50 and 75 cm from the source was determined. The results are shown in Fig. 3a and 3b. In the case of 75 cm distance two of the surfaces of the control box were very near to the walls of the room.

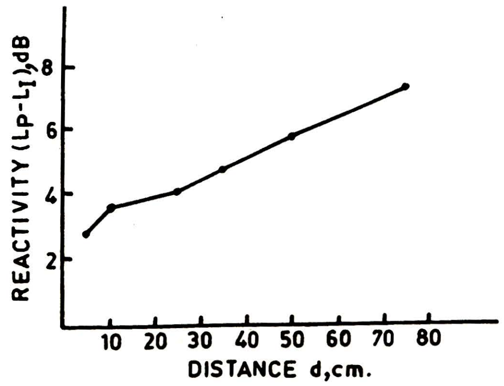


Fig.3a Reactivity of the sound field Vs. distance d from the source surface.

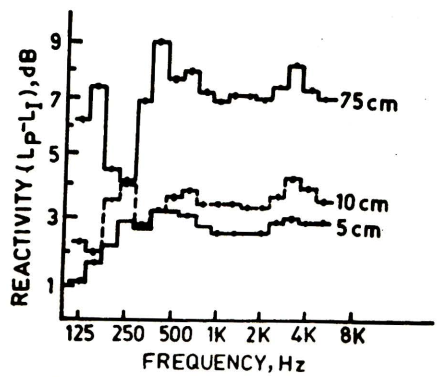
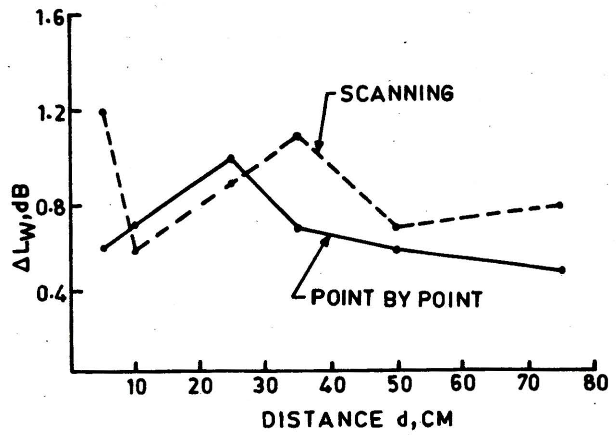


Fig.3b Frequency Spectrum of Reactivity at Distance d of 5,10,and 75 cm from the surface of the source

3. RESULTS

3.1 THE EFFECT OF DISTANCE FROM SOURCE

To study the effect of the measurement distance from the source, the control surfaces of the rectangular box were defined at a distance of 5,10,25,35,50 and 75 cm. Average sound intensity was measured at these surfaces by scanning and point by point method to obtain the sound power level of the source. The difference ΔL_W between the known sound power level and the measured levels against the measurement distance, d is plotted in Fig. 4. The results show small errors even at larger distances. In the near field also at least the point by point method gives small error.



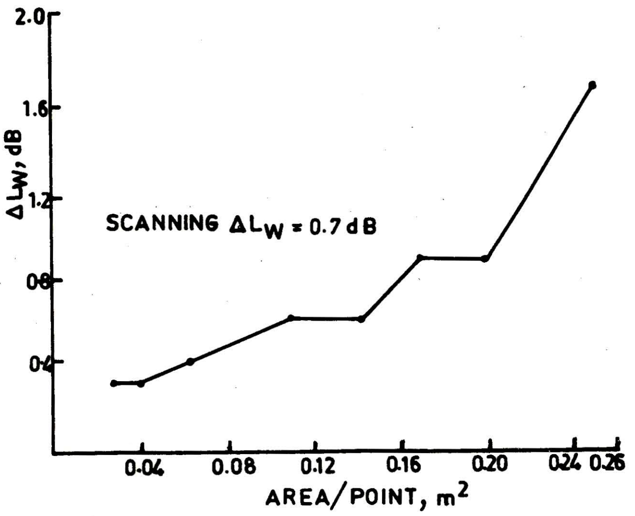


Fig.4 Deviation of measured sound power level from the calibrated level, ΔL_W as a function of distance from source.

Fig.6 Deviation of measured sound power level from the calibrated level, ΔL_W as a function of area per point.

3.2 CONFIGURATION OF POINTS AND POINT DENSITY

Measurements were performed at a $1m^3$ cubical box control surfaces around the source using the measurement points as shown in Fig. 5. As expected the first configuration of only four points on corners gives a large error of 1.7 dB whereas addition of one point in the middle of square surface reduces it to 0.9 dB. The error, ΔL_W versus area per point is plotted in Fig. 6 for these measurements. The plot shows that for an error of 1 dB approx. $0.21 m^2$ area/point is required. But for good results at least $0.04\ m^2$ area/point should be used. Sound power by scanning method for the same cubical box surfaces gives an error of 0.7 dB.

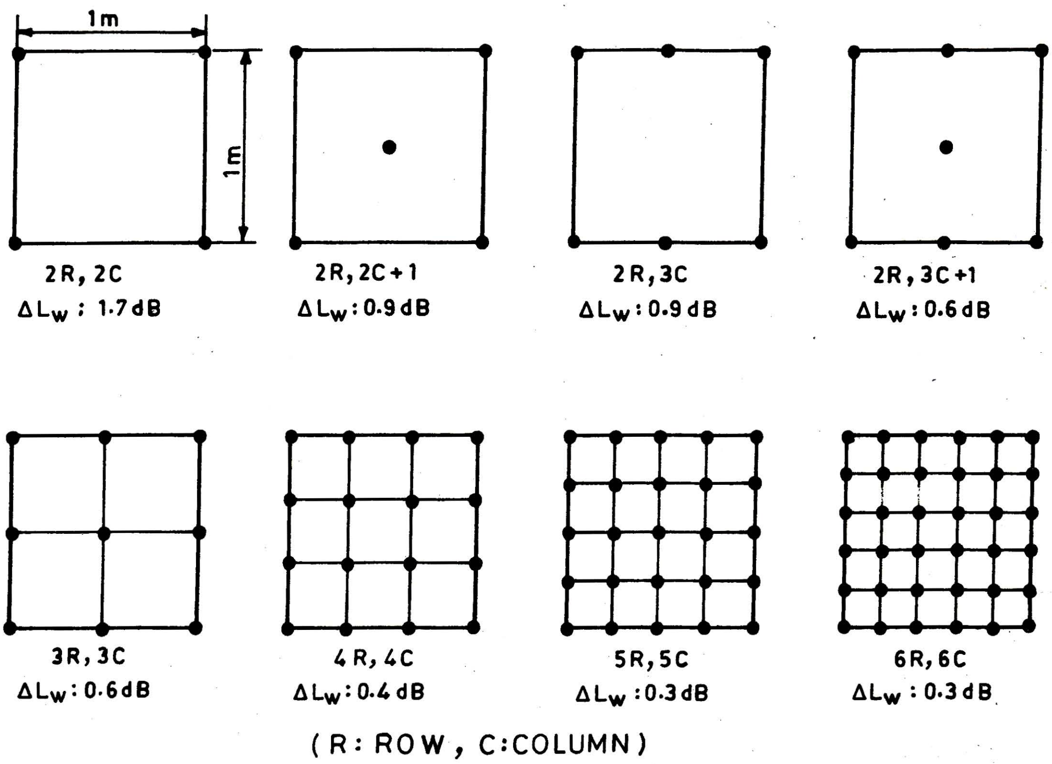


Fig.5 Configuration of points on the surfaces of the cubical box of 1 Meter and the respective ΔL_W levels obtained.

3.3 SCANNING VERSUS POINT BY POINT METHOD

The results of Fig. 4 show that in general the sound power levels obtained by point by point method are more accurate as compared to the scanning method. Usually measurements by point by point method take much more time. The results of the measurements on the cubical box surfaces shown in Fig. 6 also indicate that more accurate results can be obtained by point by point method provided sufficient number of points are taken on each surface.

4. DISCUSSION

All the measurements give a positive value of the error ΔL_W. One possible explanation for this could be that the sound source is producing noise in the frequency range of 100 Hz to 10 KHz where as sound intensity measurement and analysis has been performed in the frequency range of 125 Hz to 5 KHz only because of larger errors in this measurement technique at lower and higher frequencies.

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Acoustics in the Eighties

Session 4B: Measurement Technology II

Acoustics in the Eighties

OBSERVATION OF ACOUSTIC AMPLITUDE AND IMPEDANCE BY LAST DOPPLER METHODS

M.R. Davis

OBSERVATION OF ACOUSTIC AMPLITUDE AND IMPEDANCE BY LASER DOPPLER METHODS

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ABSTRACT

This paper describes the acoustical use of the methods of laser Doppler anemometry. Seeding particulates suitable for laser-Doppler use are generally found to have satisfactory and accurate acoustic response in air and in water. The achievable signal to noise ratio in laser Doppler systems places a constraint on system resolution, and methods based on analysis of Doppler signal spectra are therefore preferable. Such an approach thus provides statistical average information relating to the acoustic field. Under pure tone acoustic conditions the system dynamic range is determined by the requirement to analyze the sidebands generated in the Doppler spectrum. However acoustic amplitude can be determined to 0.1 dB or better. Determination of the phase between acoustic pressure and velocity is possible by simultaneous amplitude and frequency modulation of the Doppler signal combined with the pressure signal, and regression analysis of asymmetric sideband peaks allows determination of the phase to within a standard error of 1°.

ACKNOWLEDGEMENT: This paper is based on collaborative work with the CSIRO Division of Applied Physics, and the substantial work of that division in the development of these methods is acknowledged.

1. INTRODUCTION

The problems involved in applying the laser Doppler anemometer to acoustic measurements can be conveniently grouped into several categories. Firstly, it is essential that light scattering particles are embedded in the acoustic medium and that such particles are of an appropriate size to activate the Doppler system whilst ensuring that these particles move with the fluid as it undergoes acoustic disturbance. Secondly, it is necessary to make provision for absence of a substantial mean velocity in many acoustic applications and to determine under what conditions measurements are possible. Here it must be emphasized that one of the fundamental diffficulties associated with the operation of any laser Doppler anemometer system is the ratio of the Doppler signal to noise signals inherent in the sensing electronics. In the acoustical application, this presently restricts operation to a limited dynamic range of pure tone acoustic fluctuations. Thirdly, special techniques have to be employed in the processing of Doppler signals if acoustic power measurements are to be undertaken. Each of these aspects will be dealt with in the subsequent sections of this paper.

2. REQUIREMENTS FOR SEEDING WITH LIGHT SCATTERING PARTICLES

The light scattering particles are subject to two distinct requirements: firstly they must respond to the acoustical velocity fluctuations in the medium so that they effectively move dynamically with the fluid, and secondly they must be of a size appropriate to effective operation of the laser Doppler system. As discussed by Taylor (1976), the response of a spherical particle of diameter d and density ρ, immersed in a fluid of viscosity μ at frequency f gives rise to a ratio of particle velocity (v_p) to velocity of the fluid (v_f) given by

$$v_p/v_f = [(\pi \; r^2 \rho_p f/9\mu c)^2 + 1)^{-\frac{1}{2}} \tag{1}$$

where c is the Stokes-Cunningham correction factor which is very close to unity for the conditions to be considered here (see Brandt et al., 1937). It follows directly from this relation that the attenuation of seeding particle motion relative to that of the medium can be computed for various sizes of particles and frequencies of excitation. Figure 1 illustrates this for air and for water. It can be seen that particles smaller than about 3 microns must be used in air at a frequency of 1 KHz if errors due to the particles exhibiting a reduced response are to be less than 0.1 dB, whilst in water at 1 Khz particles smaller than about 30 microns must be used for this same error and frequency requirement.

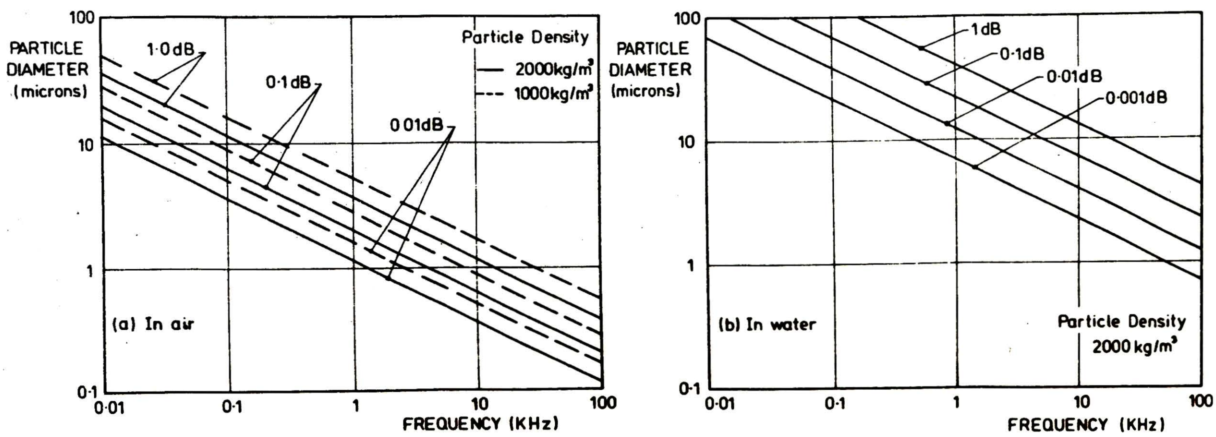

Figure 1. Error due to particle slip relative to medium

The required size of the scattering particles is also influenced by the optical parameters of the laser Doppler sensing system. As illustrated in figure 2 this comprises a split laser beam arranged so that the two beams intersect at an angle of θ. Since light scattered from the particles is sensed in the forward scatter direction in the present system (although backscatter systems can also be used) the frequency shift between the two optical beams due to the Doppler effect on the scattered light due to motion of particles at velocity v_p at right angles to the beams is given by Δf,

$$\Delta f = [\sin(\theta/2)] |v_p/\pi\lambda| \qquad (2)$$

where λ is the wavelength of the light. An important practical requirement is that the detected light fluctuations due to the particle motion is a significant fraction of the total scattered light so that the signal to noise ratio of the sensing system is acceptable. For this to be the case, the particles must have a diameter not substantially in excess of the wavelength of the illimunating beam. In the case of a helium-neon laser it is therefore necessary to use particles smaller than wavelength, which in this case is 0.6328 microns. Thus it is seen from figure 1 that if particles of this order of size are used quite small errors due to particle slip will arise, being less than 0.01 dB in air at 10 KHz or less than 0.01 dB in water at 100 KHz.

3. OPERATION OF THE LASER DOPPLER SYSTEM

As shown in figure 2 the laser Doppler system used here is operated in forward scatter configuration, the light scattered from particles illuminated by the two intersecting beams being detected by a photomultiplier located on the system axis. From equation 2 it is seen that under conditions of acoustical observations in a stationary medium periodic reversal of the velocity of particle motion is detected by the frequency of the scattered light fluctuations as particles move relative to the incident illumination in magnitude only. For this reason it is necessary to insert two Bragg cells in the incident beams with a small differential frequency (f_B) so that the detected signal shows a fluctuation of illumination at frequency f_B without particle motion. When the particles move under acoustic excitation the frequency of observed optical fluctuations (f_D) then moves up and down about f_B indicating the velocity of motion without ambiguity of direction,

$$f_D = f_B + (2\sin(\theta/2))(v_p/\pi\lambda) \qquad (3)$$

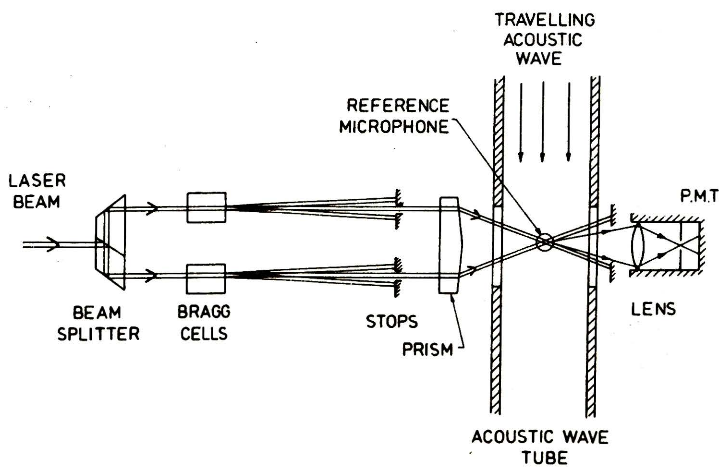

Figure 2. Components of measuring system

In the experiments to be described the Bragg cells operated at 40 MHz and 40.02 MHz, giving rise to a differential frequency f_B of 20 KHz. As with any laser Doppler system, it is then necessary to relate fluctuations of f_D (above and below 20 KHz in this case) to the velocity of movement of the particles. It can be seen that the system is also sensitive to small mean drift currents in the acoustic medium since a velocity of 0.01 m/s will produce a mean frequency shift of 5.49 KHz. Since acoustic velocity magnitudes are generally quite small it is virtually inevitable that a system designed for acoustical measurement becomes sensitive to quite small convection currents in the acoustic medium, and it might be necessary, under difficult conditions where external convection currents intrude, to shield the detection point by an acoustically transparent container. In this regard the maintenance of an acceptable level of scattering particles in the detection volume needs to be carried out carefully as turbulent motions of the acoustic medium are usually caused by the injection of the necessary smoke particles. It is generally necessary to allow such mixing disturbances to decay before commencing acoustic observations, as the random velocity fluctuations due to the turbulence have the effect of causing the Doppler signal to be modified from that with a sharp spectral peak at f_B to one with a broad peak in its spectrum centered at f_B before any acoustic fluctuations are applied. It is in practise quite possible for the Doppler peak in the spectrum to become unidentifiable if turbulent mixing effects broaden it and reduce its level close to that of the background noise in the detected signal due to such effects as electronic noise and Brownian motion of the particles. However, with reasonable precautions it was possible to observe the Doppler system peak at close to 20 KHz for the system in use without excessive mean frequency shift or broadening of the peak due to mean motion and turbulence respectively.

The discussion in the remainder of this paper will be confined to consideration of measurements under pure tone acoustic excitation. Under such conditions the Doppler signal shows the characteristics of a harmonic signal with sinusoidal frequency modulation, the spectrum of which contains the Doppler signal energy distributed into a set of sidebands. As the amplitude of the harmonic acoustic signal is increased, so the energy of the Doppler signal is distributed into an increasing number of sidebands, with an associated general decrease of energy in any one sideband.

