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FOREWORD


The publications of the Congress include over 850 abstracts of the contributed papers on all branches of acoustics, published as submitted, in two volumes. Ten invited lectures, suggested by ICA, are published in full version in a separate volume.

The full version of the specialized lectures presented at the Barcelona Symposium on "Sound recording and reproduction" and at the Sevilla Symposium on "Hearing and industrial noise environment" are also published in separate volumes.

In accordance with the directions of the ICA, one page only was allocated to each abstract and the publication is to be ready to be sent to registered participants in advance of the Congress. They will also have the possibility of having, with the permission of the authors, a photostatic copy of the full version of the paper, at cost price.

Volume II of the Contributed Papers contains also the abstracts of the specialized lectures of the Satellite Symposia in Barcelona and Sevilla. It also includes summaries of papers of the Special Sessions on coordinated research promoted by ICA on five subjects related with Noise Pollution, of interest to the Scientific Committee on Problems of the Environment (SCOPE), a body of the International Council of Scientific Unions (ICSU).

Madrid, March 1977

Prof. Andrés Lara-Saenz
President of the Congress
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1. *Environmental acoustics*

A. *ACOUSTIC CRITERIA*
Acoustical parameters of concert halls are typically determined from impulse responses, "echograms" or decay curves. They are obtained by exciting the hall with test signals such as electrical sparks, tone bursts or switched noise. These test signals are convenient only for use in empty halls; they are "uncomfortable" (or even unacceptable) in the presence of an audience.

An elegant method to overcome these problems is to utilize the music played on the stage as a test signal [1]. To obtain impulse responses, the music is recorded on the stage and at a seat in the audience. The received signal $r(t)$ in the audience is given by

$$r(t) = s(t) * h(t) + n(t),$$

where $h(t)$ is the (unknown) impulse response between stage and audience seat and $n(t)$ is a (likewise unknown) noise (air conditioning, people talking etc.). A maximum-likelihood estimate of the Fourier transform $\hat{H}(\omega)$ of the impulse response is given by [1]

$$\hat{H}(\omega) = \frac{\sum R(\omega) \cdot S^*(\omega)}{\sum |S(\omega)|^2},$$

where $S(\omega)$ and $R(\omega)$ are the Fourier transforms of the stage and received signals, respectively. The product $R(\omega) \cdot S^*(\omega)$ is the Fourier transform of the crosscorrelation function between $r(t)$ and $s(t)$. The summations indicate averaging over different time segments of the signals.

To test this method in a concert hall, a reverberation-free, two-channel music signal was radiated from the stage via two loudspeakers (simulating an orchestra). A moving noise source was used to simulate the noise $n(t)$. The stage signal $s(t)$ was recorded with a supercardoid microphone, the audience signal with condenser microphones in the "ears" of a dummy head. Both signals $s(t)$ and $r(t)$ were subdivided into time segments of 1.6 s duration and processed according to equation (2).

The computations show that a summation over 50 (or more) time segments leads to useful impulse responses. Evaluation of these impulse responses yields acoustical parameters, such as reverberation times, that are in excellent agreement with values measured by standard methods under identical conditions.

REFERENCE

DIGITAL SYNTHESIS OF SOUND FIELDS: A NEW METHOD TO STUDY THE INFLUENCE OF SINGLE ACOUSTICAL PARAMETERS ON PREFERENCE JUDGEMENTS

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In case of N point sources on the stage of a concert hall the signals at the two ears of a listener are described by

\[ s_{L,R}(t) = \sum_{i=1}^{N} \int_{-\infty}^{\infty} m_i(t') h_{iL,R}(t-t') \, dt' \]

\( s_{L,R}(t) \) the signal at the left (right) ear, \( m_i(t) \) the signal of the \( i \)-th sound source, \( h_{iL,R}(t) \) the impulse response between the source \( i \) and the left (right) ear.

Using stereophonic (N=2) reverberation-free music \([1]\) as source signals this sum was evaluated with a digital computer. The impulse responses were calculated from spark echograms recorded in concert halls with an artificial head. All signals were digitized with a sampling rate of 26 kHz and an amplitude resolution of 12 bit. The impulse responses were modified to change only single acoustical parameters at a time while all other parameters were held constant. The resulting recordings were presented to 15 experienced subjects for preference judgements in paired-comparison tests.

The main results are:

1. The early decay time (EDT), derived from the reverberation curve between 0 and 10 dB \([2]\) is more important for the subjective preference than the classical reverberation time (0 - 60 dB; 5 - 35 dB).

2. The interaural crosscorrelation (ICC, defined as the absolute maximum of the normalized interaural crosscorrelation function within \(|\tau| < 1 \text{ ms} \,[3]\)) has only a strong influence on the preference if ICC > 0.5 \([4]\).

3. For each parameter there is an interval of indifference instead of an optimal value. The contribution of a parameter to the subjective preference depends on values outside that interval: the more the parameter differs from it the more the sound field is disliked.

4. If the parameter is varied only within the interval of indifference it does not affect the subjective judgement. In this case the preference is very often related to spectral differences of the sound fields. These may be measured by the energies at low and high frequencies.

\[ [1] \text{Burd, Rundfunktechn. Mitteilungen 13 (1969), 200} \]
\[ [3] \text{Damaske, Ando, Acustica 27 (1972), 232} \]
INTRODUCTION

L’acoustique des salles classiques nous propose un certain nombre de méthodes permettant de mesurer quelques unes des variables qui conditionnent la qualité acoustique d’une salle : durée de réverbération, coefficient d’absorption... Mais l’expérience pratique montre que ces données sont insuffisantes lorsqu’il s’agit de définir la qualité d’un local relativement à l’intelligibilité de la parole, ou son adaptation aux prestations musicales. Une longue suite de recherches et d’expérimentations nous a montré que pour définir de façon réaliste les particularités sonores d’une salle, il fallait la tester avec la réalité sonore (parole et musique) dans les conditions normales d’emploi, et imaginer des méthodes pour objectiver les résultats obtenus. Voici les deux tests principaux que j’ai mis au point : méthodes et appareillages.

MÉTHODE ET APPAREILLAGER

Pour définir l’adaptation d’un local à la parole, tout en évitant les effets de la prévisibilité qui permettent la reconstitution mentale des mots mal perçus, on prépare un "discours synthétique." On prend un texte de quelques lignes, et tout en conservant la longueur des mots successifs, on interchampe les syllabes de ces mots afin d’en faire des mots "nouveaux", de contenu phonétique identique à celui de la langue considérée, mais qui, n’étant pas connus, ne peuvent être reconstitués mentalement par un auditeur. On diffuse alors ce texte dans le local à étudier, et on enregistre simultanément en divers points du local le discours tel qu’il est perçu aux points considérés. On fait ensuite écouter ces enregistrements à des sujets de test, en salle insonore, et on compte pour chaque cas le nombre de syllabes correctement perçues. Le pourcentage de ces syllabes est multiplié par deux, et on obtient ainsi un taux d’intelligibilité valable pour des discours normaux (où la prévisibilité joue), qui s’est avéré être beaucoup plus réaliste que le taux obtenu par d’autres méthodes.