4. OBSERVATION OF ACOUSTIC AMPLITUDE

The output signal from the photomultiplier as described by equation 3 is a harmonic signal whose frequency (f_D) is modulated about the differential frequency of the two Bragg cells (f_B) by an amount proportional to the velocity of the particles in the observation volume. The output signal can thus be written as having a frequency given by

$$f_D = f_B + [2\sin(\theta/2)/\lambda]V \cos 2\pi f_a t \tag{4}$$

where f_a is the acoustic frequency and V is the amplitude of the acoustic velocity fluctuation. It follows from standard frequency modulation analysis (see Hund, 1942 for example) that the output of the photomultiplier is of the form

$$E(t) = E_o \sin 2\pi(f_B t + \beta\sin 2\pi f_a t) \tag{5}$$

where β is the index of frequency modulation given by

$$\beta = [2 \sin(\theta/2)/\lambda].(V/f_a) \tag{6}$$

Spectral analysis of the photomultiplier signal given by equations (5) and (6) then shows that the fluctuating signal energy is moved from the single peak at the centre frequency f_B into a set of sidebands to either side of f_B. As the modulation index β is increased so the energy of the Doppler signal is distributed over an increasing number of sidebands, the sidebands being spaced at frequency intervals equal to the acoustic frequency f_a:

$$E(t) = E_o\{J_o(\beta)\sin 2\pi f_B t + J_1(\beta)[\sin 2\pi(f_B + f_a)t$$
$$+ \sin 2\pi(f_B - f_a)t] + \ldots + J_n(\beta)[\sin 2\pi(f_B + n\, f_a)t$$
$$+ \sin 2\pi(f_B - n_a)t] + \ldots\} \qquad (7)$$

where $J_n(\beta)$ are Bessel functions of the first kind and order n. We thus see that analysis of the amplitudes of the peaks in the frequency spectrum can form the basis for determination of the acoustic amplitude. This approach was first described by Taylor (1976, 1981) and has the advantage that the acoustic amplitude is determined in terms of the beam intersection angle (θ), the optical wavelength of the laser source (λ) and the acoustic frequency (f_a). It is therefore possible to establish acoustic amplitudes in terms of geometric, wavelength and frequency references, and thus to relate acoustic absolute standards to standards for measurements of these references.

Examples of the spectrum observed from the output of the photomultiplier are shown in figure 3. It is clear that the energy of the Doppler signal is distributed to an increasing number of sidebands as β increases, and that cyclical variations of the centre and low order sidebands occur due to the periodical variations in the corresponding low order Bessel functions. It is also clear that the signals contain a significant background noise due to particle motions in the gas and other random noise effects such as electronic noise of the photomultiplier. It was found necessary to make observations about 1 minute approximately after injecting smoke into the acoustic wave tube used in these tests (45mm x 50mm cross section) to allow turbulence induced by smoke injection to decay and for the Doppler peaks to become sharp. Cigarette smoke was found to be a relatively easy source for these tests with a particle size of approximately 0.6μm and relative density of approximately 1.0, giving a slippage error of 0.0002 dB at 1000 Hz and 0.000002 dB at 100 Hz. It was found that the smoke eventually disperses and deposits on the tube walls and that a clear Doppler signal could be maintained for about 5 minutes approximately without injecting additional smoke into the air in the tube.

A number of techniques can be devised to determine the modulation index from the set of amplitudes of the peaks in the spectrum. A satisfactory approach was found to involve appreciable computation by means of a least squares regression to the complete set of peaks using library subroutines for the

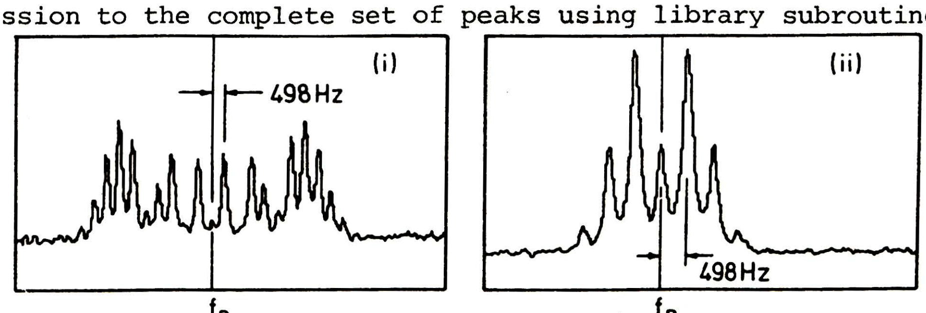

Figure 3. Spectra of laser Doppler system with travelling acoustic waves (f_B = 20 KHz, f_a = 498 Hz) (i) SPL = 100.8 dB, β = 8.296

(ii) SPL = 87.8 dB, β = 1.863

Bessel functions and the IMSL subroutine LMM for the regression. In this way the complete set of peak amplitudes is used as the data base for determination of the modulation index β . Simpler but less accurate approaches could be developed by using only selected peaks (perhaps the largest) in the spectrum and evaluating β by a simplified iteration. Since the number of sidebands containing significant energy is closely related to β , first estimates of for iterative purposes are not difficult to devise.

The acoustic wave tube used in these initial tests was arranged with a non reflecting termination so that the conditions were those of a plane progressive wave. It follows therefore that the sound pressure level (L) is related to the modulation index of equation 6 by

$$\mathrm{S\ P\ L\ (dB)} = 20\ \log_{10}(\rho c \lambda f_a \beta / 2\sqrt{2}\ p_{ref}\ \sin\theta/2) \tag{8}$$

where p_{ref} is the standard reference acoustic pressure (20μPa). Comparison with a reference calibrated microphone lcoated in the tube showed that the sound pressure level could be determined to 0.1 dB or better, with the major contributing uncertainty being the average standard error for the regression analysis to determine β which corresponded to an error in sound pressure level of 0.06 dB. It should be noted that the background noise in the system was subtracted from the observed peak values observed on the basis that it was not correlated with the Doppler signals. It was found convenient to tape record the Doppler signals and replay for spectral analysis on account of the limited duration of 5 minutes obtained before the smoke particles dispersed.

The dynamic range over which laser Doppler measurements are practicable is essentially determined by the requirement to determine the modulation index from the set of observed peaks. Modulation index values less than 0.5 are difficult to handle because the Doppler signal energy does not move sufficiently out of the central peak at f_B into the sidebands. Modulation index values in excess of 50 are difficult to handle because the acoustic energy is then distributed into a large number of peaks all of which are quite small. It follows that the useful dynamic range is for $0.5 < \beta < 50$, and figure 4 shows how this translates to a useful dynamic range in terms of sound pressure level

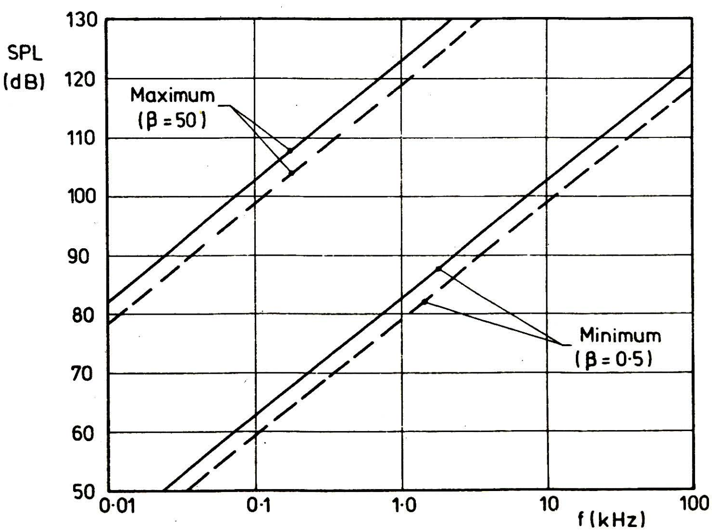

Figure 4. Useful dynamic range of laser Doppler system
Solid line: air (dB re. 20μPa) Dashed line: water (dB re 0.1 pa)

depending upon frequency. These results of course apply for a plane progressive wave where the acoustic velocity is everywhere related to the sound pressure by $V_f = p/\rho c$. Under standing wave conditions or conditions where there is some intermediate combination of acoustic waves travelling in opposite directions, the acoustic velocity and pressure are no longer related in this simple manner (most obviously at the velocity node in a standing wave) and it becomes necessary to examine the phase between acoustic velocity and pressure as an indication of the acoustical fluctuations. This is considered in the next section.

5. OBSERVATION OF ACOUSTIC IMPEDANCE

For acoustic disturbances moving in a line perpendicular to the laser Doppler anemometer axis, conditions intermediate between true standing waves and unidirectional progressive waves can be represented by the general solution of the wave equation, the pressure being

$$p = Ae^{j\omega(t-x/c)} + Be^{j\omega(t+x/c)}.e^{j\phi} \tag{9}$$

and velocity

$$v = -(\frac{A}{\rho c})e^{j\omega(t-x/c)} + B(\frac{B}{\rho c})e^{j\omega(t+x/c)}\ e^{j\phi} \tag{10}$$

where ϕ is the phase between the two component waves at $x = 0$. If we observe the pressure and velocity amplitudes (P and V) at $x = 0$, and if we also observe the phase (ϕ_m) between the pressure and velocity fluctuations at $x = 0$, then it becomes possible to determine the relative amplitudes of the forward and backward moving waves and their phase at $x = 0$ from the observed parameters.

$$(B/A = [\bar{P}^2 + 1 + 2\bar{P}\cos\phi_m)/(\bar{P}^2+1 - 2\bar{P}\cos\phi_m)]^{\frac{1}{2}} \tag{11}$$

and also the phase (ϕ) between the two constituent waves

$$\phi = \mathrm{Tan}^{-1}(\bar{P}\sin\phi_m/(1+\bar{P}\cos\phi_m)) - \mathrm{Tan}^{-1}(\bar{P}\sin\phi_m/(\bar{P}\cos\phi_m - 1)) \tag{12}$$

where $\bar{P} = P/\rho cV$. This determination of the two acoustic waves comprising the total acoustic disturbances could alternatively be viewed in terms of the acoustic impedance. Either way, the acoustic conditions can be specified by measurement of pressure and velocity amplitude and their relative phase (ϕ_m). Figure 5 illustrates these relationships, conditions being for an exact standing wave along $B/A = 1$ and tending towards undirectional waves for $B/A \to \infty$ or 0.

The measurement of acoustic impedance requires a microphone to be used in conjunction with the laser Doppler system. However bearing in mind the limited signal to noise ratio which can be achieved with the laser system, it is desirable to determine the phase ϕ_m by direct spectral analysis methods rather than by attempting frequency to voltage conversion to generate a velocity signal. In order to achieve this a method has been devised (Davis and Hews-Taylor, 1986) whereby the Doppler signal is multiplied by the pressure signal, the latter being biased so as to always remain positive. The resulting signal created is thus, normalising the multiplying pressure signal to unit amplitude,

$$E_\phi(t) = E(t)(1 + \sin(2\pi f_B t + \phi_m)) \tag{13}$$

It can be seen that the spectrum of this signal (see equation 7 for E(t)) is again comprised of a number of sidebands, the nth sideband being $E_n \sin 2\pi(f_B + n\ f_a)t$, where

$$E_n = E_o[J_n^2(\beta)(1 - \frac{n\sin\phi_m}{\beta})^2 + (J'_n(\beta)\cos\phi_m)^2]^{\frac{1}{2}} \quad (14)$$

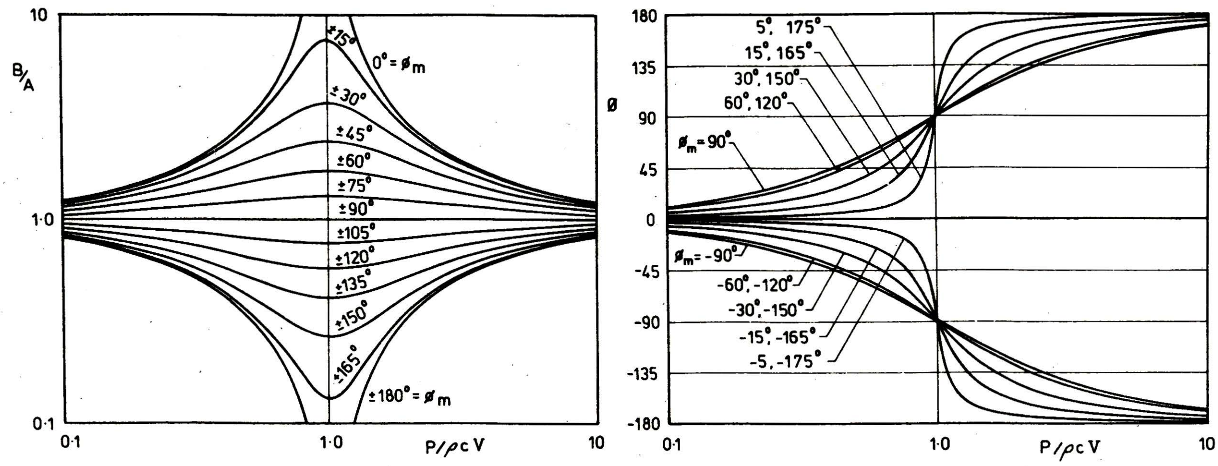

Figure 5. Dependance of ratio of component acoustic amplitudes and phase upon observed pressure, velocity and phase.

The effect of multiplication by the biased pressure signal is that when the pressure is a maximum, $E_\phi(t)$ is large and when pressure is a minimum $E_\phi(t)$ is near to zero. Thus if the Doppler signal has its maximum frequency (i.e. corresponding to maximum particle velocity) at times corresponding to maximum pressure and vice versa, the energy content of $E_\phi(t)$ at higher frequencies is increased. The result is that the sidebands are no longer symmetrical (unless the instant of zero pressure disturbance corresponds to either a maximum or minimum velocity).

In order to develop a measuring and analysis method capable of determining both β and ϕ_m by regression analysis of the spectral peaks, four spectral records of $E_\phi(t)$ were made. These four records were made with 0, $\pi/2$, π and $3\pi/2$ additional phase shifts applied electronically to the normalised, biased pressure signal. The four resulting spectra showed corresponding variations in the symmetry of the sidebands as shown in figure 6. By applying the IMSL routine LMM to the complete set of peaks shown in all four spectra, it was possible to determine both the modulation index β and the phase angle (ϕ_m) between pressure and velocity fluctuations. Extended tests in a standing wave tube for a variety of acoustic conditions obtained by adjusting the tube termination showed that the phase angle was determined in the regression with an average standard error of 1.1°. It should be noted that the results of figure 6 are influenced by phase shifts in the measuring microphones as well as the displacement between the laser measuring system and microphone. Close agreement (within 1°) was found between the observed phase lags and those expected (allowing for separately measured microhpone phase shifts), the main practical problem being the maintenance of reflection free conditions at the end of the acoustic wave tube.

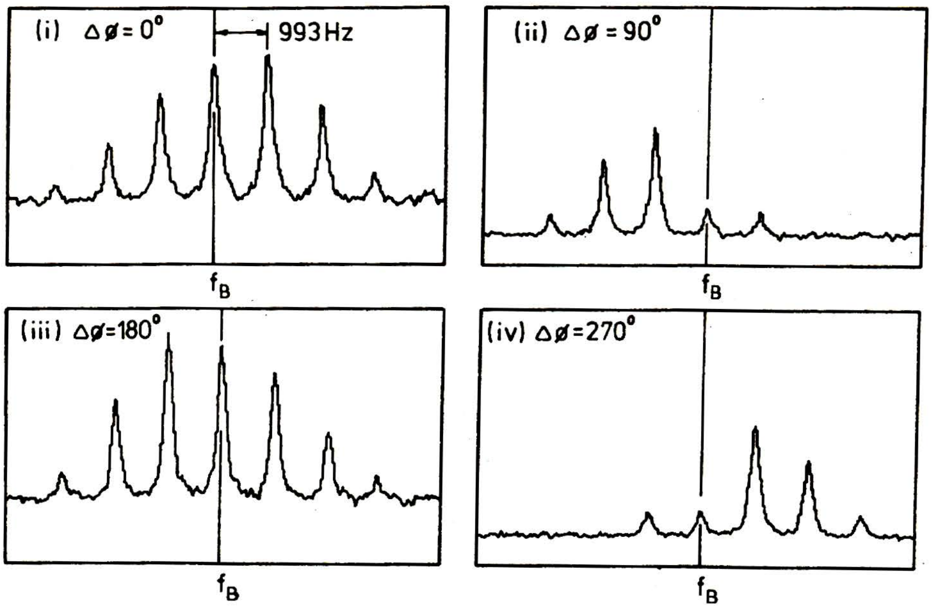

Figure 6. Spectra of pressure - Doppler product signals in plane progressive waves. Microphone displaced Δx beyond laser system to induce phase shift; multiplying pressure signal electronically shifted by $\Delta\phi$ f_B = 20 KHz, f_a = 993 Hz, SPL = 94.4 dB, β = 1.89, ϕ_m = -101.9°.

6. CONCLUSIONS

Observations of acoustic amplitude and phase are practicable using the laser Doppler method to within 0.1 dB and 1° standard errors. The system has a limited dynamic range under pure tone conditions, and practical considerations associated with the introduction of seeding smoke particles may also restrict operations in terms of system noise, turbulent contamination of signals and duration of observations. Particle slip does not introduce significant errors under typical operating conditions.

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SIMPLE METHODS FOR MEASURING REVERBERATION TIME AND ABSORPTION COEFFICIENT

Sun Chao and F. Fricke

SIMPLE METHODS FOR MEASURING REVERBERATION TIME AND ABSORPTION COEFFICIENT

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ABSTRACT

In this paper, measurement methods for absorption coefficient and reverberation time are briefly reviewed. A simplified method of measuring reverberation time is described which uses tape recorded repetitive white noise or filtered white noise pulses to excite the enclosure. A range of time intervals between pulses must be used in order that the maximum and minimum sound level during the pulse cycle may be read from a general purpose sound level meter. This method can be used for on-the-spot assessments of rooms with reverberation times greater than one second. The maximum error is less than 20%. A simplified in-situ method of measuring absorption coefficient is based on the coherence of incoming and reflected signals close to the surface of the material . If the microphone is very close to surface the sound level reflected from a perfectly reflective surface should be 6 dB greater than the sound level without the surface present and the sound level in front of a perfectly absorptive surface should be the same as that without the surface. The present experimental results have shown an acceptable degree of accuracy from 250Hz to 1kHz, for thin porous materials.

ACKNOWLEDGEMENT: We like to acknowledge the financial assistance given by Richard Heggie Associates for this research project.

1. INTRODUCTION

The reverberation time of rooms and the absorption coefficient of materials are both important parameters in room acoustics, transmission loss between rooms and noise control in general. If the acoustical conditions in buildings are to be improved then the existing acoustical conditions must be evaluated cheaply and effectively. This means that they must be under-taken with little or no specialized equipment so that Architects and Building Surveyors can undertake their own assessments of acoustical conditions.

2. REVIEW OF REVERBERATION TIME MEASUREMENT METHODS

Many attempts have been made at improving methods of measuring reverberation time. Sabine used a stopwatch to measure the duration of a decaying sound (Cremer 1982). Later objective measurement methods improved upon Sabine's subjective assessment. More recently research workers have concentrated on finding alternative methods of measuring reverberation time. Steady state methods of assessing reverberation time make use of the facts that the reverberant field sound pressure level depends on the sound source power and absorption in the room(Fothergill, 1982). Recently automatic measurement methods have shown a high degree of reliability (2.4% relative standard deviation, 1/1 octave) compared with the manual method(3.9% relative standard deviation, 1/1 octave) (Vigran, 1978 and Bjor 1979). A number of electronic circuits have been designed and used to measure reverberation time to an accuracy of ±0.01Sec.(Aljudi, 1986).

3. SIMPLIFIED SOUND LEVEL METER METHOD

Different measurement methods should be available to suit different purposes. For precise measurement, e.g. in absorption measurement, modern sophisticated instruments can be used. But there is also a considerable interest in simple methods which can be used with sufficient accuracy for noise control and transmission loss measurement and estimation. In transmission loss tests, for instance, a measurement error of ±30% is probably quite acceptable as it results in an error of only 1dB in T.L.. So for cases where there are no instruments to measure reverberation time, an alternative simple method is proposed and investigated. This method uses tape recorded repetitive white noise or filtered white noise pulses to excite the enclosure. A range of time intervals between pulses must be choosen in order that the maximum and minimum sound level during the pulse cycle can be read from a general purpose sound level meter, set on 'fast' response.