Pour définir l’adaptation d’un local à la musique, on diffuse des œuvres complètes de musique de type envisagé—soit directement (concert) soit à l’aide d’une chaîne de haute-fidélité. On enregistre simultanément l’œuvre à diverses places de la salle et on mesure ensuite en pourcentage la répartition de l’énergie dans un certain nombre de bandes "sensibles." Ces bandes ont été déterminées expérimentalement à l’aide de tests faits avec la participation de sujets, musiciens ; l’opération consiste à couper telle ou telle bande lors de l’écoute de musiques de types divers, et à chercher à équilibrer le contenu informationnel de ces bandes dont les fréquences limites sont : 50-300-400-600-1200-1800-3500-6000 et 15000 Hz. Un intégrateur électronique fournit le % d’énergie contenue dans les bandes ainsi déterminées, et on reporte les valeurs sur un diagramme, hautement significatif du jugement subjectif sur la "sonorité" d’une salle—ou d’un instrument. Un appareillage spécifique a été construit pour donner directement le diagramme relatif à un événement sonore de longue durée, de nature quelconque (l’Intégrateur de Densité Spectrale : IDS).

RESULTATS ET CONCLUSIONS

La méthode précitée ci-dessus a été appliquée à l’étude de nombreuses salles, et il a été possible grâce à elle de prédir les jugements formulés par les auditeurs usagers des salles considérées. Son intérêt est de partir de la réalité de la parole et de la musique plutôt que d’artefacts (simulations, bruits blancs...). On peut l’appliquer à d’autres recherches (qualité de la voix; qualité des chaînes électro-audio-techniques destinées à diffuser de la musique...).

Références :

ANALYSIS OF SUBJECTIVE JUDGEMENTS OF LISTENING CONDITIONS IN CONCERT HALLS

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INTRODUCTION

Subjective investigations of listening conditions in concert halls cannot manage nowadays without modern statistical procedures of data processing. Statistical analysis prepares the ground for studying the relations between subjective and objective characteristics of listening conditions.

TEST PROBLEMS

The arrangement of subjective experiments is of great importance. Listening tests are made either directly in a hall or indirectly by means of stereo-records provided by an artificial head. In the first method, the defined subjective parameters are judged on prescribed rating scales, in the second one, the records are compared according to their mutual dissimilarity. Testing procedures are connected with many problems as for instance: the selection of listeners and of the type of music, the manner of its presentation, the cyclic shift of listeners in the hall, the list of subjective parameters, etc.

DATA PROCESSING

Results of subjective tests in the case of the direct method are processed by parametric statistical procedures which also include the factor analysis and in the case of the indirect method (using the artificial head) by non-parametric procedures including multidimensional analysis. Unambiguous solution and easier interpretation of results could be achieved by rotation of the principal solution according to a suitable criterion.

The first one of the findings is the number of independent factors or psychological dimensions involved in a set of subjective judgements. For instance, the results of our experiment (triadic comparison of 8 records of the same violin passage played under different listening conditions in the same hall) tend according to the method described under (1) to 4 dimensions. See Fig.1. Geometrical representation of the solution (configuration of subjective parameters in factor space or configuration of stimuli in multidimensional space) suggests the first ideas about physical causes of the inter-relations among the parameters or stimuli.

REFERENCES

(1) J.B.Kruskal, Psychometrika, (1964)29, No.1, 1.
NEW RESULTS ON THE HUMAN HEAD DIRECTIVITY IN SPEECH EMISSION

Moreno, A. 
Instituto de Acústica 
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INTRODUCTION

We can state that since 1939 (1), important progresses in determining human head directivity have not been achieved, either in obtaining more complete results or in developing more powerful methods of analysis.

THEORY

The method of analysis derived from anamorphical relations between two signals, as indicated in previous papers (2)(3) dealing with electroacoustical radiating systems, has proved its suitability in measuring human head directivity. This physical magnitude corresponds to the anamorphical relation between instantaneous speech signal levels picked up in two points, one moving around a circumferential path centered in the speaker's mouth and the other one resting at a zero azimuth position.

RESULTS AND CONCLUSIONS

Following these lines, graphical directivity patterns for an ensemble of ten people under normal fluent Spanish speech, in both vertical and horizontal planes have been obtained in anechoic room. The 1/3 octave frequency bands extending from 100 to 4000 Hz (ISOR. 266) and the whole signal have been explored. In all cases, these directivity patterns show a high analogy of human head with other radiating systems, in a slightly different manner than that derived from results of other authors (1)(4)(5). Well defined lobes continuously varying with increasing frequency and related to specific body parts have been encountered. Directional patterns of a standard person, set up for a radius of 1.4 m, are shown in the figure. Some of the above mentioned characteristics and an indication of the quality of the method can also be seen.

Directivity diagrams of all people analyzed are well related to each other. So we have obtained the mean directivity diagrams inside each 1/3 octave frequency band.

Another not less important conclusion is that this method of analysis is a direct, powerful, and easily operated procedure for obtaining directivity patterns of a particular person especially well fitted for studies where this individual characteristic is needed.

(2) A. Moreno, J. Pfretzschner. Elec. Fis. Apl. (1971) 14, 369
(3) A. Moreno, J. Pfretzschner, M. Romera. 8th ICA London (1974) 107
(4) F. Trendelenburg. Z. Tech. Phys. (1929) 10, 558
(5) I. Janůška. Electroacoustique (1968) 13, 5
In spite of evident progress in teaching process no programming and/or cybernetics may fully replace the live spoken word. Consequently, it is of overwhelming importance for the pupil (student) to hear, perceive and understand each word coming ex catedra.

Hence improving intelligibility of classrooms and amphitheatres, the working place of one-fifth of the whole population, met large understanding in educational forum of Cluj-Napoca municipium (Romania); subsequently long term investigation was run in a scholar campus of the town (High-School No. 15).

Computation as well as tests with logatoms gave similar average values, PA = 78-86%; in addition, there are significant differences in intelligibility depending on the sitting of pupils: PA = 86% for the first, down to 70% for the bottom rows.

EFFECTS OF DECREASED INTELLIGIBILITY

If intelligibility of about 80% may be considered to be quite good as far as theatres are concerned, in teaching, however, such PA does not seem satisfactory. Paradoxically, we have to accept that at school sentences lack sufficient internal connection, since the lessons contain a lot of new elements: notions, terms, proper nouns. And still more difficulties arise when teaching languages. Consequently, reconstruction of the sentence is difficult and pupils are obliged to make supplementary effort in order to simply perceive, having left but little concentration for assimilating.

The supplementary effort involves premature fatigue in pupil and even in the teacher who will have to resort to a tiresome, loud and very distinct pronunciation of the syllables.

Hence the nonfavourable acoustical factors involve low speech rate, deficient assimilation of new notions, slowing down of the teaching process.