4. EXPERIMENTAL VERIFICATION

Reverberation time measurements were carried out in a reverberation room with different absorption in the room and in an office. The one inch microphone was placed in the test room and an omnidirectional loudspeaker was used. The experimental results are shown in Fig.1. When the reverberation time is more than 1 sec. the results show good agreement with those measured using a level recorder. The maximum difference was 0.5sec., when the reverberation time measured by standard level recording method was 2.4sec.. So the accuracy of this method is approximately within 20%. When the reverberation time is less than 1sec. the sound level meter cannot respond fast enough as the meter indicator shall decay by 10dB in a time of 0.5sec., or less, for "Fast" response(AS1259-1982, Sound Level Meters).

The main source of error in this method are due to the detector-indicator error of the sound level meter, fluctuation of the source input signal and the subjective reading error.

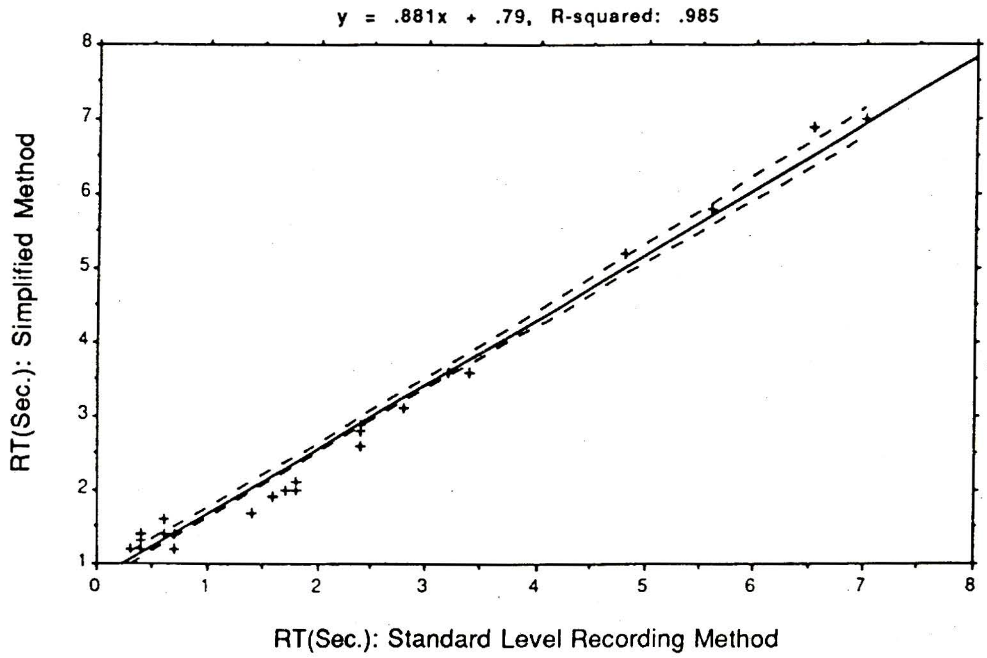

Fig 1. Comparison Between Simplified Method & Standard Level Recording Method of Measuring Reverberation Time

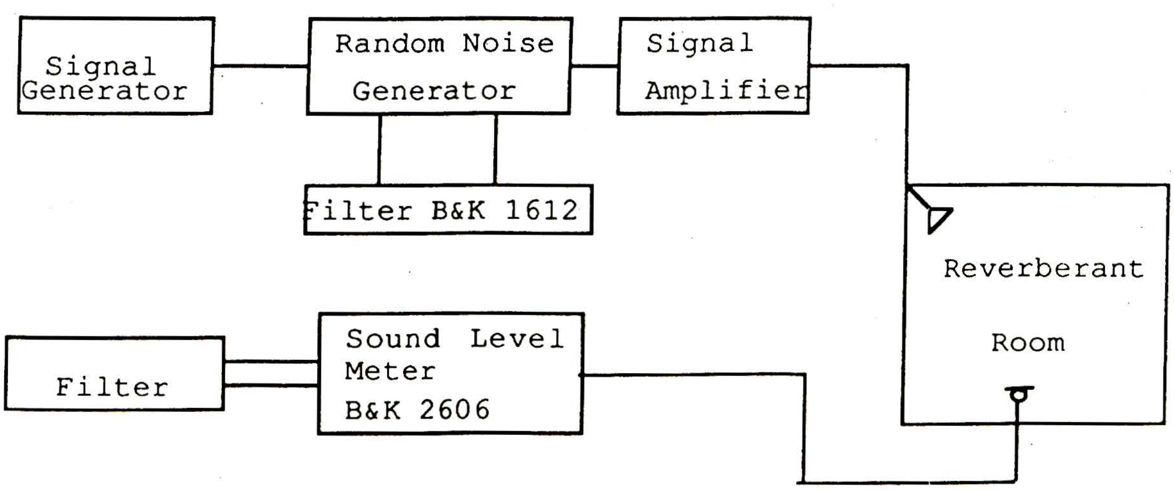

Fig 2. Diagram of the Experimental Arrangement for Measuring Reverberation Time using the Simplified Method

5. REVIEW OF ABSORPTION COEFFICIENT MEASUREMENT METHODS

The main methods used to measure sound absorption coefficients, the reverberation room method and the impedance tube method, have been used for many years and are well documented eg.((AS1045-1971) and (AS1935-1976). The reverberation time method will not be discussed further because it is unsuitable for in-situ measurments of absorption coefficient.

There have been some more recent developments of the impedance tube method which are of relevance to in-situ mesurements. Powell and Houten(1970) used a tone-burst technique to get higher sound level. A spark tube has been used(Methew and Alfredson, 1984) in which the Fourier transforms of input and reflected signals are used to calculate the transformed impedance. This method has the significant advantage that the impedance, and consequently absorption coefficients, can be obtained in a fraction of the time required when using the conventional steady state technique. A simple two microphone method of measuring the incident and reflected travelling wave components in a standing wave tube is described (Elliott, 1981) which gives good results. In this method there is no need for a long tube and the method can be used to measure absorption coefficients over bands of frequencies. A further development by Dunlop(1985), using a flanged end to the impedance tube, allows the impedance tube to be used without an enclosed sample.

To measure reflection coefficient of the ground , a wall or a test panel (provided the size of the panel is approprate so that diffracton and edge effects are negligible), both steady state in-situ and impulse in-situ methods can be used.

A correlation technique has been used (Hollin and Jones, 1977). Normal incidence absorption coefficients are derived from a comparison of the Fourier transforms of the correlation functions obtained by reflecting the noise from the sample and from a "perfect" reflector at the same distance from the microphone. The experimental results compare favourably with other methods. A cancellation method of absorption coefficient measurement makes it possible to obtain oblique incidence absorption coefficients. The reflection from a sample can be obtained by combining the outputs from two non-directional microphones through a phase inverter. By comparing the combined output with the direct sound, measured separately, the absorption coefficient can be estimated. The results for the normal incidence absorption coefficient, obtained by the cancellation method agree, to within a few per cent, with those obtained from the tube method(Yuzawa, 1975). Sound intensity measurement methods of absorption coefficient show good agreement with impedance tube results at high frequencies (Atwal and Bernhard, 1984).

Steady state methods, whether free or in a waveguide, tend to suffer from an extreme sensitivity to test geometry. Consequently the necessary degree of accuracy is difficult to achieve under field condition. Steady state free field measurements are not limited to normal incidence but are subject to contamination by spurious reflections and must be performed in anechoic conditions.

environments without disturbing the surface under test. The pulse method of measuring impedance and absorption coefficient has been studied by many researchers. The reflection coefficient is obtained on the basis of Fourier transforms of the direct and reflected pulses. Cepstral processing techniques allow the separation of superimposed pulses which occur in the acoustic reflection, where a reflected pulse is a delayed and distorted version of the incident pulse(Bolton and Gold, 1984). If both the source and receiver are placed near to the test sample, the effects of spherical waves on the reflection must be taken into account. Absorption coefficient can be obtained by a curve-fitting of experimental waveforms(Chen Tong and Zheng Minhua, 1986).

6. SIMPLIFIED IN SITU METHOD

If the incoming and reflected signals are coherent ie the microphone is as close as possible to the surface and the surface is perfectly reflective, then the sound level measured should be 6dB greater than the sound level without the surface present. If the surface is perfect absorbing the sound level should be the same as without the surface present. Thus the absorption coefficient of an unknown material can be estimated by comparing the sound level reflected from this surface and the maximum sound level reflected from a perfectly reflecting surface(6dB).

It is assumed in this method that the phase change of incident sound after reflecting from the surface is negligible. Provided the sample material is thin , of low absorption and is measured at low frequencies. If the maximum sound level differences of 6dB can be obtained and if sound levels can be measured to even 0.5dB, then the absorption coefficient can be measured, in-situ, to an acceptable degree of accuracy, ±0.1, for most purposes.

The sample was placed on an 18mm thick plywood surface in an anechoic room and the extension tube of a horn driver positioned so that it was just touching the sample surface. A quarter inch microphone (B&K 4135) was suspended horizontally so that its diaphragm was vertically above the end of the tube attached to the driver. The signal was filtered white noise. The sound level meter B&K 2606 was set on "impulse hold" and each reading was taken over ten seconds to provided a repeatable measurement. The readings of sound levels were taken at 250Hz, 500Hz, 1KHz, 2KHz,and 4KHz. Then the sample was placed in a 100mm diameter standing wave tube and the normal incidence sound absorption coefficients were measured at these frequences of interest.

7. RESULTS USING THE SIMPLIFIED IN-SITU METHOD

The experimental results are shown in Fig.3 and Fig.4. The sound level difference at high frequencies increased suddenly in Fig.3 because of the phase change of reflecting sound. So this coherence method appeared to be suitable for thin porous material and porous material at low frequencies. But absorption coefficients at higher frequencies could be predicted by extrapolation as most materials have a similar absorption versus frequency characteristics. The spherical wave spreading was another source of experimental error which sould be considered too.

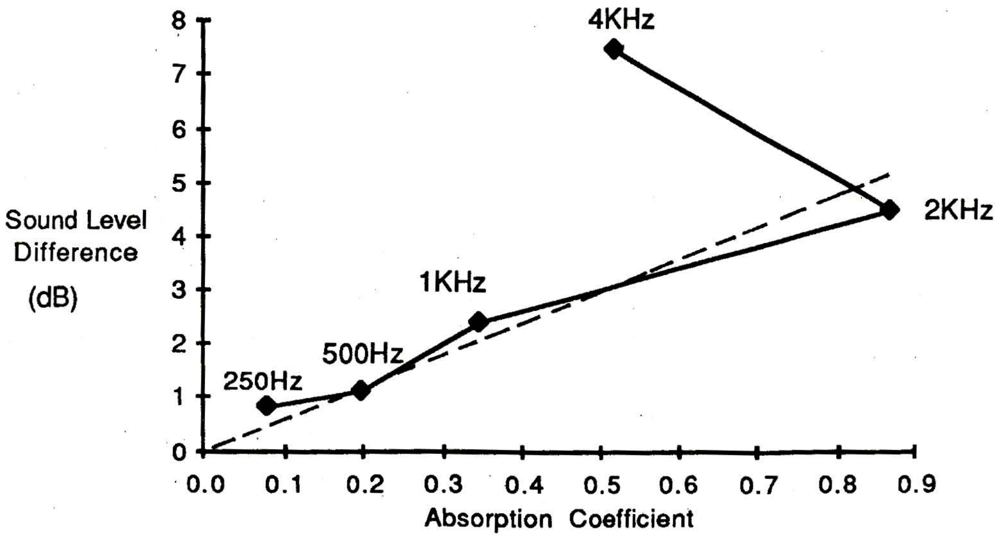

Fig.3 Correlation of the sound level difference tested in the anechoic room with normal incidence absorption coefficient. Absorbing material: 25mm fiberglass sheet. Sound source: horn driver with extension tube touching sample surface. Microphone 1/4'' diameter and distance 100mm above the sample surface.

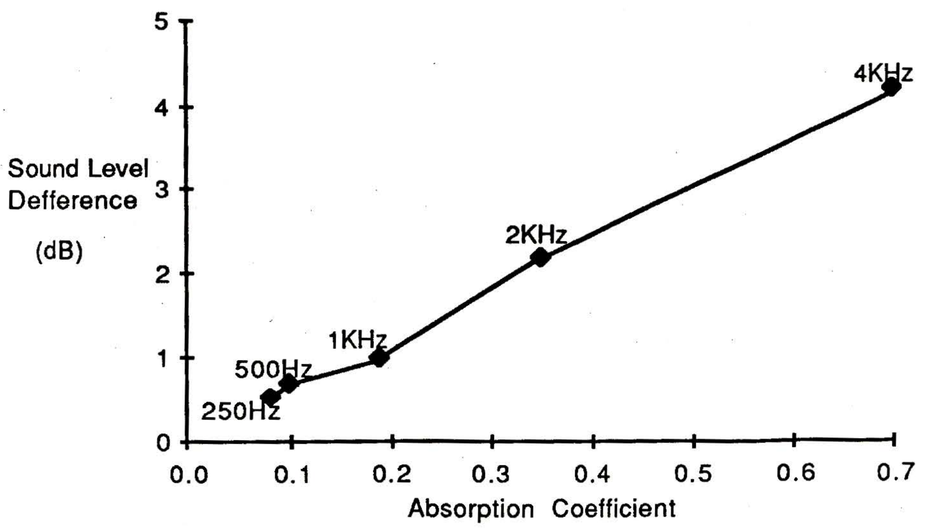

Fig.4 Correlation of the sound level difference tested in the anechoic room with normal incidence absorption coefficient. Absorbing material: 10mm jute backed carpet. Sound source: horn driver with extension tube touching sample surface. Microphone 1/4'' diameter and distance 100mm above the sample surface.

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THE DEVELOPMENT OF AN IMPROVED NOISE MONITORING SYSTEM

D. Watkins and P. Dale

THE DEVELOPMENT OF AN IMPROVED NOISE MONITORING SYSTEM

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ABSTRACT

The accurate long term monitoring of background sound levels is an important part of assessing industrial noise in Victoria. Recently the Victorian Environment Protection Authority developed a new noise monitoring system using a sound level meter and a Sharp PC-1600 portable computer.

The system has the advantage of being composed of readily available and relatively inexpensive components. It can operate completely automatically for long periods of time and has a very low power consumption. The monitoring system can be used for both short and long term measurements and provide a variety of noise indices as well as a chart recording of the measured noise level. The Sharp PC is compatible with IBM computers and data collected during field measurements may be transferred for storage and subsequent processing on a larger computer.

1. INTRODUCTION

The Victorian Environment Protection Authority has used State Environment Protection Policy No. N-1 to assess noise from industrial and commercial premises since 1981. The assessment procedure requires the determination of permissible noise levels based on the type of area in which the premises receiving the noise are situated. The determination of permissible noise levels takes into consideration the planning scheme zoning and the background sound level of the area.

Because of the variable nature of background levels it is necessary to take measurements over a period of at least 24 hours and preferably longer. A major problem concerning the measurement of background levels has been the availability of inexpensive portable noise monitoring systems capable of operating unattended for long periods of time.

Recently the Environment Protection Authority developed a new noise monitoring system consisting of a sound level meter and a Sharp PC-1600 portable computer. The major problems associated with the older systems used by the Environment Protection Authority have been overcome with the new system. The system may be used for monitoring background sound levels or a variety of other sources including industrial and traffic noise. The system is significantly cheaper than the older systems and has very low power consumption.

2. METHOD

Fig. 1 shows a schematic diagram of the basic noise monitoring system.

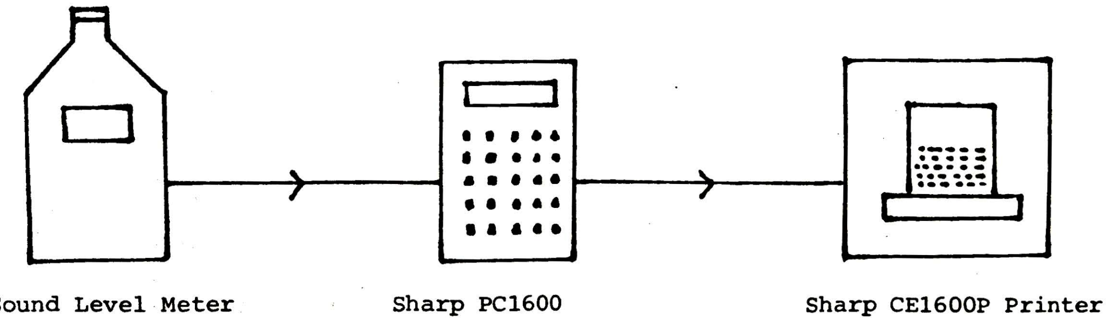


Fig. 1: Schematic diagram of basic noise monitoring system

The basic system consists of a sound level meter with a D.C. output, a Sharp PC 1600 computer and a Sharp CE 1600P printer. The Sharp computer was selected for the system because it contains an analogue to digital converter, is small in size and require little power to operate.

The sound level meter is used to provide an accurate D.C. voltage representing the level of the noise being measured. The output of the sound level meter is fed to the analogue to digital converter built into the Sharp PC 1600 computer. Since the RMS converter of the sound level meter has a logarithmic function, the D.C. voltage fed to the analogue to digital converter is directly proportional to the dB value.

The Sharp PC 1600 is programmed to sample the output of the analogue to digital converter into 0.25 dB steps over a range of 50 dB and store the samples in the form of an array. The computer has been programmed to sample at a rate of approximately 1 sample per second. The values are stored as follows:

(a) The digital code from the analogue to digital converter is used to select one of the 200 different locations (arranged in 0.25 dB steps) and a value of one is added to the selected location for each sampling.

(b) The equivalent energy value (determined from the antilog of the level divided by ten) is determined from each sample and added to the running total of the energy stored at a suitable location to give the function:

$$E = \sum 10^{L/10}$$

where: E is the total equivalent energy
L is the level of the sample

(c) A running total of the number of samples processed is continuously stored.

(d) Levels that fall outside the range of the system are stored in the lowest or highest locations. For example: with the system set to a range of 20-70 dB, levels below 20 dB are stored in the 20 dB location and levels in excess of 70 dB are stored in the 70 dB location.

The system is also programmed to print a chart of the sound level versus time. The Sharp 1600 is used to drive a Sharp CE 1600 P printer. The average of a number of samples are determined and plotted on the chart. It has been found in practice that the average of 9 samples provides a convenient sized chart with good resolution and also minimizes power consumption. Such a chart provides a complete picture of the noise being measured and enables a check to be made of the validity of the results.

The Sharp PC 1600 has an inbuilt clock that enables the results to be printed at suitable intervals of time. The system is programmed to carry out the following functions:

(a) Calculate percentile levels to the nearest 0.25 dB for the measurement period (normally hourly). For example the 90 percentile level is determined by accumulating successive locations starting from the base level until 10% of the total number of samples is reached. The location at this point represents the 90 percentile level. Similarly the system can be used to calculate any percentile level.

(b) Calculate the equivalent continuous sound level, Leq, for the measurement period (normally hourly). The Leq is calculated from the total number of samples and the sum of the equivalent energy level from the following formula:

$$\text{Leq} = 10 \log_{10} \frac{1}{N} E$$

where N is the total number of samples
E is the total equivalent energy

(c) At the end of the measurement period the percentile and Leq results are printed, the chart axis are drawn for the next measurement period and the time and date are printed.

3. ANALOGUE TO DIGITAL CONVERTER

The internal analogue to digital converter of the Sharp computer was found to be affected by internally generated jitter or noise creating significant errors in the digital output. It was also found that the converter was slightly non-linear at low levels. Two systems were developed to overcome these problems.

3.1 GENERAL PURPOSE SYSTEM

A general purpose system was developed using the Sharp internal analogue to digital converter. The jitter or noise internally generated in the analogue to digital converter was found to cause random errors. By averaging 20 samples the random error was greatly reduced. The non-linearity of the analogue to digital converter is not sufficient to seriously affect the results. However, the program can be written to include an off-set to overcome the non-linearity problem. It has been found that this system provides sufficient accuracy for most purposes.

3.2 PRECISION SYSTEM

The basic system was modified by adding an external analogue to digital converter and dispensing with the use of the converter built into the Sharp PC 1600. A very accurate A/D converter was available from the supplier of the Sharp computer. The output of the analogue to digital converter is connected to the RS 232 port of the computer. This system was favoured by the Environment Protection Authority as it was easily calibrated, more accurate than the general purpose system, and could readily be shown to confirm with Environment Protection Authority's statutory standards. Fig. 2 shows a schematic diagram of the system adopted by the E.P.A.

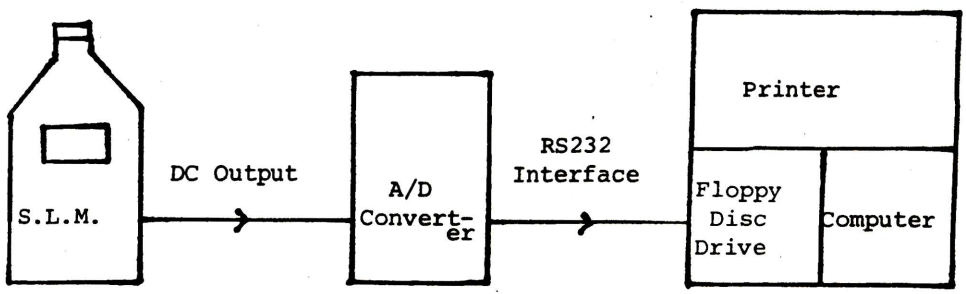


Fig. 2 Schematic diagram of precision noise monitoring system

The system uses the following components:

(a) Sound Level Meter

(b) Datacq 1616 A/D Converter

(c) Sharp PC 1600 portable computer

(d) Sharp CE 1600F floppy disc drive

(e) Sharp CE 1600P printer

The floppy disc drive is primarily used to store the noise monitoring program but may be used to store data collected by the system.

4. DISCUSSION

The noise monitoring system was developed by the EPA to measure background levels over long periods of time. But it may be used to monitor a variety of noise sources using a variety of noise indices.

The main advantages of the system are as follows:

(a) Small, light and compact.

(b) Low power consumption approximately 160 ma at 12V.

(c) Relatively inexpensive compared with other systems.

(d) Compatible with IBM computers.

(e) Individual components of the system may be used for other purposes when the monitoring system is not being used.

The main advantages of the [illegible]

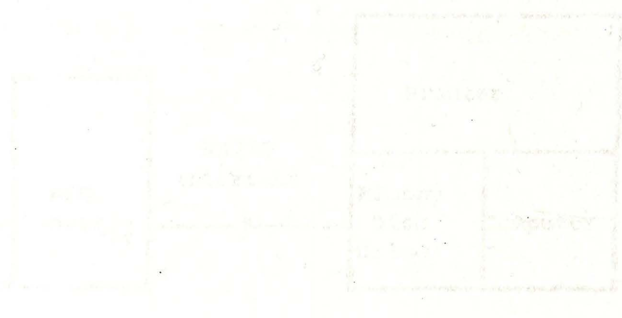
(a) Small, light and compact.

(b) Low power consumption [illegible]

(c) Relatively inexpensive [illegible]

(d) Compatibility with [illegible]

(e) Individual components of the [illegible]



Acoustics in the Eighties

Session 4C: Transportation Noise

SOME ASPECTS OF ROAD TRAFFIC NOISE AT INTERSECTIONS

S.E. Samuels

SOME ASPECTS OF ROAD TRAFFIC NOISE AT INTERSECTIONS

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ABSTRACT

Reliable techniques are available in Australia at present for the prediction, measurement, assessment and control of road traffic noise. However, these are all based on the assumption of freely flowing traffic conditions. In many cases when the traffic flow is interrupted, for example in the vacinity of intersections, this assumption is invalid and the established techniques are generally unsuitable. No alternative techniques to handle such cases are available. Recent research at ARRB has therefore been concentrating on traffic noise under interrupted flow conditions, with particular emphasis on signalised intersections. A descriptive and predictive model, which is both theoretically and empirically based, is being developed and evaluated. A brief outline of the model is given in the paper, along with some details of the inherent differences in traffic noise under interrupted and freely flowing conditions. The development and evaluation program is also mentioned and some potential applications of the model considered.

1. INTRODUCTION

Road traffic noise may be conveniently defined as the noise produced by traffic operating on the road system. In urban areas, traffic noise represents the major and most pervasive source of community noise (Brown 1980, Delaney et al. 1976, Kugler et al. 1976). Unlike many sources of community annoyance, traffic noise may be quantified. Consequently, both the measurement and prediction of it frequently represent important components in the preparation of an Environmental Impact Statement or in a planning study (CRBV 1980, Jameson 1979). Furthermore, the control of traffic noise, either from an existing or a new road facility, tends to involve rather limited options. It is generally recognised that the best long-term solution lies in what is termed ‘control at source’. However, this solution is only possible once the source behaviour is understood adequately.

Reliable techniques are available in Australia at present for the prediction, measurement, assessment and control of traffic noise (Samuels 1986, Samuels and Fawcett 1985, Stone and Saunders 1982). However, these are all based on the assumption of freely flowing traffic conditions. In the many cases when traffic flow is interrupted, for example in the vacinity of intersections, this assumption is invalid and the established techniques are generally unsuitable. Alternative techniques to handle cases involving interrupted flow are not available (Samuels 1982). Consequently, recent ARRB research has been concentrating on such cases, with particular emphasis on signalised intersections.

It is the intent of the present paper initially to explore some of the fundamental differences between the noise produced by traffic operating under both free and interrupted flow conditions. This will background the current approach to developing an interrupted flow noise model. The theoretical and empirical bases of the model will be considered along with some future developments. This paper may be regarded as documenting some ‘research in progress’.

2. FREELY FLOWING TRAFFIC NOISE

It is appropriate to commence by briefly considering some of the relevant aspects of freely flowing traffic noise.

2.1 DESCRIPTION

Noise produced by freely flowing road traffic may be regarded essentially as the complex aggregation of the noise generated by individual vehicles in the traffic stream. The magnitude, or level, of the traffic noise at a particular location varies with time in both a microscopic and a macroscopic fashion. Microscopically, the noise fluctuates with the passage of an isolated vehicle or platoon of vehicles. Macroscopically, it varies during the course of a day, primarily in response to changes in both traffic volume and composition (Saunders, Samuels et al. 1983). Once generated the noise propagates to a nearby receiver or listener. In doing so it is attenuated and the degree of attenuation depends on both the topography and the infrastructure of the site where the traffic and listener are located and whether barriers are present (UK DoE 1975, Kugler et al. 1976).

2.2 PREDICTION

Predictions of freely flowing traffic noise in Australia have to date involved the application of techniques derived overseas. The methods most frequently adopted are one developed in the U.K. (UK DoE 1975) and another produced in the U.S.A. (Kugler et al 1976), although use of the former is more widespread than that of the latter. On the basis of extensive, statistically based evaluations (Samuels and Saunders 1982, Samuels and Fawcett 1984), both methods were found to be robust and reliable and their accuracies under Australian conditions have been quantified.

3. INTERRUPTED FLOW TRAFFIC NOISE

3.1 DESCRIPTION

The noise generated by traffic under interrupted flow conditions may also be regarded as an aggregation of individual vehicle noise outputs. Vehicle operating conditions here are predominately those of acceleration and braking, rather than the constant speed situation of free flow. The resulting microscopic variations in noise level might therefore be expected to vary somewhat from those of the free flow case. Interrupted flow conditions occur at intersections and other road facilities where acceleration and braking manoeuvres are common. This induces an additional degree of periodicity into the microscopic variations since the vehicle manoeuvres involve a time during which the vehicles are stationary.

3.2 PREDICTION

Generally it is not possible to apply free flow traffic noise prediction models to interrupted flow conditions. Samuels (1982) demonstrates this clearly for both the UK and US methods (UK DoE 1975, Kugler et al. 1976). It was shown that neither method could adequately handle the complexities of the interrupted flow condition. This was attributed firstly to the invalidity of each method's assumptions when applied to interrupted flow conditions and secondly to variations in vehicle noise output which were not allowed for during development of the models.

A reliable method for the prediction of interrupted flow noise is not yet available. While some studies of the problem have recently been reported (e.g. Radwan and Oldham 1987) these have largely been concerned with propagation effects in streets containing multi-storey buildings. Also, they have been falsely based on constant speed vehicle noise data. The need for an interrupted flow traffic noise model remains.

4. AN INTERRUPTED FLOW NOISE MODEL

4.1 MODEL FRAMEWORK

A new model is targeted initially on prediction of traffic noise levels at various locations around a typical signalised intersection. Based on aggregating the noise from individual vehicles passing through the intersection, it commences with Vehicle Behaviour descriptors, which map the progress of each vehicle through the intersection. From there, Source-Receiver Geometry functions are created to determine the instantaneous distances between each observation location and every vehicle as the vehicles proceed through the intersection. At the same time, Noise Source Relationships are formulated to quantify the noise output of each vehicle at all times during the intersection traverse.

The three primary model components of Source-Receiver Geometry, Vehicle Behaviour and Noise Source Relationships may be combined mathematically to calculate, at each observer location, the noise produced by a platoon of vehicles passing through one arm of the intersection. Superposition techniques are then applied to total the noise due to both arms. By repeating these calculations over time, the noise history around the intersection may be obtained. Histories may be provided in both the frequency and the time domain, the latter allowing subsequent calculation of the conventional traffic noise indices such as L10(1h) and Leq.

4.2 THEORETICAL AND EMPIRICAL BACKGROUND

Having established an array of observer locations around the intersection through which a platoon of vehicles is proceeding, at any instant the distance between each vehicle and all observers must be calculated. These trigonometric calculations require firstly determining the location of each vehicle at every instant. Vehicle trajectories through the intersection are therefore required and one such is given in Fig. 1. This composite curve maps the approach to (constant speed followed by a deceleration and a stationary period) and the departure from (acceleration followed by constant speed) a typical signalised intersection. For given cycle phases, traffic speeds and platoon dispersions, families of these curves, such as the example of Fig. 2, can be established for each platoon proceeding through both arms of the intersection. Empirically based relationships, derived from extensive observations of Australian vehicle operating characteristics (Samuels and Jarvis 1978, Jarvis 1982), provide the deceleration and acceleration components. The ensuing source - receiver distance equations are coupled with established propagation theory (e.g. UK DoE 1975, Wang 1975) to quantify the sound pressure level variations at each observer location for every vehicle passing through the intersection.

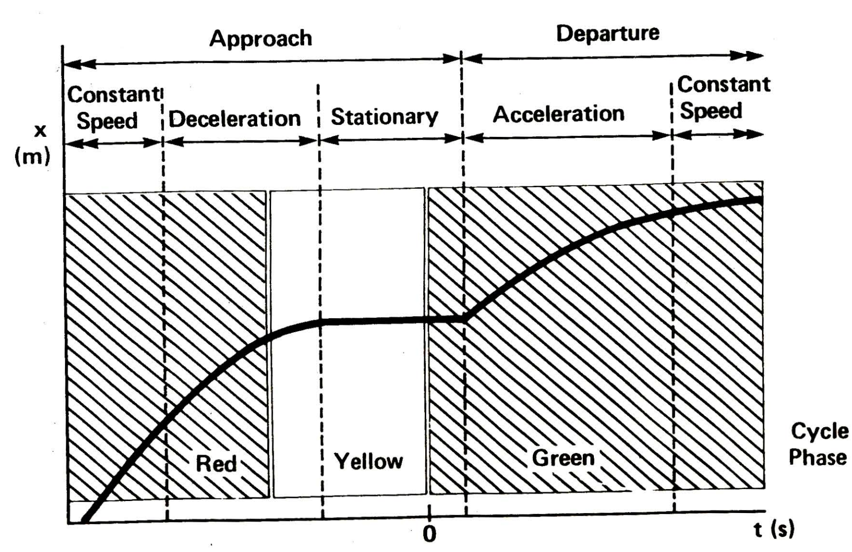

Fig. 1 - Vehicle trajectory through intersection

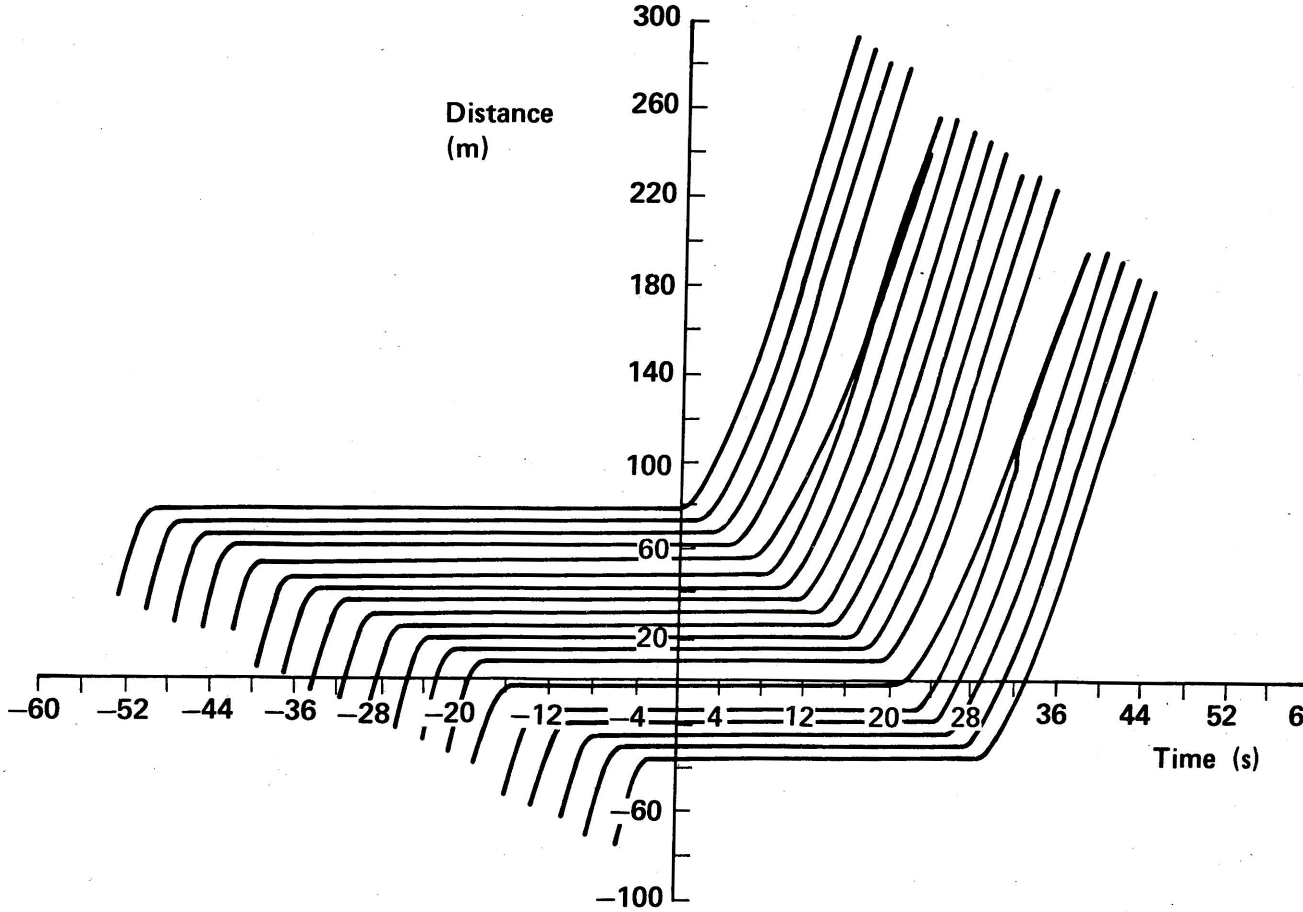

Fig. 2 - Trajectories for a platoon of vehicles

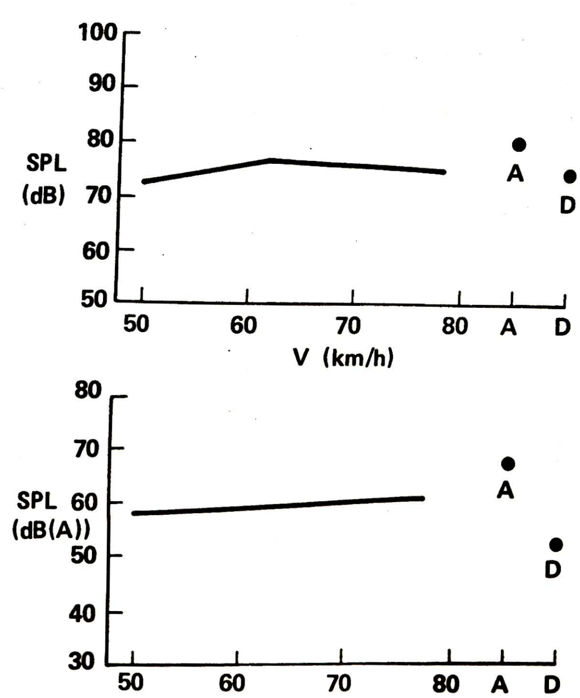

Fig. 3 - Passenger car noise source levels

Vehicle noise source data were obtained empirically (Samuels 1985) using well established passby measurement techniques on an isolated test track. During these measurements vehicles were operated under conditions of constant speed, acceleration and braking specified by the model. Once corrected (Wang 1975) for propagation distance, the noise data so obtained provided the vehicle noise source levels, at a reference distance of 15 m, for each operating condition. Typical of the noise source data are those for a passenger car which are graphed in Fig. 3.

4.3 THE NOISE HISTORIES OF A VEHICLE PLATOON

Various predictions are provided by the model and include the conventional traffic noise indices such as L10(1h) and Leq. For the purposes of the current paper, however, sample predictions are presented to illustrate various aspects of the noise generation behaviour of a platoon of vehicles proceeding through a signalised intersection. Specifically, the platoon is that whose vehicle trajectories appear in Fig. 2. For a straightforward signalised X intersection of 20 m wide pavements, the noise levels were determined at a location 10 m equidistant from each roadway during one complete cycle of the traffic signals. The resulting family of noise histories is graphed in Fig. 4, but for clarity histories for only four of the 19 vehicles in the queue are depicted. As shown, the vehicles are two cars, a medium and a large truck. These curves reflect variability in both the vehicle noise source outputs and the trajectory related, propagation distance attenuation rates associated with each vehicle.