IMPROVED CLASSROOMS — CONCLUSION

In order to demonstrate the effect of acoustical parameters, a follow-up study was carried out: parallel survey of collectivities working in normal and others in acoustically improved classrooms. Indices (see table) for parallel classes at High-School No. 15, averaging a two-year period, show improvement of the teaching efficiency in acoustically treated classrooms, as compared with common ones.

<table>
<thead>
<tr>
<th>Matter</th>
<th>Index of improvement</th>
<th>Matter</th>
<th>Index of improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Languages:</td>
<td>Romanian 0.36</td>
<td>Physics</td>
<td>0.17</td>
</tr>
<tr>
<td></td>
<td>English 0.29</td>
<td>Natural sciences 0.08</td>
<td></td>
</tr>
<tr>
<td></td>
<td>French 0.18</td>
<td>History</td>
<td>0.09</td>
</tr>
<tr>
<td>Mathematics</td>
<td>Chemistry 0.24</td>
<td>Geography</td>
<td>0.09</td>
</tr>
<tr>
<td></td>
<td>0.29</td>
<td>Music</td>
<td>0.26</td>
</tr>
</tbody>
</table>

Improved acoustics followed by a reasonable placing of pupils in the bench rows may lead to higher intelligibility all over the room, more activity in pupils with lower auditive acuity, finally increasing and homogenizing results in educational process.
Einleitung

Gegenstände in Räumen beeinflussen die Schallpegel nicht nur durch die Absorption, sondern auch durch die Schallstreuung. Dieser Einfluss wurde theoretisch und experimentell untersucht.

Theoretische Grundlagen

Ausgehend vom mittleren Schicksal eines Energieteilchens in einem Medium mit Streukörpern wurde die Schallausbreitung in Räumen verschiedener Formen untersucht. Dabei wurden geometrische und statistische Betrachtungen kombiniert.

Experimentelle Untersuchungen

Die theoretisch gewonnenen Ergebnisse wurden durch Messungen der Schallausbreitung aus Schallquellen bekannter Leistungen überprüft. Darunter wurden auch einige Modelluntersuchungen durchgeführt.

Ergebnisse und Schlußfolgerungen

Die von den Streukörpern reflektierte Schallenergie ruft in einem ziemlich grossen Raumbereich eine Pegelerhöhung hervor, die manchmal mehr als 5 dB betragen kann. In sehr grossen Abständen von einer Schallquelle nimmt der Schallpegel stark ab. Der Einfluss der Schallstreuung auf die Schallausbreitung ist am grössten in ausgedehnten geschlossenen Räumen mit gleichmässig verteilten Streukörpern, am geringsten in bebauten Gebieten /Ortschaften/ oder Industrieanlagen in Freien.

Since noise was first recognized as a serious environmental pollutant, a number of social surveys have been conducted in order to assess the magnitude of the problem and to develop suitable noise ratings, such that, from a measurement of certain physical characteristics of community noise, one could reliably predict the community’s subjective response to the noise.

In all the surveys, however, the correlation between the measured noise and individual subjective reaction was poor, though when group responses were pooled, the correlation between the noise and the median response was much improved.

Much has been made, recently, of the importance of intervening, non-acoustical variables, such as attitudes, fear, hostility, etc. in determining the amount of annoyance felt by people exposed to noise; so much attention has been given to constructing annoyance measures, with varimax analyses of non-acoustical factors, that we have almost lost sight of the noise at which the annoyance is supposedly directed.

When the noise exposure is felt to be extreme, however, people seem to have little difficulty in sorting out their feelings about the noise from their other, non-acoustical attitudes. For planning and monitoring purposes, then, the percentage of the population who are "highly annoyed," when plotted against some measure of the noise exposure, is a more stable indication of community response than the "mean annoyance" of the community.

Recently, the author has reviewed the data from eighteen social surveys concerning the noise of aircraft, street traffic, expressway traffic and railroads [1]. Going back to the original published data, the various survey noise ratings were translated to day-night average sound level, and an independent judgment was made, where choice was possible, as to which respondents should be counted as "highly annoyed".

The results of eleven of these surveys show a remarkable consistency, as shown in Fig. 1. The average of these curves is given in Fig. 2; it is proposed that this relationship is the best currently available for predicting community annoyance response to noise of all kinds.

The responses from the remaining surveys scatter widely (see Fig. 3), usually for reasons that are understandable (e.g., the highest two of only four or five categories were counted as highly annoyed, the survey was conducted only in summertime, etc.). However, even these results, when averaged, agree with the curve of Fig. 2.

References:
A SYSTEMIC STUDY OF THE HUMAN EFFECTS OF EVERYDAY OCCURRING IMPACT NOISE

Powell, J.A. Reader in Building Utilisation, School of Architecture, Portsmouth Polytechnic, King Henry 1st Street, Portsmouth, Hampshire, England

INTRODUCTION

In Britain building engineers and architects have no suitable criteria to enable them to design floors, walls and screens that will efficiently isolate unwanted impact noise. As a first step towards rectifying this situation a systemic testing technique has been designed to monitor the human discomfort caused by noise in homes and offices; the aim was to define limits of aural discomfort.

THE SYSTEMIC STUDY

In this systemic study the aim was to explore real-life-man-environment systems as a whole without attempting to control in any way individual variables (as is normally the case with traditional scientific experiment); rather the objective is to monitor all variables, including human response, as gross constraints are imposed or relaxed on the system.

In these studies the system under consideration was the impact noise-typical user (and his responses)-users own environment system. In short, commonly occurring impact noise was injected into the environments of typical users and the necessary variables monitored. The technique adopted automatically builds up a picture of individuals real responses to noise in their own environments - a natural history of peoples actions, behaviour, physiological stress and attitudes when continually bombarded with noise. Powell (1973) has given a detailed statement of the testing technique elsewhere.

RESULTS AND CONCLUSIONS

Early results suggest that the way individuals and groups respond to their environment is largely governed by their expectation of that environment what they feel is reasonable for that situation.

In offices individuals are prepared to spend time in fairly noisy conditions for long periods of time. For office users it is the loudness (weighted energy) of the intruding signal which seem to govern response. The twenty-fifth percentile aural environment free from discomfort occurs when the Noise Pollution Level is 58.5 dB. If these levels are exceeded in work environments individuals can exhibit a characteristic syndrome associated with tiredness and irritability; decrease in urinary I7 ks.; and changes in cellular composition of blood.

In flats the individual will not tolerate the same high levels of noise. Here any noise which intrudes on an individuals privacy is likely to cause annoyance. The noises which flat dwellers judged to be unacceptable were only just detectable, or in terms of signal detection theory, the twenty fifth percentile aural environment free from discomfort occurred when 10 log d'' was 4.5 dB.

Understanding of man's response gained from these studies led to the development of criteria for design in flats and offices. These criteria should enable designers to gauge the effects of impact noise on man and to design more effectively against it.

REFERENCES

A GENERAL APPROACH TO NOISE ANNOYANCE

Mantel, J. Munich and Haifa

Noise levels ($L_A$ in $\text{dB}(A)$) can be corrected to an annoyance level ($L$ in $\text{dB}(A)$) by correction factors for the temporal properties of the noise ($\Delta L_T$), correction factors for the rhythm of recurring noise ($\Delta L_R$), a correction factor for the tonality of the noise ($\Delta L_T$) and a correction factor for the special information content of the noise (alarm, screaming of children) ($\Delta L_{\text{INF}}$).

Between the annoyance level or corrected noise level in $\text{dB}(A)$ and the estimate percentage of listeners annoyed by the noise there is a direct relation which lead to

a) the introduction of the concept of mean indifference level which is tolerated by 50% of those interviewed

b) an arbitrary division of people into three groups according to their sensitivity to noise:

- intensive 20%
- normal 60%
- sensitive 20%

where by the extreme groups differ from the "normal" group by 8 $\text{dB}(A)$ on the average.

The total fluctuation range for most types of noise is 30 $\text{dB}(A)$. The basic pattern of the relations can be seen from the table and figure.
Environmental assessments of noise climates in major metropolitan areas have been a useful tool in presenting large cities with accurate baseline values of existing noise levels. Urban noise survey analyses have been the concern of Armour Research Foundation (ARF) now IIT Research Institute (IITRI) and the City of Chicago as far back as the late 1940's. This paper will present an overview of results from earlier noise surveys and compare these data to assessments performed in the Chicago area during the past 5 years.

The earliest noise survey performed in Chicago took place in 1941 under the direction of Dr. H. Leedy of ARF. Noise readings were made in the greater Chicago area, with Dr. Leedy concluding that the dominant source of noise was traffic oriented. The methodology used in 1941 was very similar in nature to that used today. Dr. Leedy noted "at a given station the noise level was usually recorded for a 10-minute period and from the record thus obtained, the maximum and minimum noise levels determined" (1). Dr. Leedy in the talk referenced below called for an increase in public interest in noise and a large study to assess the environment so as to set up a city noise code. This leads us to the next chapter in the noise survey story.

From 1948 to 1950, H. Hardy, and G. Bonvallet both of ARF undertook an extensive survey of existing noise climates in the Chicago area. The purpose of the survey was to determine acoustic levels in various zoning districts in order to place noise limit restraints on various district boundary configurations. The results of the 2 year Bonvallet and Hardy study became part of Chapter 194 of the Chicago Municipal Code. From 1971 to 1973 the City of Chicago Dept. of Environmental Control reassessed district boundary noise levels in order to note if the new comprehensive noise code promulgated in July of 1971 was an accurate descriptor of existing Chicago noise climates. The octave levels used in the 1971 code revision are identical to the Chapter 194 numbers from Bonvallet and Hardy's study. Until recently no hard copy printout of the 1948 to 1950 study was available for comparison to the 1970 data. In July 1976 the original Bonvallet and Hardy data was found buried under a mass of old papers in IITRI's noise lab. A comparison between the 1973 Caccavarri and Schechter Study (2), and Bonvallet and Hardy's data for the downtown Chicago area is presented in the accompanying figure. Allowing for differences in weighting and time duration the two data results are almost identical.

It can be concluded then that utilization of the Bonvallet and Hardy data as presented in Chapter 194 of the City Code in the new 1971 Noise Regulation was valid and that noise levels in Chicago over the past 25 to 30 years may not have been rising at the alarming rate once supposed. It should be stated that the 1973 data taken at State & Madison is representative of noise levels 1.5 years after the promulgation of the 1971 regulation. In some instances 1971 data for this location are 5dB higher.

With effective implementation on the Chicago Noise Regulation we are at least as quiet as 1948. Further implementation of noise control principles in Chicago will hopefully produce a quieter environment in which to live and work.

(1) Presented at 26th Meeting of Acoustical Society of America, Oct. 24-25, 1947, N.Y.
(2) Presented July 1973, Air Pollution Control Association Meeting, Chicago, Ill.
DAILY NOISE EXPOSURE OF PEOPLE IN JAPAN

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INTRODUCTION

Daily noise exposure is one of the fundamental problems in dealing with the environmental noise and appears in U.S. EPA document. This paper gives the results of preliminary survey, which was carried out in Sendai city of 620,000 population for daily exposure of people to acoustic stimuli. It shows a variety, as a matter of course, depending on the occupations, places of residence, family size and so on. Most people seem to live in noise environment of 65 to 75 dB(A) in Leq. 

MEASURING APPARATUS

Two kinds of equivalent sound level meters were developed for the purpose of this survey. One of them indicates 24 hour noise exposure index. It has a weight of 468 grams including 1/2 inch electret condenser microphone. A quartz timer is built in the apparatus and the measurement ends automatically at 24 hours after its start. Noise from 40 to 110 dB(A) can be measured by this meter and the noise exposure index is directly indicated by a number of seven figures. Another meter can store Leq in every 10 minutes over 24 hours. 144 values for 24 hours are typed out by connecting the meter to the decoder. A weight of the meter is 445 grams including 1/2 inch electret condenser microphone. These meters are carried by subjects in the survey.

SURVEY

As to housewives four areas were selected for every district of four categories according to the Noise Regulation Law of Japan and three subjects were sampled from each area. On the other hand, 168 people were selected according to their occupations, places of offices or industries, and places of their residents. Fig.1 shows an example of a subject who lives in the residential area and works at a university located in an educational district.

As another way to know the daily noise exposure, the equivalent sound levels for 30 minutes or an hour were surveyed in various environments, and Leq were estimated from "1975 Survey on the Use of Time in Japan" issued by the Japan Broadcasting Corporation (NHK). 

This survey is carried out under the scientific research programs of the Ministry of Education and the support of Sendai City, and is now continued.

REFERENCES

(1) U.S. EPA Document 550/9-74-004
(2) M. Hashimoto, INTER-NOISE 75 Proc.; (1975)719
(3) NHK Public Opinion Poll Lab., 1975 Survey on the Use of Time in Japan; (1967)
NOISE LEVEL AS AN AREA CHARACTERISTIC

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INTRODUCTION

Where measurement of a background level of noise is not convenient BS4142: 1967 (amended 1975) suggests a method for predicting background noise level as a characteristic of an area and a time of day in assessing the likelihood of annoyance from a source of industrial noise in a mixed residential area. However the background level is excluded from noise units that are cited in legislation against noise because it is difficult to predict. This paper discusses the concept of background noise level and examines the results of a large number of measurements of background noise levels made throughout the U.K. by Open University students (1).

PREDICTIVE MODELS

Six predictive models can be derived from this data expressing any one of mean minimum level; L10; L90; Peak level; L10 - L90; L10 - mean minimum level; (all in dB(A)) as a linear combination of region (5 categories); type of area (6 categories); time of day (6 categories); type of nearest major noise source (8 categories) distance (estimate of measuring position from nearest major noise source (9 categories).