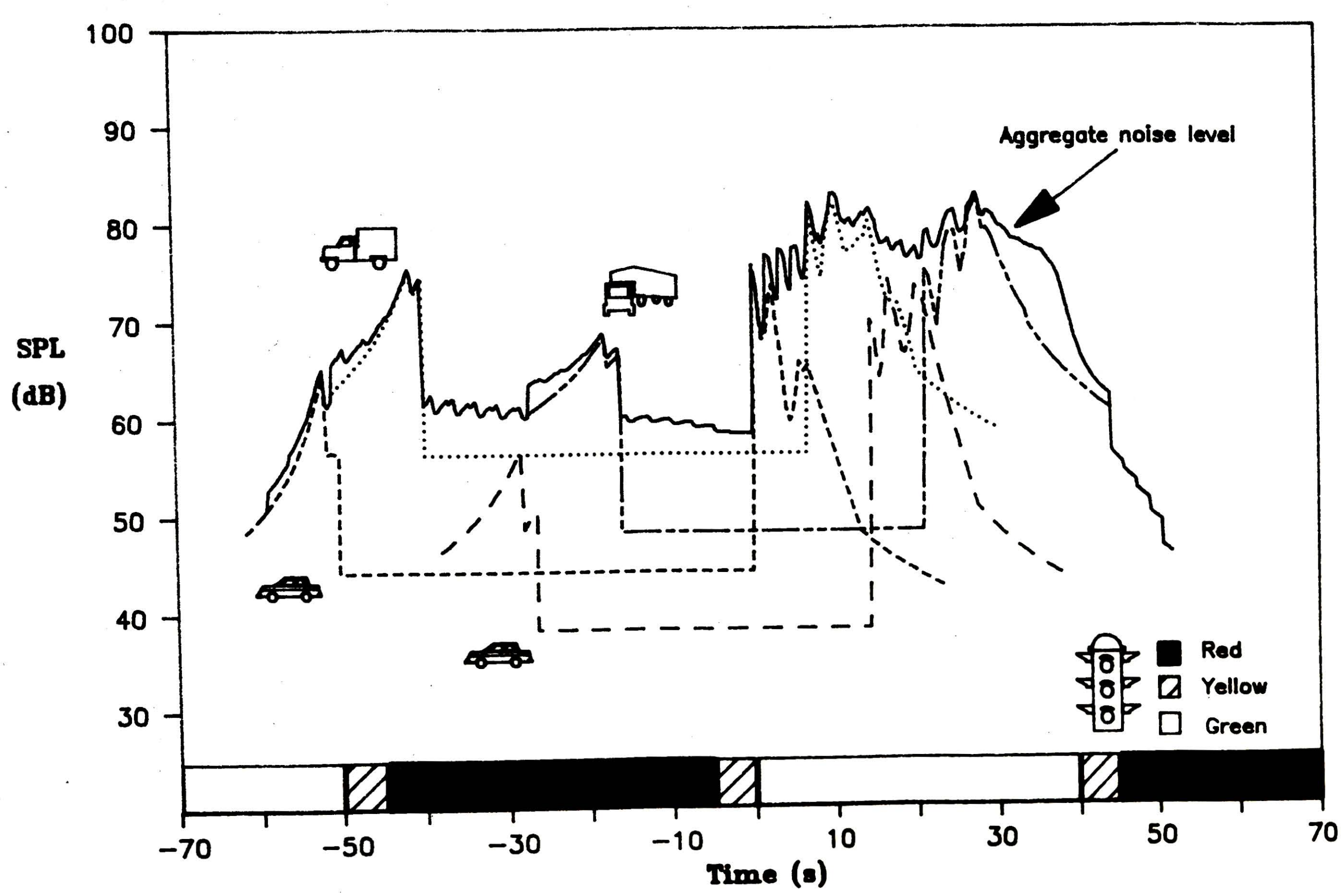

Fig. 4 - Roadside noise levels of platoon passing through intersection

The two car curves (Vehicles 1 and 10 in the 19 vehicle queue) of Fig. 4 best exemplify the propagation factors. The latter car is poisitioned midway in the queue, hence is displaced to the right of the first car in time. The latter car is initially located further from the Observer than the first car and thus has lower approach and stationary levels than the first car. On departure, the noise histories of both vehicles exhibit a three peaked curve, the relative magnitudes of which are determined by the instantaneous attenuation rates associated with each vehicle. Both the approach and departure observer levels of the higher noise producing trucks exceed those of the cars as might be expected. However, an interesting observation is the smaller 10 dB variability in the departure levels compared to some 20 dB for the approach levels. Again this is a function of source output and instantaneous attenuation rates. While the departure levels generally exceed those of the approach levels, the curves suggest that the attenuation process moderates these differences, particularly when an approaching heavy vehicle is compared to a departing car. Given this observation, it would appear that both traffic flow and composition are important in the generation of interrupted flow noise as they are in the free flow situation (UK DoE 1975).

Also shown in Fig. 4 is a plot of the 19 vehicle platoon aggregate noise level at the particular observer location. This curve was obtained by summing the 19 individual vehicle noise curves such as the four of Fig. 4. Clearly the approach levels are dominated by the heavy vehicles, which also make a major contribution to the departure levels. Overall, the departure levels exhibit fluctuations determined primarily by the acceleration noise profiles of the individual vehicles in the platoon. This aggregate curve spans a period of approximately two minutes, during which there are considerable changes in both the level and temporal fluctuations of the noise history of the platoon of vehicles.

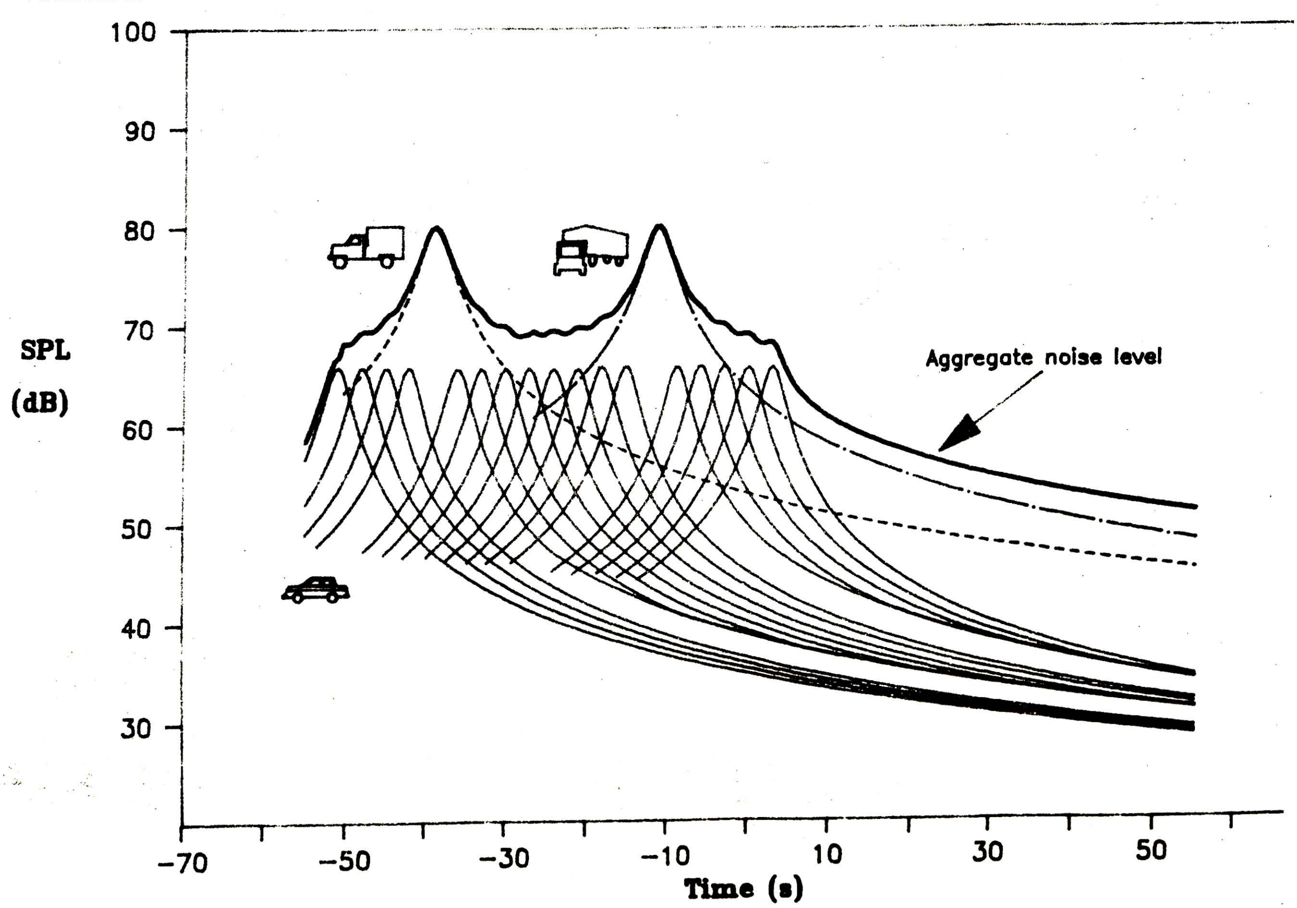

Fig. 5 - Roadside noise levels for freely flowing platoon

It is of interest to compare the noise histories of Fig. 4 with those resulting from a free flow, constant speed passby of the same platoon. The constant speed histories, obtained by suitably relaxing the vehicle operating condition algorithms of the model, are shown in Fig. 5. This series of uniform passby curves is also dominated by the higher levels of the two heavy vehicles in the platoon. On comparing the free and interrupted flow aggregate noise histories in Fig. 6, some differences are apparent. Both the level and temporal fluctuations differ for the two cases. While the free flow case exhibits a steady build up a subsequent decline in noise level, almost the reverse is true for the interrupted flow situation. Furthermore, the peak levels of the two curves are comparable, but it is apparent that the interrupted flow curve includes, during the departure phase, a substantial period of sustained, high level noise. The importance of the Fig. 6 observations is their demonstration of real and quantifiable differences between the noise accompanying the free and interrupted flow passby of a vehicle platoon. It is reasonable to hypothesise, therefore, that these two noise histories would be subjectively assessed as indeed different. These initial results also indicate the potential of the model to describe and quantify interrupted flow noise.

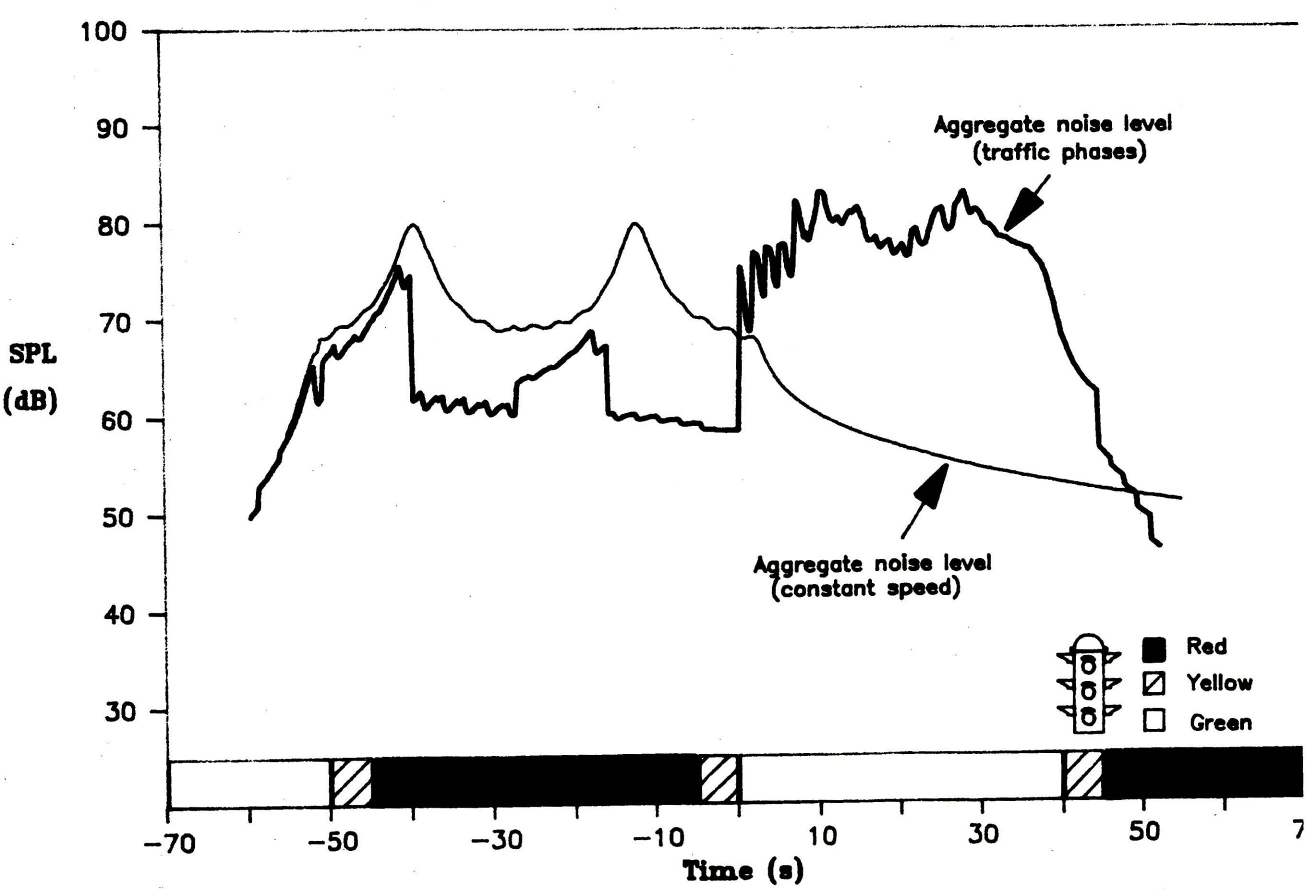

Fig. 6 – Comparison of roadside noise levels for interrupted and free flowing conditions

4.4 FUTURE DIRECTIONS

At the time of writing this paper, steps were in hand to aggregate the noise histories associated with series of platoons proceeding in two directions in both arms of the intersection. Subsequently, measurements at real intersections will be undertaken to calibrate and validate the model. It is planned that the model will be applicable over a suitable range of intersections and traffic conditions. From there it is proposed to incorporate the model into both signal design and traffic engineering practice and to produce a commercially available package for the prediction of traffic noise under interrupted flow conditions.

5. CONCLUSIONS

Road traffic noise has been considered under interrupted flow conditions such as those of a typical signalised intersection. It has been indicated that the reliable techniques for the prediction of freely flowing traffic noise are not applicable to interrupted flow situations. The development of a model to handle interrupted flow noise has been outlined. Interim results from the model, which is both empirically and theoretically based, have illustrated some aspects of traffic noise at intersections. Real and quantifiable differences between the roadside noise produced by a platoon of vehicles under free and interrupted flow conditions were demonstrated. After further development and validation the model will provide a useful means of dealing with traffic noise at intersections.

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PREDICTING SWEDISH TRAFFIC NOISE USING A MODIFIED AUSTRALIAN MODEL

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PREDICTING SWEDISH TRAFFIC NOISE USING A MODIFIED AUSTRALIAN MODEL

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ABSTRACT

In order to estimate on regional or national scales historical changes in road traffic noise exposure, as well as future predicted exposures, it is wise to use a computer model. This paper describes the adaptation to Swedish conditions, and improvement, of a model originally developed to predict traffic noise in Sydney over the years 1965 to 2015. In most computer models there is a conflict between striving for rigorous accuracy on the one hand and keeping the data input requirements and model's complexity down to manageable proportions on the other hand; this model strikes a good balance and the trade-offs in its design and adaptation are discussed. Results are presented of the application of this model to three areas in Sweden representing fairly well the cases of rural, small town and city areas. It seems possible to restore the noise environment by the year 2000 to that of the early 1960s but only if a number of measures are used together such as EEC vehicle noise standards in 1990, bypass roads and low noise road surfaces. The model was validated in that its results gave fairly good agreement with manual calculations and previously published work.

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1. INTRODUCTION

Traffic noise is one of the major environmental problems of developed countries (OECD 1985) and desire for its control has been growing, as has the problem, over many years during which society has come to rely on a great deal of mobility. The form of this mobility (bicycle, car, bus, railway etc) and its spatial/temporal distribution (town planning issues etc) determine the environmental effects, such as air and noise pollution. In order to quantify these effects it is often desirable to have computer models, which can not only predict scenarios into the future (how bad will it be? what if we did this or that?) but also estimate backwards in time (how bad was it? what were the major factors involved in bringing the problem up to its current dimensions?).

This paper reports the application to Sweden of a traffic noise model developed in Sydney (Stewart, Reardon-Small and Scott 1986). It covers results, the trade-off between simplicity and accuracy in these sorts of models and the factors in improvement of the model and adaptation to Swedish conditions.

2. CONSIDERATIONS IN CHOICE AND DESIGN OF MODEL

2.1 FREQUENCY OF USE AND SIZE OF MODEL

For a single estimation of noise impact in an area a manual calculation using nomograms is to be preferred. For many estimations a computer model is best used. The Batelle Institute (1985) developed an extremely sophisticated model for the West German Environment Department, while Wyle Research (1977) developed a model for the US Environmental Protection Agency. In Sydney a massive computer model was developed (SPA 1974) for estimation of many transport parameters (trip times, accident rates, traffic air and noise pollution etc). Because the size and complexity of a model and the extent of its data input requirements affect the cost to develop and operate it, it is unlikely that the SPA model or the West German model could be readily used again. The model of Stewart et al (1986) was developed to be reasonably sophisticated but not too large.

2.2 ACCURACY

A high degree of accuracy is desirable, but the realities of the limitations in input data often constrain precision in the output results. The accuracy of a model may be tested in absolute terms by 'validation' (i.e comparison with results from an independent source - see section 4.2) but, even if a model cannot deliver absolute accuracy, it may still be useful in relative terms (e.g. the fact that answers are, say, always 10 to 20% too low compared to the 'true' situation, if we know it, does not prevent a meaningful comparison being made between two scenarios with results differing by 60 to 70%).

2.3 AVAILABILITY OF INPUT DATA

It is rare to have sufficient data to enable a fully accurate estimation of traffic noise even in the 'one-off' situation calculated by hand, so computer modelling over a region, of necessity, requires assumptions and simplifications to be made. This is particularly the case with data on traffic counts usually expressed as annual average daily traffic or 'AADT'. For the year of interest perhaps the AADTs were not even measured for half the streets in the study area, so judgements have to be made in terms of what traffic flows had been measured previously etc. The form and availability of AADT data usually dictate the choice of road classification system - see section 3.1.

2.4 UNCERTAINTY OF INPUT DATA

Sampling errors on input parameters such as AADT, vehicle kilometres travelled (VKT) and percent heavy vehicles need to be considered, especially when one is projecting growth rates into the future.

For the computer model developed by Stewart et al (1986) the AADTs are "lumped" into frequency distributions and the shapes of these distribution curves are observed historically. It is thus possible to not only reduce the uncertainty of existing AADT data but also to make better estimates of AADT on streets where they are unavailable (see section 2.3 above) and to predict AADTs into the future. This is a powerful and unique feature of the model, especially as these AADT frequency distributions are 'kept on track' by being made to agree with data on VKT, these usually being more reliable than AADT data.

In order to test the likely sampling errors in AADT data the frequency distributions of AADT for Linköping for 1984 were derived in two ways a) by "lumping" data together, b) by detailed analysis of each segment of road assessing historical trends and avoiding double-counting of AADT measurements.
The frequency distribution curves fitted to the lumped and detailed data were found to be almost identical; errors due to lumping seem to be very small.

2.5 ADVANTAGES OF VARIOUS MODELS AND CHOICE OF MODEL

The above discussion shows the inherent conflict between accuracy of a model on the one hand and the availability of data and ease of use of the model on the other hand. Such factors are summarised in Table I.

TABLE I
PROPERTIES OF VARIOUS METHODS AND MODELS FOR
CALCULATION OF TRAFFIC NOISE IMPACT IN REGIONS

TYPE OF CALCULATION	VERY COMPLEX	COMPLEX BUT NOT TOO MUCH	BY HAND	SIMPLE
EXAMPLE:	Batelle, (1985)	The Current SPCC Model	Manual Calculation	Stewart and Rogers(1984)
ACCURACY	Very Good	Adequate	Very Good	Rough
COST:-	Very High	Moderate	High	Low
- TO DEVELOP	- A$1.5 m	- 1 Man Year	- Done	- 1 Month
- TO OPERATE	- High	- A Few Weeks	- Days	- Days
DATA INPUT REQUIREMENTS	Immense	Manageable	High	Easy
ABLE TO DO:-				
- SCENARIOS	X	#	X	#
- LOCAL AREAS	#	#	#	X
- REGIONS	*	#	X	*

X = difficult * = good # = ideal

In consideration of many factors such as these, the Swedish Road and Traffic Research Institute sought the NSW State Pollution Control Commission's model for use in Sweden, but the model required adaptation as described below.

3. ADAPTATION OF MODEL

3.1 CHOICE OF ROAD CLASSIFICATION SYSTEM

The computer model is flexible and can handle any number, n, of road types. There are two opposite ends of a 'spectrum' of characterising a road system:

a) n is very large, where each small segment of road is a different road type with a single AADT value assigned to it. For Sydney if this approach were to be used n would be about 10^4 to 10^5 and thus it can be understood why the SPA (1974) model was so cumbersome that it has not been used since.

b) n is small and AADTs are grouped into a single frequency distribution. This approach was successfully used in Stewart et al (1986) for Sydney where n was 4, and for Adelaide (unpublished) where n was 2 (due to data limitations).

In Sweden the AADT data were so comprehensive that it was possible to choose a road classification system towards the middle of the 'spectrum', with n=10, as listed in Table II. It was found that the AADT frequency distributions for two-lane roads followed the same exponentially decaying shape of curve used in Stewart et al (1986). However the distributions for four-lane roads were found to fit to a statistically 'normal' distribution. In some cases, such as freeway links with no entry or exit of traffic, the frequency distribution had, of course, a standard deviation of zero.

3.2 ALGORITHM FOR TRAFFIC NOISE CALCULATION

The Nordic Traffic Noise Model (Statens Planverk 1980), yielding results in 24 hour L_{eq}, was written into the computer program instead of the 'Welsh Office' method (UK DoE, 1975), which gives results as L_{10} (18 hour).

3.3 ADJUSTMENT OF PERCENT HEAVY VEHICLES

There is usually less information on heavy vehicle counts than on overall AADTs, so sometimes arbitrary judgements have to be made about the percent heavy vehicles, whether 5, 10, 15 or 20%. When data are available on truck and bus VKTs then heavy vehicle percentages can be adjusted accordingly.

3.4 HARD AND SOFT GROUND CORRECTIONS

In order to cope with the range of ground types that occur, two further factors were introduced: firstly the percentage of distance that is fully hard (this figure is usually small, 0 to 5%, but allows for the not infrequent occurrence of a paved car park between a block of home units and a road) and secondly the proportion of soft ground correction to be applied (less than 1.00 to allow for the fact that soft ground does not always comprise the entire distance from noise source to receiver; a figure of .85 to .95 was usually used).

3.5 DISTRIBUTION OF DISTANCES BETWEEN DWELLINGS AND ROADS

Because of the importance of attenuation of noise with distance over snow and due to non-uniform geometrical location of dwellings in the mixed residential areas in Sweden it was found necessary to use a distribution of distances between dwellings and roads rather than a single average distance used in Sydney. The proportions of dwellings at various distances for Linköping, plus other road type data, are listed in Table II. Other distance distributions were used for Kisa and Östergötland. The distance distribution factors were also adjusted to take account of the change in noise levels for higher floors of dwellings close to streets in central districts.

TABLE II
TRAFFIC AND LOCAL CONDITIONS FOR EACH ROAD TYPE USED AS MODEL INPUTS FOR LINKÖPING

ROAD TYPE	SPEED LIMIT km/hr	No.OF LANES	HEAVY VEH. %	PEOPLE PER KM	DISTANCE DISTRIBUTION (metres) 10	20	30	50	70	100	150
1	Local	2	3	300	.45	.40	.10	.05	-	-	-
2	50	2	5	350	.25	.60	.10	.05	-	-	-
3	70	2	15	250	-	-	.15	.25	.20	.20	.20
4	90	2	10	13	-	-	-	-	-	-	1.0
5	110	2	10	13	-	-	-	-	-	-	1.0
6	50	4	15	350	.25	.60	.10	.05	-	-	-
7	70	4	15	250	-	.40	.25	.10	.05	.20	-
8	90	4	10	100	-	-	-	-	-	-	1.0
9	110	4	10	13	-	-	-	-	-	-	1.0
10	Any	>4	10	13	-	-	-	-	-	-	1.0

3.6 OPTIONAL SMOOTHING

Noise levels are stored after being rounded to the nearest decibel. In some situations where a significant segment of the road network has a uniform AADT which grows year by year, the number of people exposed to above 65 dBA can increase suddenly because in one year the calculated noise level is, say, 64.49 dBA while the next year it is, say, 64.51 dBA. An optional averaging procedure was introduced to smooth out what, at first sight, might have been regarded as anomalies.

3.7 TRANSFORMATION OF RESULTS

As output data one receives a detailed distribution of the numbers of people exposed to each noise level above 50 dBA (24 hour average outside the house). In order to simplify these results in a more general way these noise exposures were transformed into the number (or percent) of annoyed persons using a dose/response relationship (Kommunikationsdepartmentet 1980) which may be expressed as:

Percent annoyed persons= $3(L_{eq} - 46)$ for $55 < L_{eq} < 80$

4. RESULTS

4.1 NUMBERS OF PEOPLE ANNOYED

Detailed results, an explanation of the Scenarios and discussion of future policy possibilities are available in a full report in Swedish (Kommunikationsdepartementet 1987).

In the figures below it is seen how the percentages of persons annoyed by road traffic noise in three different types of area in Sweden have changed from 1965 to the present and are projected to change from now until 2015. The studied areas are a rural area (the province of Östergötland of about 400,000 people), a small town (Kisa of 4,000 inhabitants with a major highway through it) and a provincial capital city (Linköping of 100,000 inhabitants of which 20,000 are in surrounding districts).

Fig. 1. Rural Area (Östergötland)

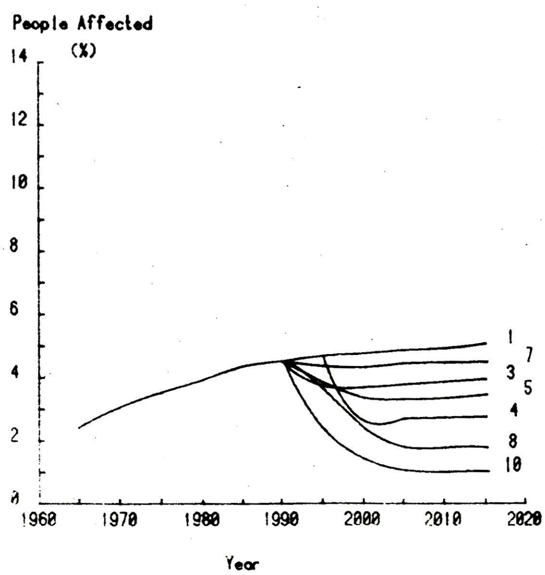


Fig. 2. Small Town (Kisa)

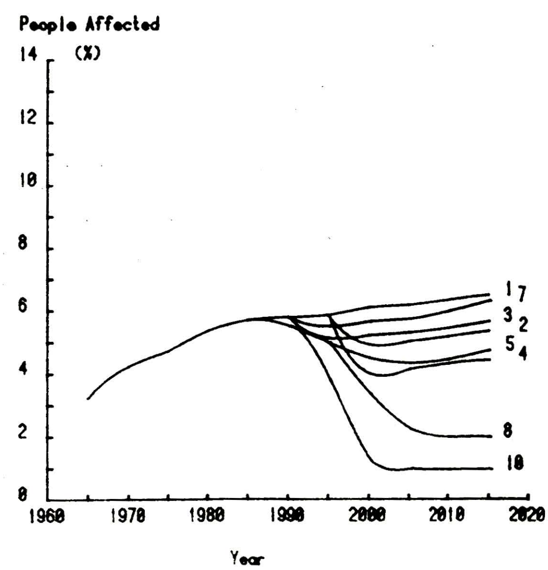
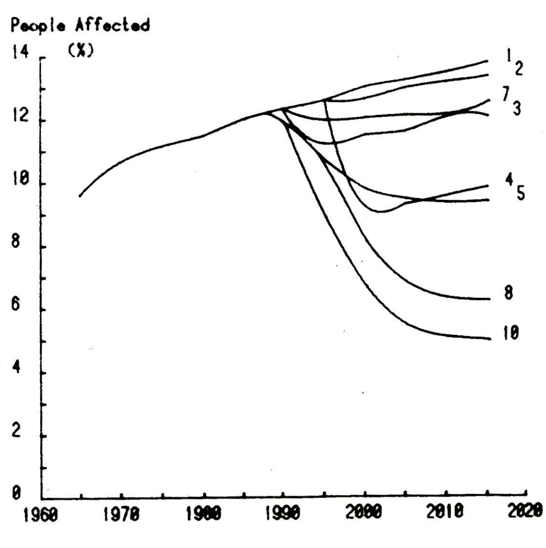


Fig. 3. City (Linköping)



INDEX OF SCENARIOS:

1. no further controls
2. new bypass road
3. low noise road surfaces
4. super low noise road surfaces
5. lower vehicle noise standards in 1990 (EEC/84/424) *
7. noise effects of proposed NO_x control measures
8. vehicle noise goals: 75 dBA for cars, 80 dBA for trucks *
10. scenarios 2,3,7 & 8 combined

* Similar reductions of tyre/road noise would be required at higher speeds to get the full benefit of these standards.

4.2 VALIDATION OF RESULTS

The results of the model were validated by comparison with some manual calculations and previously published estimates, where available.

TABLE II
COMPARISONS OF PERCENTAGES OF POPULATIONS EXPOSED
TO NOISE LEVELS FOR ÖSTERGÖTLAND, KISA AND LINKÖPING

REGION	SOURCE OF NUMBERS	Noise Levels dBA, 24 hr L_{eq}			
		55-59	60-64	65-69	> 70
ÖSTER-GÖTLAND	1985 estimate from road construction authority	-	-	0.6	0.1
	Model: 1985	5.9	3.4	0.8	0.1
KISA	Manual calculation for 1985	9.2	4.6	1.0	0
	Model: 1985	9.3	3.2	1.3	0
KISA	1973 rough estimate from SOU (1974) for a small town	9	4	0	0
	Model: 1975	9.6	2.9	0.9	0
LINKÖPING	1981 estimate from Linköping Municipality	-	-	-	0.5 to 1.1
	Model: 1980	9.9	7.2	4.6	1.0
LINKÖPING	1973 rough estimate from SOU (1974) for a medium city	24	15	11	2
	Model: 1975	9.4	6.7	4.4	1.3

5. CONCLUSIONS

The model has demonstrated its utility and flexibility, being applied first to Sydney then in Sweden, after adaptation, to a city (100,000 people), a small town (4,000 people) and a rural area (400,000 people). Now that the model has been improved it could be applied to a new area in about 4 weeks.

Interpretation of the model's results should be made with an understanding of the considerations involved in the choice and design of these sorts of computer models as described in Section 2.

The computer model was validated insofar as its results gave fairly good agreement with manual calculations and previously published work.

The model estimates that, if no measures for noise reduction are made, the number of annoyed persons (in their permanent residences) will increase by 20 to 30% from 1980 to 2015 in the studied areas, having increased strongly (30 to 60%) since the early 1960s.

It seems possible to restore the noise quality by the year 2000 to that of the early 1960s but a number of measures would be needed together, such as EEC vehicle standards in 1990, bypass roads and low noise road surfaces.

Looking beyond 2000 a scenario achieving technologically achievable vehicle noise goals of from 75 dBA for cars to 80 dBA for the heaviest vehicles plus corresponding reductions in tyre/road noise and some supplementary measures would bring the traffic noise problem well under control.

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Acoustics in the Eighties

ENVIRONMENTAL NOISE FROM AIRCRAFT OPERATIONS

L.C. Kenna

ENVIRONMENTAL NOISE FROM AIRCRAFT OPERATIONS

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ABSTRACT

The 1980s has been a decade in which notable progress has been made in reducing the environmental noise heard in the vicinity of airports from aircraft operations. The changes have been due to a combination of technical developments, revised operating procedures and regulatory requirements. The paper outlines the contribution of each of these components to the overall situation and illustrates the changes which have been brought about. This has also been a decade in which a combination of planning and assessment tools, specifically designed for Australian conditions, has become available for land use planning and building design in relation to aircraft noise. These comprised the Australian Noise Exposure Forecast system which was researched, developed and brought into use in the early 1980s, and the publication in 1985 of a revised version of AS 2021, the SAA's code for building siting and construction to avoid aircraft noise intrusion. There have been other developments which have been, or may become, a cause for concern. These have included the increasing use of helicopters in urban areas, the introduction of new types of aircraft such as ultralights and airships, and the proposed introduction of new aircraft power plants such as the unducted fan engine.

1. INTRODUCTION

Although some of the residents of communities situated beneath the flight paths surrounding the busier airports may not readily acknowledge the changes, the 1980s has been a decade of progress in reducing the environmental noise heard in the vicinity of airports from aircraft operations. The reduction of noise levels has been a consequence of a number of factors acting in combination. Such factors have included technical developments, administrative and operational procedures, and regulatory requirements imposed because of community concern.

2. CHANGES IN AIRCRAFT NOISE LEVELS

2.1 TECHNOLOGICAL CHANGES

Propulsion Systems

The noise of the engines is an inevitable by-product of the operation of aircraft. However, perhaps partly due to the relatively low number of operations and the novelty, it was not until the introduction of jet aircraft into commercial service in the early 1960s that aircraft noise became a major problem for those people living close to airports.

The first generation of jet aircraft to operate into Australia included the Boeing 707 and DC-8 operating on international routes. These aircraft were powered by turbo-jet engines, basically similar to the engines being used on military jet aircraft at the time. In a turbo-jet engine, air is drawn into the front of the engine where it is compressed. It then passes into the combustion chambers where fuel is injected and continuous combustion takes place. The heated gases emerge from the combustion chambers at very high pressure and are then discharged, via a turbine which extracts the power needed to operate the input compressor, fuel pumps and electric generators, into the jet exhaust. Because of the high gas pressure, the exhaust stream emerges into the air at very high velocity. The resultant reaction propels the aircraft forward.

Most of the noise of the engine is a consequence of the mixing of the hot, high velocity exhaust gases with the still outside air and is produced external to the engine. The higher the exhaust velocity, the greater the noise level, with the sound power increasing in proportion to the eighth power of the exhaust velocity.

Early endeavours to quieten these engines involved fitting of exhaust nozzles designed to improve mixing of the exhaust gases with the air by sub-division of the exhaust stream. This approach had limited success. Noise levels due to the mixing process were lower, but at the expense of reduced engine performance and increased fuel consumption.

A development aimed mainly at improving fuel consumption fortuitously turned out to produce significantly lower noise levels than the turbo-jet engines. This was the development of the by-pass, or turbofan, engine. Turbofan engines powered the next generation of jet aircraft, such as the Boeing 727s and DC-9s which entered service in the mid-1960s. In a turbofan engine, not all the air entering the engine passes through the compressors, combustion chambers and turbines. Some of the air is blown by the intake fan around the outside of the engine core, so that it forms a slower-moving, colder envelope of air surrounding the hot, high velocity exhaust gases. The effect is to create a broader transition zone in which mixing occurs, rather than the original sharp, shearing change and the jet exhaust noise is reduced.

The air which has not passed through the engine core is referred to as the bypass flow. The ratio of the bypass flow to the core airflow is the bypass ratio of the engine. The early turbofan engines had bypass ratios of about 1:1 and are now known as low bypass ratio engines. Spurred on by the quest for improved fuel consumption and the need to satisfy mandatory noise limits, bypass ratios have progressively increased. Bypass ratios of the current generation of engines are about 5:1, and for the next generation will be about 20:1, which is the theoretically optimum value. Above 20:1, the weight and size of the engine nacelle begins to cancel out the benefits of further increases in the bypass ratio.

When operating at take-off power, the dominant source of noise is the jet exhaust, so the bypass engine is substantially quieter for a given thrust level than the original turbo-jet engine. At the lower power levels used for approach and landing, exhaust noise is not so predominant, and fan noise becomes significant. Noise control methods have included modifications to the aerodynamics of the intakes and lining the fan ducts with acoustical absorbent materials, but the dramatic noise reductions achieved for aircraft taking off have not been equalled for aircraft landing.

The environmental effects of these changes can be seen by comparing the noise levels produced at a given point by aircraft of the different generations. Such data is available from the output of the noise monitoring system at Sydney airport. Table I shows some comparative levels. The figures are dBA maximum levels, S-response, averaged over a three month period (first quarter of 1987) at a monitoring terminal which is approximately 6,500 metres from start of roll for takeoffs, and 4,000 metres from threshold for landing. The aircraft masses shown are also approximate, since they will vary depending on the loading and destination of the aircraft. The aircraft are arranged in order of age of the type, with the oldest first.

TABLE I

AIRCRAFT NOISE LEVELS COMPARED

AIRCRAFT	MASS (tonne)	SOUND LEVEL (dBA) TAKEOFF	LANDING
B707	145	101 (est.)	100.2
B727	90	91.0	89.2
B747	370	83.9	92.6
A300	150	82.1	88.4
B767	140	78.2	85.6
B737-300	60	77.8	84.7

Airframes and Aerodynamics

Development work in airframes, such as the use of new composite materials to reduce aircraft weight while maintaining strength, and developments in aerodynamics in the form of methods of reducing drag, such as the use of winglets have been principally aimed at decreasing fuel consumption, but they also have the potential to improve noise characteristics of aircraft, particularly on takeoff. They enable higher rates of climb so that aircraft are at a higher altitude before passing over noise sensitive areas.

Electronics

Developments in electronics have also created the possibility of reduction of aircraft noise exposure around airports. The International Civil Aviation Organization (ICAO) has selected a new type of landing aid known as the Microwave Landing System (MLS) for introduction into service during the 1990s. Existing landing systems guide the aircraft along a linear, 3 degrees glide slope. The MLS system will provide continuous three-dimensional coordinate information on the aircraft's position and this data will enable an aircraft to fly curved and stepped approaches if required. It will be possible to arrange flight paths which deviate around or maintain greater height above noise sensitive locations on the ground.