CONCLUSIONS

(i) That noise levels tend to increase with increasing urbanisation but that there is little difference between urban and suburban areas,

(ii) that there is a substantial difference between L90 (which is usually described as the background level) and mean minimum level which tends to be more characteristic of an area type and more independent of other factors,

(iii) that variability of noise level (L10-L90) tends to be maximum where (i) railways or aircraft are the noise source (ii) in a rural area and (iii) where the observer is close to the noise source,

(iv) that minimum variability occurs in an industrial area, where industry, or construction or 'other' activity is the main noise source and where the observer is some distance from the source (201-500 m),

(v) that the inability of the models to explain a large proportion of the variance of reported noise levels may be related to (a) interaction between variables, (b) important parameters that determine the characteristic noise level of an area and that have not yet been identified and (c) the poor categorisation of area.

REFERENCES

INTRODUCTION

Noise from construction and demolition sites can be a nuisance and consequently many countries have approved legislation to control such noise. This paper gives guidance as to the viability of applying construction site noise prediction techniques to show compliance with any specified noise criteria.

NOISE PREDICTION

In order to ascertain whether the noise levels from the construction techniques to be adopted on a given site are likely to exceed any noise limits imposed, the contractor or developer will need to estimate the expected noise levels. These predicted levels are required at a very early stage, because if noise control is deemed necessary provision should be included in the tender price. Further predicted levels may be required at other stages of the contract due to changes in the methods of working.

Prediction of Leq noise levels at the various stages in the construction programme using plant sound power level data similar to that contained in the current UK Code of Practice BS 5228:1975 (1) have, in most cases, yielded levels which are considerably higher than levels measured on site. The primary reasons for such discrepancies are: lack of time information, lack of precise noise level data and lack of precise plant details and positioning of working time at maximum level.

However, research has shown that it is possible to carry out much more accurate predictions of construction site noise by adopting methods based on the Leq level of the operation or plant in any activity. The techniques employed in such predictions are not simple and the process can be complicated and time consuming particularly for larger sites.

Further, it is our experience that a large number of building sites do not really have to be considered. Under the present UK system of legislation it is felt that a considerable amount of time and money could be wasted establishing this fact for individual sites.

CONCLUSIONS

The difficulties of accurate noise prediction based on available data at various stages in the construction programme have been shown. These inaccuracies are exacerbated by the use of sound power levels as a basis for prediction in the absence of detailed information. Research has shown that more accurate noise prediction is possible based on the Leq level of the operation or plant in any activity.

In the authors opinion a good deal of effort would be saved if construction sites with a potential noise nuisance could be identified from data banks of construction operation noise levels, then only these sites would need to be the subject of further action.

REFERENCES

The advent of the Health and Safety at Work Act (1974(1)) issued in the United Kingdom has increased the impetus towards efficient noise control in industry. Often however, the issue of hearing protection and a programme of industrial audiometry is accepted as an interim measure. Persuading a workforce to use hearing protection, and increasing the awareness of noise as a hazard to hearing are tasks which are not carried out with a great deal of success in the majority of industries. The cooperation of the employees must be obtained, and it is necessary therefore to obtain a greater understanding of the problems involved. Some of the many factors affecting the success of a hearing conservation programme have been studied in industry.

METHOD

The approach included noise measurement in the areas concerned, production of experimental educational programmes, continuous monitoring of the use of hearing protection, industrial audiometry, and in particular, interviewer completed questionnaires and self-completed attitude questionnaires. This paper describes results arising from the two questionnaire methods.

Use of the interview technique provided a flexible forum for the assessment of attitudes, with the interviewer completing a questionnaire on behalf of the respondent. This exercise involved 80 men from 3 plants. The self-completed attitude questionnaires were distributed in two phases. A pilot attitude survey was undertaken involving 124 employees from 13 plants. This was followed by a major attitude census of 2109 people on 10 plants. Each participant was sent an attitude questionnaire designed according to the Likert procedure (Oppenheim 1966(2)) The questions used were derived from taped interviews, and investigated many aspects of an employee's reaction to noise, hearing conservation programmes, the use of hearing protection and deafness.

RESULTS

Factor analysis of the pilot attitude survey and associated statistical procedures, reduced the number of questions for inclusion in the main attitude census and produced 10 identifiable scales. Of special interest were those scales and interview responses describing attitudes towards the use of hearing protection. Initial results indicate that the physical bulk and the comfort rating of the protectors generally issued are the overriding factors influencing their usage. Asked to describe the main reason for not wearing hearing protection, 55% of the responses encompassed these two problems. A further 15% indicated that difficulties in using hearing protection with other forms of safety equipment was the main problem.

Difficulties experienced with communication and the hearing of indicator sounds whilst wearing hearing protection are of secondary importance compared with those given above. However 63% of users remove their hearing protection in order to communicate, 29% remove their protection in order to monitor the sound being made by their machine, and 31% take off their hearing protection in order to localise sound. The implications of these facts, and the results and inferences to be drawn from other sections of the questionnaires will be presented and discussed.

REFERENCES

(1) Department of Employment "Health and Safety at Work Act" 1974, chap.37. H.M.S.O.
INTRODUCTION

The impact of noise on the everyday life of urban populations is felt by millions of people in various forms. The solution implies general information at a level of "basic competence", to which it is necessary an efficient communication between the specialist and the layman.

NOISE INDEX (R)

Fixed on basis of the equivalent sound level considered in dB(A) and corrected acording the time of the day and the spectral characteristics of the noise, the noise index is evaluated in seven steps according to the following scale:

<table>
<thead>
<tr>
<th>R</th>
<th>0</th>
<th>Silence</th>
<th>1</th>
<th>Quiet</th>
<th>2</th>
<th>Moderately noisy</th>
<th>3</th>
<th>Noisy</th>
<th>4</th>
<th>Very noisy</th>
<th>5</th>
<th>Traumatic</th>
<th>6</th>
<th>Strongly traumatic</th>
<th>7</th>
<th>Painful</th>
</tr>
</thead>
<tbody>
<tr>
<td>Leq [dB(A)]</td>
<td>15</td>
<td>25</td>
<td>35</td>
<td>45</td>
<td>60</td>
<td>80</td>
<td>105</td>
<td>130</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

For the approximate determination of the value of R, corresponding to a given L eq, the following expression may be used:

\[ R = 11 + 8.5 \log_{10} L_{eq} \]

For practical application, the elements indicated in the following table are presented, which constitute a proposal that is, of course, only applicable to the Portuguese social conditions (urban).
ZUR MESSUNG UND BEWERTUNG VON VERKEHRSLÄRMBELÄSTIGUNGS-
REAKTIONEN—ERGEBNISSE EINER FELDUNTERSUCHUNG

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Buchta, E.

I. Einleitung
Mittels einer Felduntersuchungsreihe wird eine Analyse des psychologischen Inhalts der Verkehrslärmbelästigungsreaktion (VLBR) und eine Bestimmung der Beziehung der VLBR zum akustischen Lärmbelastungsgrad durchgeführt. Vermutet wird, daß der Inhalt der "formalen" Belästigungsklage "Ich bin durch Verkehrslärm belästigt" sich mit zunehmendem Grad der Belastung verändert.

II. Methode

III. Ergebnisse und Schlußfolgerungen
Zur Frage nach dem Inhalt der Belästigungsreaktion wurden die 38 Lärmbelästigungsreaktionsvariablen über die 274 Vpn korreliert und einer Hauptkomponentenanalyse unterzogen. Es zeigte sich, daß die VLBR in 3 Komponenten zerfällt:
1. die generalisierte sensorische Belästigungsreaktion (formale Belästigungsklagen, Wahrnehmungen v. Lärmmissionsmerkmalen)
2. die som.-emot. Reaktion (lärmbedingte Kopfschmerzen, Schlaf- und Konzentrationsstörungen, emot. Verstimmungen)
3. akustisch-physikalische Reaktionen (Kommunikationsstörungen).
The difficulties in the implementation of present criteria to control both airborne and impact noises in buildings have been originating many proposals to simplify current criteria.

**THEORETICAL BASIS**

Former criteria have been based on the concept of insulation of building elements both in the laboratory and in the field (T.L., D, R, R', N.R., etc.).

But insulation does not guarantee the desired isolation or needed "privacy" in present day construction. In later years many efforts have been directed towards single number measurements and indices.

**OUR PROPOSAL**

- **Airborne Isolation Level**
  
  Source: An adjustable power pink noise stabilized source with a range of 125 Hz - 2 kHz in the source room which generates a 90 dB(A) level in the receiving room, normalized to a source room absorption of $A = 10$ m$^2$.

  Measurement: With minimum background noise conditions read S.L.M. in dB(A) and find average sound level which should not exceed 45 ± 3 dB(A). This means that an isolation of 42 to 48 dB(A) is required for approval of the inspected building.

- **Impact Level**

  Source: A single grilon hammer rotating at 120 r.p.m. and weighing 200 g. is applied on the center of the source room floor and the receiving room level should not exceed 60 ± 3 dB(A).

**CONCLUSIONS**

Our field measurements encourage us to believe that our proposal will be useful to a future effective implementation of control of noise in buildings.

**REFERENCES**

Dans la mesure où les produits industriels liés à l'activité humaine sont sources potentielles de bruit, le développement des échanges commerciaux rend nécessaire un large consensus : sur les méthodes d'évaluation des niveaux sonores en relation avec les réactions de gêne et de nuisance des individus, les limites qui en découlent les méthodes de mesures des sources et des caractéristiques acoustiques des diverses structures matérielles que les ondes sonores rencontrent en cours de propagation.

L'obtention de ce consensus implique un important effort de normalisation. Effort qui doit précéder toute action de réglementation si l'on veut éviter des distorsions préjudiciables à l'efficacité de l'action préventive.

Pour illustrer ce propos, il suffit d'examiner les problèmes que posent encore :

1°) L'établissement de méthodes d'évaluation de la gêne pour les collectivités et des risques pour l'audition (1) (2), ceci suppose :
   a) le choix de la méthode d'évaluation des niveaux sonores, niveau pondéré ou non,
   b) le choix des corrections liées au caractère du bruit : son pur, bruit impulsif,
   c) le choix de la méthode permettant de prendre en compte le caractère fluctuant du bruit et la durée d'exposition - intégration énergétique ou autre - calcul du ECPNL - NNI - TNI - Leq - Ldn,
   d) la caractérisation du risque.

2°) L'établissement d'une méthode pour caractériser les sources (3-4-5-6) :
   a) choix d'une méthode d'évaluation des niveaux sonores,
   b) choix de la grandeur à spécifier (puissance ou pression),
   c) choix de la grandeur à spécifier pour le marquage.

L'absence d'un large consensus sur le choix des solutions fait que les autorités réglementaires ont tendance à se substituer aux Organismes de Normalisation. Même si elles acceptent d'utiliser les normes existantes, il est de leur responsabilité de :

   a) Fixer les niveaux limites acceptables pour la gêne des collectivités et préciser les plans d'occupation des sols tenant compte du zonage.
   b) Fixer les niveaux limites d'exposition pour la protection de l'audition.
   c) Fixer les niveaux limites pour les matériels et les méthodes de marquage.

Le choix des méthodes et des limites n'est pas exempt d'arrière-pensée économique, ce qui conduit actuellement à des divergences qu'il illustrent le tableau 1 et la figure 1. De telles divergences sont préjudiciables aux actions concertées, à l'échelon international, en vue de la réduction du bruit. Les acousticiens se doivent de réduire ces divergences en effectuant les recherches nécessaires pour une meilleure connaissance des effets du bruit sur l'homme et l'amélioration des méthodes de mesure.

TABLEAU 1

<table>
<thead>
<tr>
<th>ISO R1996 (x)</th>
<th>Zone A</th>
<th>Zone B</th>
<th>Zone C</th>
</tr>
</thead>
<tbody>
<tr>
<td>ALLEMAGNE (BDR)</td>
<td>Jour 35/45</td>
<td>Nuit 20/35</td>
<td>Jour 30/45</td>
</tr>
<tr>
<td>FRANCE (xx)</td>
<td>Jour 35/45</td>
<td>Nuit 20/35</td>
<td>Jour 30/45</td>
</tr>
<tr>
<td>GRANDE-BRETAGNE</td>
<td>Jour 35/45</td>
<td>Nuit 20/35</td>
<td>Jour 30/45</td>
</tr>
<tr>
<td>SUISSE</td>
<td>Jour 35/45</td>
<td>Nuit 20/35</td>
<td>Jour 30/45</td>
</tr>
</tbody>
</table>

Zone A : rurale et détente, Zone B : résidentielle
Zone C : à prédominance industrielle

(x) Proposition de valeurs enveloppe
(xx) circulaire du 26/07/1976

Für einen bestimmten Flugeugtyp, seinen definierten Betriebszustand und die atmosphärischen Standardbedingungen kann der durch den Überflug hervorgerufene äquivalente Dauerschallpegel als Funktion der Entfernung von seiner Trajektorie ausgedrückt werden. Andere Einflüsse z.B. das Abhebegewicht und ähnl. können nur eine Profiländerung der Flugtrajektorie verursachen, aber die Eigenschaften der Schallquelle und die ermittelnde Funktionsabhängigkeit werden hierdurch nicht beeinflusst.