2.2 ADMINISTRATIVE AND OPERATIONAL PROCEDURES

Administrative procedures reduce noise without changing the method of operation of an aircraft: the most notable example is the curfew. Operational procedures involve restrictions on the flight paths or runways that may be used by an aircraft, or specify a method of flying an aircraft. Both methods involve costs caused by delays, flying of longer flight paths or by decreasing the handling capacity of an airport.

Curfews

Curfews are one of the most well-known and controversial procedures for reducing aircraft noise exposure. Several Australian airports have curfews, which generally prohibit operations by some types of aircraft between the hours of 11pm and 6am. The original curfews placed a total ban on the operation of jet aircraft but placed no restrictions on propeller-driven aircraft. Since there are now some jet aircraft which are quieter than some propeller-driven aircraft, a new curfew policy has been proposed, based on noise certification of individual aircraft types rather than type of power plant. The new policy would provide benefits both to the operators and to the communities around the airport.

Preferred Runways and Flight Paths

Preferred runway procedures have been developed so that, for a given number of aircraft movements from an airport, noise exposure will be a minimum. First preference is given to ensuring that departing aircraft use the runway that will cause the least annoyance: for example, in Sydney the first preference is for departures over Botany Bay. It is obvious that when traffic is heavy, the preferred runway for arrivals cannot be in the reciprocal direction without risking safety or causing extended delays. The overall preference order is a compromise between environmental, economic and safety aspects.

Preferred flight paths have also been prescribed for aircraft approaching to and departing from a particular runway in order to either avoid overflying noise sensitive areas, or to fly over at such a height that the effect of the noise will be reduced. These are also a compromise, since they almost invariably involve the aircraft travelling extra distance.

Noise Abatement Operation

Noise abatement climb procedures are required at a number of airports. These procedures specify engine power and altitude requirements for aircraft departing from certain runways, with the intention of achieving as great a height as possible before passing over noise sensitive areas.

2.3 REGULATORY REQUIREMENTS

ICAO Noise Certification Requirements

ICAO was established, following the Chicago Convention on International Civil Aviation in 1944, to develop and regulate international civil aviation. This mainly involved considerations of safety and efficiency, but in the late 1960s, the problem of aircraft noise around airports had become sufficiently serious to warrant a special meeting of ICAO, which was held in 1969. As a result a set of International Standards and Recommended Practices for aircraft noise was developed, which became Annex 16 to the Chicago Convention.

Annex 16 divides aircraft into various categories, by type of engine, aircraft weight and date of manufacture. It sets upper limits to the amount of noise, measured by prescribed procedures, which is permissible from aircraft in each category. An aircraft which satisfies the limits for its category will be "noise certificated" to a particular Chapter of the Annex. Categories range from the smallest propeller-driven aircraft through to the largest jets, and include helicopters. In the period since the Annex was first produced, ICAO has refined and tightened the requirements. The enforcement of the requirements by ICAO member States has been a major factor in aircraft and engine manufacturers developing new, quieter models.

Air Navigation (Aircraft Noise) Regulations

Australia is a signatory State to the Chicago Convention, and thus has an obligation to follow the ICAO standards and practices. In 1984, the Australian Government promulgated the Air Navigation (Aircraft Noise) Regulations. Under these regulations, Australian-registered aircraft may not operate unless they meet the appropriate ICAO noise certification requirements.

Aircraft Noise Policy

The Air Navigation (Aircraft Noise) Regulations alone would not prevent the operation of noisy jet aircraft built before the preparation of ICAO Annex 16, or the operation into Australia of foreign-registered, non-noise-certificated aircraft. Using a mixture of powers available to it under various Commonwealth Acts, the Australian Government has had, since 1977, a specific policy on control of aircraft noise. The policy is as follows:-

(a) All aircraft imported into, or built in, Australia are required to meet the applicable noise certification standards of ICAO Annex 16;

(b) All Australian operators were required to ensure that their subsonic jet-propelled aircraft having a maximum certificated take-off mass of 34,000 kg or more, met the requirements of ICAO Annex 16, Volume 1, Chapter 2 or 3 by 1 January 1985.

(c) From 1 January 1988, foreign-registered subsonic jet-propelled aircraft having a maximum certificated take-off mass of 34,000 kg or more will not be permitted to operate in Australia unless they meet the requirements of ICAO Annex 16, Volume 1, Chapter 2 or Chapter 3.

3. PLANNING AND ASSESSMENT TOOLS

3.1 GENERAL

The 1980s saw the introduction in Australia of a set of planning and assessment tools which have the potential to greatly reduce the possibility of unwise land use decisions being made concerning the developments of airports and their environs. These tools comprise the ANEF system for aircraft noise assessment, and Australian Standard AS 2021-1985 (Acoustics - Aircraft noise intrusion - Building Siting and Construction).

3.2 ANEF SYSTEM

In 1970, the House of Representatives Select Committee on Aircraft Noise looked at systems for predicting noise environments near airports and, for land use planning purposes, recommended the application of the Noise Exposure Forecast (NEF) system. The US-developed NEF system was to be used with caution until its compatibility with Australian conditions was tested. This testing was undertaken at the beginning of the 1980s in a large scale study combining a social survey and a noise assessment around five major Australian airports. The research was undertaken by the National Acoustic Laboratories and the project was jointly funded by the Australian Departments of Defence and Transport (later Aviation). The report on the study was issued in 1982.

The study found that an equal-energy measure such as the NEF was the best predictor of Australian community reaction to aircraft noise, but with minor modifications to improve the correlation. The modified system was referred to as the Australian Noise Exposure Forecast System (ANEF) and was adopted by the two sponsoring Departments for all future aircraft noise assessment. Its use was endorsed in the 1985 report of the House of Representatives Select Committee on Aircraft Noise.

The two Departments produce ANEF charts for most airports throughout Australia. The charts are maps of the airport and the surrounding areas, on which noise exposure contours of 20, 25, 30, 35 and 40 ANEF units have been drawn. The charts also include a table of recommendations of land uses compatible with the noise exposures. These recommendations were refined from those in the original US system using the data obtained in the NAL survey.

3.3 AUSTRALIAN STANDARDS

In 1985, the Australian Standards Association published a revised version of AS 2021, which incorporated the use of the ANEF system. This standard provides guidance on selection of sites suitable for various building types, and details of methods of construction for new buildings exposed to various levels of aircraft noise. It can also be used to assess the acoustical adequacy of existing buildings. The standard includes the ANEF land use compatibility recommendations.

4. NEW DEVELOPMENTS

4.1 HELICOPTER USAGE

The sound of helicopters has become relatively common around Australian cities. Helicopters are in daily use by the police, the media, rescue services and various other government and private organisations. Helicopters generally are permitted to fly at a height of 1,000 feet over populous areas, although police helicopters may operate lower subject to certain conditions. In an effort to reduce the air traffic control load in Sydney, several helicopter transit lanes were approved. These are located over the Harbour and rivers, and in the lanes helicopters may operate at heights lower than 1,000 feet during daylight hours. The noise criterion used for environmental assessment of the lanes was that noise exposure at the ground should not exceed a 12-hour (7am to 7pm) L_{eq} of 55 dBA.

There is also the problem of noise from the use of helicopter landing sites. Depending on the type of operation being performed or the distance from aerodromes, such landing sites may require the approval of the Department of Aviation or they may fall into the responsibility of State or local governments. Consequently, the manner in which proposed landing sites are environmentally assessed may vary from site to site.

4.2 NEW AIRCRAFT TYPES

New aircraft types, such as ultralights and airships, have appeared in recent times, bringing new environmental concerns. For ultralights, operation is generally away from populous areas so the noise affects fewer people.

Airships have been the greater concern because they have been operated on sightseeing flights over major population centres. There is no noise certification criterion applicable to airships. The noise levels produced by the airships are not particularly high, but because of their low speed they are audible for longer periods than conventional aircraft. This was allowed for in the assessment of the airship's noise by setting a criterion level in terms of Sound Exposure Level (SEL). Noise of the airships has caused many complaints in the Sydney area, but appears to have triggered complaints for other reasons, such as invasion of privacy, visual pollution, advertising of beer and cigarettes, and disturbance of animals.

4.3 NEW POWER PLANTS

Flight tests have begun for a new type of aircraft engine which at least two manufacturers propose fitting to aircraft in the early 1990s. The new engine is the propfan, or unducted fan. It consists of a gas turbine engine core, as in a normal turbofan engine, driving two contra-rotating rows of fan blades. There is no outer duct surrounding the fan as there is in a turbofan engine. In theory, the engine is much more fuel-efficient than existing engine types.

Aircraft powered by the new engine must satisfy the ICAO Chapter 3 noise requirements, i.e. the most stringent requirements. The certification levels are measured at prescribed points during takeoff and landing. The US Federal Aviation Administration has issued for comment a notice which suggests that, while a propfan-engined aircraft may satisfy those requirements, during cruise flight it may be substantially noisier than existing aircraft. The reason is that when the rotational speed of the blades is added to the forward velocity of the aircraft, the blade tips may be travelling at supersonic speed when the aircraft is cruising. It was suggested that the noise levels of a twin-engined propfan aircraft could be up to 25 dBA higher than the noise levels of a twin-engined turbofan aircraft flying at the same altitude. Sound levels of that magnitude could mean the redesign of air routes to avoid cruising overflight of cities and towns between the origin and destination points. Engine and aircraft manufacturers have stated that wind tunnel and flight tests have shown that noise levels of propfans will be of similar magnitude to or even slightly less than those of turbofans. With such a spread of possible levels, definitive results will be awaited with great interest because of their potential effects on aircraft routes.

5. CONCLUSION

A combination of factors has lead to a reduction of exposure to aircraft noise in the vicinity of airports during the 1980s. This reduction has occurred in spite of increasing volumes of air traffic. It is forecast that further reductions will occur over a number of years before they are counter-balanced by extra traffic.

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HEDE,A.J. and BULLEN,R.B.(1982). Aircraft Noise in Australia: a Survey of Community Reaction. N.A.L. Report No. 88. AGPS.

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REPORT OF THE HOUSE OF REPRESENTATIVES SELECT COMMITTEE ON AIRCRAFT NOISE SEPTEMBER 1985. Aircraft Operations and the Australian Community. AGPS.

U.S. DEPARTMENT OF TRANSPORTATION: FEDERAL AVIATION ADMINISTRATION (1987). Noise and Emission Standards for Aircraft Powered by Advanced Turboprop (Propfan) Engines: Proposed Rule. 14 CFR Part 36. Federal Register.

INFORMATION RETRIEVAL

The abstracts and keywords are provided in the interests of improved information retrieval. They were prepared by the authors who used various Theasuri to select keywords.

BROWN, T.E. (1987) : COMMUNITY NOISE AND PLANNING. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 1-12.

KEYWORDS : Community/noise surveys/planning/noise assessment/environmental noise/legislation/noise control
ABSTRACT : The effects of noise on the community is an ever increasing intrusion on the living environment, and in addressing this problem scientific and engineering professionals must consider noise reduction strategies which will enable the increasing development of our planet's resources with a minimum of disturbance to the lifestyle of the population. The application of land use planning as a means of controlling noise must be considered along with other strategies, as more effective noise reduction can be achieved by this means than by all other strategies in relation to minimizing community annoyance.

RENEW, W.D. (1987) : COMMUNITY NOISE SURVEYS – WHAT DO THEY TELL US? Proc. 1987 Aust. Acoustical Soc. pp. 13-22.

KEYWORDS : Australia/noise surveys/community/cumulative distribution/noise spectra/survey method/regression analysis/mean levels/variation
ABSTRACT : This paper describes the major community noise surveys carried out in Australia since 1974. Comparisons are made of survey techniques and of the values of the major noise descriptors in noise area categories specified in AS 1055-1984. Descriptions are given of the more usual methods of data presentation including noise probability distributions and noise spectra. Some discussion is given of accuracy and reliability of measurement.

EISNER, M., TAYLOR, E and BURGESS, M. (1987) : BACKGROUND NOISE LEVELS FOR COMMUNITY NOISE ASESSMENT. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 23-30.

KEYWORDS : Noise/background noise/noise assessment/environmental noise*
ABSTRACT : The method used for the assessment of offensive noise in most parts of Australia is based on a comparison between the background noise levels, ie when the potentially offending noise is not present, and the levels with the noise present. When measurements are possible, the background levels are considered as the L_{A90} and the noise levels as the L_{A10}. This paper discusses situations where the use of L_{A90} does not give a true indication of the type of background noise in the area. Consideration of the noise climate based on the L_{A10} and the L_{A90} for both the noise present and noise absent situations is discussed as a fairer method for noise assessment.

MURRAY, B.J. (1987) : COMMUNITY REACTION TO OVERPRESSURE FROM BLASTING.. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 31-37.

KEYWORDS : Blasting/overpressure/community reaction/saound/vibration/quarry

ABSTRACT : Overpressure from blasting has been identified for some time as causing damage to buildings and annoyance to building occupants. A survey has been carried out around a quarry in Sydney which is surrounded by residential areas to quantify the degree of annoyance caused by overpressure. Overpressure measurements were made during a series of blasts and the residents were surveyed to determine their response to the blasts. A correlation between overpressure level and the percentage of respondents highly annoyed was obtained and even at a level of 115 dBL a significant percentage of respondents was highly annoyed.

LAWRENCE, A.B. (1987) : ACOUSTICS IN HARD TIMES. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 39-45.

KEYWORDS : Acoustics/economics/legislation

ABSTRACT : This paper discusses the potential danger of the erosion of environmental and workplace noise controls in times of economic contraction, owing in part to the emphasis on commercially-oriented research and development.

DAVY, J.L. (1987) : THE ABSORPTION OF SOUND BY AIR. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 47-54.

KEYWORDS : Acoustics/sound absorption/sound

ABSTRACT : The mechanisms of the absorption of sound by air are described. These mechanisms are the quantum mechanical effects of vibrational and rotational relaxation and the classical mechanical effects of viscosity and heat conduction. The history of the development of the absorption formulae and the work of Kneser, Evans and Bazley, Harris, Monk, Delany, SAE, ANSI and ISO are reviewed. The different air absorption formulae are used to remove the air absorption from the total absorption, of a bare 600 m^3 reverberation room, which has been determined by reverberation time measurements at 35 different values of temperature and humidity. This enables the calculation of the wall absorption coefficient of the reverberation room at each value of temperature and humidity. The air absorption formulae are judged by their ability to produce the smallest standard deviation of the wall absorption coefficient across the 35 different values of temperature and humidity. This is done on the assumption that almost all the variation in the total absorption is due to the air absorption. The 1986 draft ISO air absorption formula gives the lowest standard deviation compared to the other formulae at all but three of the measurement frequencies. At these three frequencies it just fails to give the lowest standard deviation.

FRICKE, F and TREAGUS, R. (1987) : SOUND ATTENUATION RATES NEAR THE GROUND. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 55-65.

KEYWORDS : Sound propagation/outdoor propagation/noise measurement/sound absorption/sound attenuation/variability/theoretical predictions/atmospheric conditions/ scattering
ABSTRACT : One of the greatest sources in variation in outdoor sound emission levels is the atmosphere. For nominally identical atmospheric conditions attenuation rates may vary dramatically, especially near sunrise and sunset. This variability should be considered indicative of the errors inherent in existing prediction techniques. In this paper results of field measurements are presented which indicate the attenuation rates and their variability are a function of time of day, relative humidity and cloud cover. The paper also compares measured and predicted sound levels in a limited number of cases. These comparisons suggest that European-based prediction methods may not have sufficient atmospheric stability categores for Australian conditions.

PEPLOE, P.E. (1987) : SOUND PROPAGATION THROUGH THE ATMOSPHERE – A DETAILED FIELD STUDY. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 67-77.

KEYWORDS : Acoustics/sound/sound propagation/field study/environmental noise/ outdoor propagation/atmospheric propagation/prediction/high-power sound source/noise measurement/meteorology
ABSTRACT : The paper describes a detailed field study of sound propagation through the atmosphere. Using a purpose-built band-limited high intensity sound source, sound levels were measured at distances up to 4 km from the source using automatic loggers and tape-recordings were also made. Sound level data were gathered for several days at different times of the year at each of 4 different locations. These locations were selected to provide differing terrain and climatic conditions. Meteorological data with altitude were also gathered and ground meteorology along the propagation was monitored. Some typical results and findings from this study are illustrated at the end of the paper.

MADRY, A.J. (1987) : PREDICTION METHODS FOR OUTDOOR SOUND PROPAGATION. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 79-87.

KEYWORDS : Outdoor sound propagation/prediction methods/ray tracing/refractive index field/meteorological profile/diffraction/atmospheric turbulence
ABSTRACT : This paper discusses the various methods of predicting sound levels out-doors and areas where inaccuracies are likely to occur. A theoretical model is presented based on a digital ray tracing method and the effects of diffraction are included by using diffracting rays. The model can simulate the effect of fluctuating atmospheric conditions and predict the range of sound levels likely to occur.

GRIFFITHS, P.J. (1987) : EXHIBITION ACOUSTICS : THE MUSEUM AS THEATRE. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 89-98.

KEYWORDS : Acoustics/museums/exhibition/theatre/Australian Pavilion Expo 85/ Tsukuba/Japan/noise control/reverberation control

ABSTRACT : In their contemporary context, museums are required to be more than a collection of objects. They are now conceived as complex environments providing many multi stimulus experiences for visitors. A museum and its exhibition experience can be considered as being similar to a theatre and an associated theatrical experience. Just as all theatrical performance contains an acoustical component, acoustics can also play a major role in the success or failure of an exhibition experience. This analogy between theatre and exhibition has been used as the conceptual basis for the acoustic design of the highly acclaimed Australian Pavilion for Expo 85, designed by the Department of Housing & Construction. Further refinement of this acoustical approach is being used for a number of museums and exhibition centres currently being designed.

PALAVIDIS, M. and FRICKE, F. (1987) : NON-AUDITORY INFLUENCES ON THE PERCEPTION OF SOUND IN CONCERT HALLS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 99-109.

KEYWORDS : Concert hall/non-auditory perception/polysensory perception/acoustics/ questionnaire/phsychological experience

ABSTRACT : The perception of sound in a concert hall is a very individual experience. Consensus may be had on a number of aspects of a performance and a performance space but the overall impressions of the audience will not generally be polarized. How do we explain this divergence of the listener's auditory perception? Given that the acoustic design of the concert hall is satisfactory, it is logical to conclude that the individual's perception of sound will depend upon factors other than auditory ones. One must consider the influence of polysensory impression and interaction; in particular it is important to consider that response to the sensory environment inside a concert hall will be the product of a limitless range of physiological and psychological experiences. A person's past experience of music and concert halls may also be important in the determination of the quality of sound. In an attempt to evaluate the influence of the phenomena on the perception of sound a questionnaire was designed and distributed to audiences attending musical performances. The results of the questionnaire highlighted a number of interesting relations on the polysensory perception of sound.

THOMPSON, W.F. (1987) : PSYCHOACOUSTIC FACTORS IN MUSICAL HARMONY. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 111-120.