Bei Bewegung der Quelle auf gerader Trajektorie können wir in bestimmter Entfernung von dieser Trajektorie das stetige Aufwachsen und Sinken des Schallpegels $L/\mathrm{t}$ verfolgen und dann gilt für den äquivalenten Dauerschallpegel

$$L_{\text{eq}} = 10 \log \frac{1}{t_2 - t_1} \int_{t_1}^{t_2} \frac{L/\mathrm{t}}{10} \, \mathrm{dt}$$


STUDY CONCERNING THE ANNOYANCE OF GENERAL AIRCRAFT NOISE IN SWITZERLAND

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AIRCRAFT CERTIFICATION

A first control over the annoyance caused by sports- or general aviation was obtained from an aircraft-noise limitation itself. Acoustic certification measurements are effected under special conditions of measuring and fly-over prescribed by the Swiss Federal Department of Transport, Communication and Energy. The tolerable level limits were defined after a period of essay during which several types of aircraft were measured and classified according to different parameters. Finally, the regulation of tolerable noise was fixed as a function of the max. take-off-weight (acc. to certification of civil aircrafts), i.e.

1-engine-aircrafts: \( L_A = (26.35 \times \log \text{max. weight}) - 3 \)
2-engine-aircrafts: \( L_A = (26.35 \times \log \text{max. weight}) - 6.1 \)

When a tested aircraft proves to be too loud, a technical modification is exacted. If this is technically impossible or the value of the aircraft does not justify an expensive transformation, a special licence may be granted. The noise of some aircrafts has been studied separately as a function of the propellers used or/and the power of their engines, and the use of silencers has been tested as well.

THE ANNOYANCE IN THE VICINITY OF AIRPORTS

Besides the certification relating above all to the noise emission, we tried to value the noise situation in the vicinity of airports not only for the present time but also for the future, referring to the probable evolution of the ten years to come. Noise curves should also be taken into consideration for the siting of new airports. We think a formula containing Swiss standards of noise judgement would be useful for a comparison between aircraft- and other noise in industry and traffic on roads and rail; for example:

\[
\text{Annoyance Level} = AL = \frac{L_{SA}}{5A} + 10 \log N - K
\]

where \( L_{SA} = \text{max. mean noise level} \quad N = \text{number of movements} \quad K = \text{to be discussed and fixed.} \)

AL corresponds to one of the limit sound values admitted by Swiss Noise Reglementation. According to the value of K, either 10,1 % = a sound level of 1-6 noise peaks p.h., or 0% = mean level of 7-60 peaks p.h. will be used.

Other criteria and formula have been studied and are now being compared with each other as a function of different parameters.
It has long been considered desirable to define acceptable noise levels in crew compartments of aircraft in order to guard against disruption of communications, auditory fatigue and temporary or permanent damage to the hearing of aircrew. A method, developed at RAE, of defining such levels was based on avoiding overload at the ear, retaining adequate intelligibility and preventing permanent noise-induced hearing loss. For this last criterion, CHABA Damage Risk contours were used. The limits for different kinds of personal equipment were given in the form of octave spectra.

These limits have been useful in setting standards, and are believed to be reasonably accurate; but several factors suggest that the methods should now be revised.

First, it has been found difficult in practice to apply the derived limits to noise spectra of other shapes. Secondly, it is usual now (at least in Europe) to rate the risk of noise-induced permanent threshold shift according to the total A-weighted acoustic energy at the ear, rather than to Damage Risk contours. And thirdly, measurements of the noise dose and recordings taken at the ears of aircrew in operational flight recently have shown that insufficient account has hitherto been taken of the contribution of communications to the total noise dose. It has been found that the Equivalent Continuous Noise Level during a flight may be as much as 10 dB higher than the ECNL due to ambient noise alone, without communications.

This paper suggests possible methods by which new standards can be derived, taking account of the contribution of communications to the noise dose. The considerable difficulties in deriving any maximum permissible noise levels due particularly to variability in the noise-attenuating properties of headgear are increased when the variability of preferred listening levels for communications is considered. Hence, it is proposed to define a method of judging the acceptability of predicted noise spectra or of spectra measured in prototype aircraft, instead of laying down maximum spectra, although maximum permissible levels for some simple spectrum shapes will be derived as a guide.

In aircraft already in service, however, the assessment can best be made on the basis of measurements taken at the ears of aircrew in operational flight and recording and analysis methods are described.

Although the methods have been derived in relation to the crews of military aircraft, they may equally be applied to any situation in which people wearing hearing protectors, with or without communications, are exposed to acoustic noise.
SUBJECTIVE ASSESSMENT OF BLADE SLAP

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INTRODUCTION

A detailed examination has been made of blade slap to determine the most suitable method for quantifying the impulsive nature and a series of psycho-acoustic tests have been conducted in order to assess the impact of such impulsive sounds (1).

The overall aim was to develop a simple method of quantifying blade slap and to establish a "correction factor" which could be applied to the conventional dBA and PNL units used to rate helicopter noise.

MEASUREMENT OF BLADE SLAP

Blade slap can be successfully quantified using the following crest factor

'Peak' measured in 250Hz Octave Band — dBA RMS SLOW

The band pass filter is necessary to provide adequate rejection of low frequency main rotor noise and also the tail rotor noise.

An analogue peak detector has been developed — this has a rise time of 200msec. and a decay rate of 50msec. This can be used for both hover and flyover analysis.

A corresponding digital approach is under development.

SUBJECTIVE STUDY

Fifteen "real" helicopter recordings were ranked in order of their blade slap severity by a small jury.

A simulated blade slap pulse was added at different levels to a non-impulsive helicopter recording to provide a range of "simulated" blade slap conditions. These were played to nineteen subjects who were asked to adjust the level such that it was "equally intruding or annoying" to a non-impulsive helicopter noise recording used as the "reference sound".

The jury were also asked to adjust the level of recording of a helicopter with severe blade slap (a Chinook) so that it was "equally intruding or annoying" to that of a non-impulsive helicopter (Wessex) recording.

RESULTS

The tests confirmed that a subjective correction was necessary — the results are illustrated diagrammatically in figure (1) as a function of the crest factor. A correction term of the form indicated is proposed — being 0 and 6dBA for respective crest factors of 11dB and 20dB.

CONCLUDING REMARKS

The dBA and PNL units do not represent the annoyance caused by a helicopter when generating blade slap and a correction is necessary.

REFERENCES

EXPERIENCE WITH A NOISE MONITORING SYSTEM AT A MAJOR UNITED STATES AIRPORT

Starr, E.A. Comp. Syst. Div., BBN, Cambridge, MA
Callaghan, T.P. Massachusetts Port Authority
Steele, D.W. Comp. Syst. Div., BBN, Cambridge, MA
Van Dean, R.F. Massachusetts Port Authority USA

A computerized automatic noise monitoring system was installed at Logan International Airport in Boston, MA in the late fall of 1974. The monitoring system has 12 remote listening sites connected to a central digital computer by ordinary telephone lines. The computer continuously receives data from all sites. The system prints out hourly and daily averages for each site, as well as individual flyover information when desired. A block diagram of the system is shown in Fig. 1. In the first 10 months of 24-hours-per-day operation at 12 sites (including winter operation in New England), data were collected 93.5% of the time. Downtime included all difficulties, even faulty phone lines. This has improved since the starting period.