KEYWORDS : Music/harmonic spectrum/pitch perception/tone sensations/scale/sensory mechanisms/history/strings/small integers/real time

ABSTRACT : Musical judgements were examined in view of possible acoustic and sensory influences. First, listeners rated how well pitches in the chromatic scale followed a major triad, a major third, and a perfect fifth. Ratings reflected harmonics and subharmonic tone sensations. Psychoacoustic models of the ratings, differing in the number of factors considered, were formulated and tested. The correlation between predicted and actual ratings for the most successful model was .88 ($p<.01$). In Experiment 2, triads, thirds and fifths were paired with each other at various transpositions with respect to one another. Listeners rated how well the second element followed the first. Psychoacoustic models were tested, in which ratings corresponded to the sharing of octave-generalised harmonics and subharmonic tone sensations. The correlation between predicted and actual ratings for the most successful model was .81 ($p<.01$). In the third experiment, listeners judged tonal movement in sequences that changed key. Sequences were judged as conveying less tonal movement if key changes related well to the structure of the overtone series, suggesting that acoustic factors may influence the perception of broad levels of musical structure. Finally, short harmonic sequences were analysed by calculating acoustic relations between sequence notes and the overall key.

DOE, P.E. and FOSTER, C.G. (1987) : ENGINEERING CONSIDERATIONS IN ESTABLISHING A MANUFACTURING INDUSTRY FOR BOWED STRINGED MUSICAL INSTRUMENTS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 121-126.

KEYWORDS : Violin/musical instrument manufacture/timber properties/acoustics of violins/non-traditional materials/computer aided design and manufacture

ABSTRACT : The University of Tasmania's Conservatorium of Music and Department of Mechanical Engineering have embarked on a project aimed at manufacturing the violin family of musical instruments using indigenous timbers. Acceptable quality instruments have been produced by carefully matching the timber properties to the acoustic performance of the instruments. It remains to be seen whether the industry can be economically established.

CARTER, N.L. (1987) : SOME FIELD OBSERVATIONS OF HEARING LOSS, HEART RATE AND BLOOD PRESSURE IN GUN CREW FIRING 105 mm HOWITZERS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 127-137.

KEYWORDS : Noise perception/sound assessment/hearing impairment

ABSTRACT : Hearing levels, blood pressure and heart rate were measured in two gun crews before and after firing Hamel 105 mm howitzers. Each gun fired a total of 40 rounds during each of two test days, the guns often firing simultaneously. Two types of ammunition were used, that with the higher (noisier) charge being fired on the second day. All gun crew were fitted with E.A.R. earplugs, and noise measurements were made at each of the crew positions. The results indicate no effect on hearing but possible effects on heart rate and systolic blood pressure.

MAHAR, D.P. and MACKENZIE, B.D. (1987) : NON-AUDITORY AIDS FOR THE HEARING IMPAIRED : MODALITY-SPECIFIC DIFFERENCES IN THE PROCESSING OF SPATIALLY AND TEMPORALLY PRESENTED INFORMATION. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 139-147.

KEYWORDS : Tactile perception/visual perception/temporal information/spatial perception/reaction time/accuracy/hearing impaired/non-auditory aids

ABSTRACT : To assess whether tactile displays have any inherent advantages over visual displays as non-auditory aids for the hearing impaired, the relative efficiency with which spatially and temporally presented information is processed by touch and vision was investigated. Subjects identified which of two vibrations or two luminous bars was most intense. Within each modality spatial, temporal, and spatiotemporal presentation methods were used. The tactile-temporal and tactile-spatiotemporal tasks were performed more quickly and more accurately than the tactile-spatial task, while the visual-spatial and visual-spatiotemporal tasks were performed more efficiently than the visual-temporal task. This suggests that touch processes temporally presented information more efficiently than spatially presented information, while the reverse is true in the case of vision. As speech is primarily characterized by acoustic changes over time, these results favour a tactile-based strategy for the provision of non-auditory aids for the deaf.

MARSTON, B.G. (1987) : SELECTION OF ACOUSTIC BACKGROUND BY SPECTRAL CONTENT. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 149-157.

KEYWORDS : Atmospheric absorption/long range noise propagation/temperature/ relative humidy

ABSTRACT : Atmospheric absorption is an important aspect in the assessment of long range noise propagation (0.1 km – 10 km). It is a function of temperature, pressure, frequency and atmospheric composition. The effects of changes to these variables on predicted and measured dB(A) sound pressure levels are examined for the range 0°C to 40°C and for 5% R.H. to 100% R.H. The implications to the measurement of acoustic background and to the prediction of sound pressure levels are discussed with reference to the spectral content of a range of acoustic sources. Variations in predicted levels of between 2dB and 40dB have been observed and comments are offered on the reasons for these variations.

TICKELL, C. (1987) : DISTANCE ATTENUATION OF MINE VENTILATION FAN SOUND EMISSIONS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 159-166.

KEYWORDS : Noise impact studies/distance attenuation/mine ventilation fans

ABSTRACT : BHP Engineering has prepared noise impact studies for developments at various BHP Colliery operations in recent years. The characteristics of distance attenuation of sound for the sites were determined for predictions of sound levels in the surrounding community from proposed developments. Mine ventilation fans on the sites were used as the sources for these measurements. Sound levels were measured at regular distances up to 500 m from the fans. Results indicated that distance attenuation for the dominant frequencies of interest were between 4 and 5dB reduction for every doubling of distance rather than 6dB plus air attenuation as may be expected. This paper presents the results and comments on their implications for predictions of community sound levels from large sound sources with associated high gas flow rates (for example fans and gas-turbines).

WU QUNLI and FRICKE, F. (1987) : ESTIMATION OF THE PHYSICAL AND ACOUSTICAL PROPERTIES OF A DUCT USING MEASURED EIGENFREQUENCIES. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 167-175.

KEYWORDS : Duct/duct length/eigenfrequency/blockage/acoustic impedance/eigen-mode

ABSTRACT : The work presented in this paper, on the estimation of physical and acoustical properties of ducts, is based on existing eigenfrequency theories. The work starts with the estimation of duct length using measured eigenfrequencies. It is extended to include the detection of a partial blockage in a duct. The experimental results show that the eigenfrequency method is capable of estimating the physical properties of a duct. The use of measured eigenfrequencies to estimate the acoustical properties of materials is also discussed.

BYRNE, K.P. (1987) : THE USE OF WAVE IMPEDANCES IN CALCULATING THE ACOUSTIC PROPERTIES OF PLACE CONSTRUCTION. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 177-185.

KEYWORDS : Acoustic/wave impedance/transmission line/oblique wave/diffuse field/ sound reduction index/sound absorbtion coefficient/insertion loss
ABSTRACT : Acoustic lagging treatments, which are used to attenuate the noise radiated from vibrating surfaces, and fabric structures, which are being used more frequently in architectural applications, are examples of acoustic constructions whose acoustic properties can be found by using wave impedance principles. These properties include quantities such as insertion loss, diffuse field sound absorption coefficient and diffuse field sound reduction index. The constructions can be modelled with a limited number of elements, for example,layers which may or may not contain porous materials and porous or impervious sheets which may or may not be flexurally stiff. Formulae are developed which relate the wave impedances across these elements. These formulae, along with formulae which relate acoustic pressures across the elements, can be used to compute the quantities of interest. A comparison is presented which related computed and measured results for an acoustic lagging and an architectural fabric sheet.

DUNCAN, A.J., PENROSE, J.D. and GREEN, J. (1987) : A DIVER OPERATED ACOUSTIC SURVEYING SYSTEM. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 189-198.

KEYWORDS : Maritime archaeology/wreck surveying/acoustic position fixing/acoustic distance measurement/marine acoustics/underwater navigation/electronic distance measurement/diver navigation
ABSTRACT : This paper describes a wreck site surveying system developed for the Western Australian Maritime Museum. The system is diver operated and uses acoustic distance measurements to determine the location of points of interest on a wreck site relative to acoustic transponders located at known points on the sea bed. Measured distances are stored in semiconductor memory together with pressure measurements which are used to determine the vertical coordinates of the points of interest. Initial data processing by a microprocessor in the diver unit includes evaluation and averaging of acoustic transit times and weighted averaging of pressure signals to offset wave effects. All data are transferred to a small shipboard computer for processing and permanent storage when the diver surfaces. The system has been used in a major field program in the Abrolhos Islands off the coast of Western Australia, yielding an extensive set of results.

LAI, J.C.S. and DOMBEK, A. (1987) : COMPARISONS OF VARIOUS METHODS FOR DETERMINATION OF SOUND POWER LEVELS OF NOISE SOURCES. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 199-208.

KEYWORDS : Sound/sound intensity/sound level/sound power
ABSTRACT : Accurate determination of sound power levels for noise sources is important in prediction of installation noise levels, comparison of noise emissions between different machines and in devising noise control strategies. Three different methods have been used to measure the sound power of a noise source. The first method involves sound pressure measurements inside an anechoic chamber built to ISO standard while the second method employs a known reference sound power source to establish the sound power of the noise source. The third method applies the sound intensity technique to determine the sound power and sound intensity mappings will also be presented. Comparisons of the results obtained by these three methods will be made and the results discussed. Conclusions regarding the limitations and applicability of each method will be drawn.

ALFREDSON, R.J. (1987) : THE MOVING OVERLAPPING FAST FOURIER TRANSFORM AS AN AID TO PHONOCARDIOLOGY. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 209-215.

KEYWORDS : Heart/ausculation/heart sounds/cardiac cycle/murmur/click/FFT/ spectrum/cascade plot/diagnosis

ABSTRACT : This study has examined the merits of using the moving overlapping Fast Fourier Transform for converting heart sounds into a visual format which can be used to identify heart abnormalities. It includes a brief description of the functioning of the heart and an account of the origin of normal and abnormal heart sounds. Several different heart sounds were analysed and the resulting time dependent spectra were shown to be able to identify easily normal heart sounds and abnormal heart sounds such as clock and murmur. The significance of some of the smaller components in the spectra was less obvious. Work is continuing with a wider range of heart sounds and the importance of the infrasound associated with the heart is being investigated.

ALFREDSON, R.J. and PHELAN, I.J. (1987) : THE EFFECT OF MODAL PATTERNS OF VIBRATION ON NEAR FIELD ACOUSTIC INTENSITY. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 217-224.

KEYWORDS : Acoustic intensity/source identification/near field/vibration/piston/ranking

ABSTRACT : This study reports the first stage of a systematic investigation aimed at understanding the limitations and the validity of using near field acoustic intensity measurements as a means of identifying and ranking noise sources in a multi-source situation. Several different modal patterns of vibration of a baffle mounted circular piston were generated theoretically and the acoustic intensity calculated using numerical integration procedures. The results indicated that regions of significant vibrational activity were generally associated with higher near field intensities. Regions of low near field intensity were usually related to small amplitudes of vibration and associated anti-phase surface motion. There were nevertheless some unexpected patterns which could rise to misleading conclusions regarding the ranking of noise sources.

KOSS, L.L. (1987) : MASS DAMPERS FOR NOISE AND VIBRATION REDUCTION. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 225-232,

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KEYWORDS : Mass vibration absorber/vibration reduction/punch press/noise control/ impulsive operations

ABSTRACT : Three basic elements are used to control high levels of noise and vibration. They are stiffness, mass, and damping. This paper describes the use of a new type of mass damper to control noise and vibration in impulsively operated machinery such as punch presses and flying shears. The trade name of this damper is called 'Rodam'. The Rodam is placed between the moving slide and bed of a press to prevent the sudden unloading of the frame at the time of material fracture during a blanking operation. This action decreases the amplitude of the high frequency components of the force time history, thus reducing frame vibration amplitudes in comparison to the uncontrolled situation. Measurements made on a flying shears decking mill with and without a Rodam in place have shown that the Rodam has provided an eight decibel vibration reduction along with a two dB(A) peak noise reduction. The paper will describe the technical operation of the Rodam and its use in the flying shears machine.

LA FONTAINE, R.F., SHEPHERD, I.C. AND CABELLI, A. (1987) : TREATMENT OF CENTRIFUGAL FAN TONAL NOISE. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 233-239.

KEYWORDS : Centrifugal/fan/resonator/cutoff/tube quarter-wave/tone/reactive/silencer/perforate

ABSTRACT : A method of reducing the blade passing tones of centrifugal fans with a resonator mounted on the fan cutoff is discussed. Few guidelines are available for the design of these devices and no significant theoretical treatment has been published. Since the optimum configuration is not always a quarter-wave tube the designer must experimentally determine resonator performane in every new application. A summary of the tests conducted on two dissimilar centrifugal fans demonstrates the complexities of implementation.

TANDON. N, (1987) : THE EFFECT OF CONFIGURATION OF SOUND INTENSITY MEASUREMENTS ON SOUND POWER DETERMINATION. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 241–247.

KEYWORDS : Noise/sound intensity/sound power

ABSTRACT : The paper describes the sound intensity measurement technique using two microphone probel, dual channel real time analyzer and microcomputer. Sound power of a sound source has been determined by this method. The effect of the measurement distance and the configuration of points on the surface, on the sound power level is investigated. Fairly accurate results are obtained in the near field as well as at larger distances from the source. The comparison of the scanning and point by point method shows that better results are obtained by point by point method of measurement provided sufficient number of points are taken on the surface.

DAVIS, M.R. (1987) : OBSERVATION OF ACOUSTIC AMPLITUDE AND IMPEDANCE BY LASER DOPPLER METHODS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 249-258.

KEYWORDS : Sound/sound level/wave impedance/acoustic

ABSTRACT : This paper describes the acoustical use of the methods of laser Doppler anemometry. Seeding particulates suitable for laser-Doppler use are generally found to have satisfactory and accurate acoustic response in air and in water. The achievable signal to noise ratio in laser Doppler systems places a constraint on system resolution, and methods based on analysis of Doppler signal spectra are therefore preferable. Such an approach thus provides statistical average information relating to the acoustic field. Under pure tone acoustic conditions the system dynamic range is determined by the requirement to analyze the sidebands generated in the Doppler spectrum. However acoustic amplitude can be determined to 0.1 dB or better. Determination of the phase between acoustic pressure and velocity is possible by simultaneous amplitude and frequency modulation of the Doppler signal combined with the pressure signal, and regression analysis of asymmetric sideband peaks allows determination of the phase to within a standard error of 1°.

SUN, C. and FRICKE, F. (1987) : SIMPLE METHODS FOR MEASURING REVERBERATION TIME AND ABSORPTION COEFFICIENT. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 259-266.

KEYWORDS : Acoustics/reverberation time/acoustical absorption coefficient/simplified measurement methods/in-situ method/noise control measurement/transmission loss measurement

ABSTRACT : In this paper, measurement methods for absorption coefficient and reverberation time are briefly reviewed. A simplified method of measuring reverberation time is described which uses tape recorded repetitive white noise or filtered white noise pulses to excute the enclosure. A range of time intervals between pulses must be used in order that the maximum and minimum sound level during the pulse cycle may be read from a general purpose sound level meter. This method can be used for on-the-spot assessments of rooms with reverberation times greater than one second. The maximum error is less tha n 20%. A simplified in-situ method of measuring absorption coefficient is based on the coherence of incoming and reflected signals close to the surface of the material. If the microphone is very close to surface the sound level reflected from a perfectly reflective surface should be 6 dB greater than the sound level without the surface present and the sound level in front of a perfectly absorptive surface should be the same as that without the surface. The present experimental results have shown an acceptable degree of accuracy from 250Hz to 1kHz, for thin porous materials.

WATKINS, D.J. and DALE, P. (1987) : THE DEVELOPMENT OF AN IMPROVED NOISE MONITORING SYSTEM. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 267-273.

KEYWORDS : Noise/noise monitoring system/background noise

ABSTRACT : The accurate long term monitoring of background sound levels is an important part of assessing industrial noise in Victoria. Recently the Victorian Environment Protection Authority developed a new noise monitoring system using a sound level meter and a Sharp PC–1600 portable computer. The system has the advantage of being composed of readily available and relatively inexpensive components. It can operate completely automatically for long periods of time and has a very low power consumption. The monitoring system can be used for both short and long term measurements and provide a variety of noise indices as well as a chart recording of the measured noise level. The Sharp PC is compatible with IBM computers and data collected during field measurements may be transferred for storage and subsequent processing on a larger computer.

SAMUELS, S.E. (1987) : SOME ASPECTS OF ROAD TRAFFIC NOISE AT INTERSECTIONS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 275-285.

KEYWORDS : Sound/highway/speed/mathematical model/forecast/traffic/highway. urban area

ABSTRACT : Reliable techniques are available in Australia at present for the prediction, measurement, assessment and control of road traffic noise. However, these are all based on the assumption of freely flowing traffic conditions. In many cases when the traffic flow is interrupted, for example in the vicinity of intersections, this assumption is invalid and the established techniques are generally unsuitable. No alternative techniques to handle such cases are available. Recent research at ARRB has therefore been concentrating on traffic noise under interrupted flow conditions, with particular emphasis on signalised intersections. A descriptive and predictive model, which is both theoretically and empirically based, is being developed and evaluated. A brief outline of the model is given in the paper, along with some details of the inherent differences in traffic noise under interrupted and freely flowing conditions. The development and evaluation program is also mentioned and some potential applications of the model considered.

STEWART, A.C. and SANDBERG, U. (1987) : PREDICTING SWEDISH TRAFFIC NOISE UNDER A MODIFIED AUSTRALIAN MODEL. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 287-295.

KEYWORDS : Road traffic noise/Sweden/computer model/prediction/validation/population noise exposure/noise dose-response
ABSTRACT : In order to estimate on a regional scale historical changes in road traffic noise exposure, as well as future predicted exposures, it is wise to use a computer model. This paper describes the adaptation to Swedish conditions, and improvement, of a model originally developed to predict traffic noise in Sydney over the years 1965 to 2015 as reported in the 1986 AAS conference. In most computer models there is a conflict between striving for rigorous accuracy on the one hand and keeping the data input requirements and model's complexity down to manageable proportions on the other hand; this model strikes a good balance and the trade-offs in its design and adaptation are discussed. Results are presented of the application of this model to three areas in Sweden representing fairly well the cases of rural, small town and city areas. It seems possible to restore the noise environment by the year 2000 to that of the early 1960s but only if a number of measures are used together such as EEC vehicle noise standards in 1990, bypass roads and low noise road surfaces. The model was validated in that its results gave fairly good agreement with manual calculations and previously published work.

KENNA, L.C. (1987) : ENVIRONMENTAL NOISE FROM AIRCRAFT OPERATIONS. Proc. 1987 Conf. Aust. Acoustical Soc. pp. 297-305.

KEYWORDS : Environmental noise/aircraft/aircraft engines/noise abatement procedures/ noise certification/ANEF system/Australian Standards/helicopters/airships
ABSTRACT : The 1980s has been a decade in which notable progress has been made in reducing the environmental noise heard in the vicinity of airports from aircraft operations. The changes have been due to a combination of technical developments, revised operating procedures and regulatory requirements. The paper outlines the contribution of each of these components to the overall situation and illustrates the changes which have been brought about. This has also been a decade in which a combination of planning and assessment tools, specifically designed for Australian conditions, has become available for land use planning and building design in relation to aircraft noise. These comprised the Australian Noise Exposure Forecast system which was researched, developed and brought into use in the early 1980s, and the publication in 1985 of a revised version of AS 2021, the SAA's code for building siting and construction to avoid aircraft noise intrusion. There have been other developments which have been, or may become, a cause for concern. These have included the increasing use of helicopters in urban areas, the introduction of new types of aircraft such as ultralights and airships, and the proposed introduction of new aircraft power plants such as the unducted fan engine.