Little has been reported on the use of a monitoring system as an instrument of management information and policy. Unless the technical data is converted into useful management information with a noise abatement strategy, the potential return from the system is not realized. Massport has used the data from the system in the following ways.

Chief Pilots of airlines have been contacted when their flights exceed levels that are met by the majority. The Chief Pilots have been cooperative and have reviewed the circumstances of the flights. As a result one major airline has modified its take-off procedures across the country.

Complaints from ground runway operations have been nearly eliminated. Some of the microphones are located so as to monitor nighttime noise near residents. Whenever excessive noise is registered, Massport's Night Noise Patrol Investigates. In another case, the database has allowed Massport to document an application for $2M in federal funds for land use alteration.

Massport established a noise abatement committee consisting of representatives of the community, the airline industry, and the FAA. This group provided a basis for discussion of the problems and feedback on the conditions.

The existence of the monitoring system has been, and continues to be, of assistance in the analysis of optimum runway use in conjunction with the FAA. The data allows the determination of how a change of flight pattern alters the long term noise pattern. It is projected that this use of the system will be expanded in the future.

In summary, the noise monitoring system at Logan International Airport is being used as an implement of noise abatement policy and is integrated into the operations of the airport. Its presence is helping to reduce the environmental impact of the airport.

![Fig. 1. Simple Diagram of Noise Monitoring System](image)
IMPLEMENTATION OF THE LAND COMPENSATION ACT IN RESPECT OF AIRPORTS

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House, M.E. University of Southampton, U.K.
Ludlow, J.E.

INTRODUCTION

New public works which cause property depreciation due to levels of noise from their usage are subject to claims under the United Kingdom's Land Compensation Act (1). The new runway built by the British Airports Authority at Edinburgh Airport (diagram 1) will be the first major application of the statutory provisions for civil aircraft operation. This paper describes objective noise surveys undertaken to provide background material to assist assessment of such official claims under the Act which may arise.

SURVEY

Before completion of the new runway, environmental noise was surveyed at over 50 typical locations covering areas which would become exposed to aircraft noise, and also control locations affected by the old runways. At 19 locations an 18 hour assessment was made; the rest were limited to day time samplings. One year later, the influence of peak summer schedule operations noise was measured at 50 points covering all nominal track and side-line situations for aircraft types and all runway modes. Salient environmental samples were repeated with and without aircraft events. Data on aircraft range and elevation angles were determined (by a camera method to give altitudes etc.), also meteorological data.

ANALYSIS

Environmental data was on-line or laboratory processed from tapes to provide full probability distributions of time for which levels in dB(A) were exceeded. Tables of the main $L_{eq}$, $L_{10}$, $L_{50}$ and $L_{90}$ indices were extracted. For aircraft noise, on-line time histories in dB(A) were produced, and later in the laboratory, corrections were applied, based on a range of the measured spectra, to produce levels in dB(A), dB(D) PNdB and SENEL(A) at the time of peak noise and for standard ISA + 10°C, 70% RH atmospheric conditions. The data was additionally corrected to 152m (500ft) minimal slant range reference conditions. Each major aircraft type operating from the airport was individually assessed.

RESULTS

Plots of aircraft noise versus range were produced for each case (diagram 2) and also of reference levels versus elevation angle (diagram 3). Interim work indicates little evidence of the usual over-ground attenuation below 15° elevation, but possible airframe shielding for some types. For practicable ranges to the community, jet aircraft noise in PNdB fell by an average 9% per range doubling as opposed to the currently used two part rule according to elevation angle. Examples are given (diagram 4) of the impact of the newly exposed aircraft noise in relation to the previous environment probability diagrams. The study did not include socio-acoustic surveys.

REFERENCE

(1) H.M.S.O. London 'Land Compensation Act 1973' R2 ch.26
INTRODUCTION

The American National Standards Institute (ANSI) Committee on Mechanical Shock and Vibration 5-2, sponsored an eleven member Working Group on Atmospheric Blast Effects. Their third draft standard for single point explosions in air was distributed for review in 1976. Future supplementary standards are expected for more complex explosions, of multiple sources and underground bursts.

PURPOSE

This standard gives quantitative definitions for explosion parameters, effects of atmospheric propagation, and typical responses to waves from single point explosions in air. The phenomenology is less well defined for many complex but practical explosions, although in many cases their effects may be normalized and an equivalent point explosion used for evaluation. This standard should help in design of adequate and safe systems, prediction of troublesome conditions, and evaluation of damages caused by intentional or accidental explosions. A bibliography of fifty references is provided as a source for more details.

METHODOLOGY

The recommended evaluation follows a chronological order of source description, atmospheric propagation, and effects on both structural and biological targets. Source description begins with a standard explosion wave, adopted from the U.S. Air Force Weapons Laboratory numerical model for a 1-kt nuclear explosion in sea level air. Sach's scaling laws give pressures, times, and distances for other yields and ambient conditions. Height-of-burst parameters allow estimation of an enhanced equivalent yield for predictions of surface bursts with hemispherical wave emission or elevated bursts which form fused waves from incident and ground reflected waves. Airblast yield equivalents, compared to TNT, are provided for common chemical explosives. A power law for overpressures versus distance is used at medium and long ranges in non-refracting atmospheric conditions.

In the real atmosphere, temperature and wind stratums may significantly distort airblast waves, so observed overpressures may be enhanced or diminished from standard values. Empiricisms show the range of amplifications or excess attenuations that occur under a variety of refractive conditions.

The response of simple structures to both diffraction and drag loading is described for strong shocks, but details have been left to specialized texts. An empirical relation is shown for expected repair cost fraction of replacement costs for a typical residence exposed to various overpressures. Effects on vehicles are also summarized.

Effects of low overpressures (≤7 kPa) have been addressed in detail, since this subject has not been widely published. Window breakage has been used as the primary response, although cosmetic damage and annoyance may occur without associated glass damage. Empiricisms, in power law and probability form, from both test and accidental explosions, have been used to estimate window damage, numbers and costs from imposed overpressure and exposed population variables. These relations have also been transformed to give explosion yield limits in terms of distance and population for no damage in adverse weather. For larger yields, distance must be increased or a weather restriction should be considered.

Biological responses in humans, other animals including fish, and vegetation have been summarized to include direct overpressure, displacement, and missile hazards.
In 1973 in Vienna together with a census an inquiry, how people assess the environmental quality of their neighbourhood, was performed. It showed that 51,6% felt "considerably annoyed" by noise, mainly traffic noise (53,3%).

At about the same time, 1973 and 1974, traffic noise measurements were carried out at 90 different sites in the town. The results of the objective noise level measurements were correlated with the subjective reaction of people living at these different places. Altogether the answers of 2624 persons living at 61 of the different sites were evaluated (49,2% of these people had felt considerably annoyed during day-time).

From the sound level recordings the following units, used to describe noise situations, were calculated:

\[
L_{eq3}, L_{eq4}, \frac{L_{eq3} + L_1}{2}, L_1, L_{10}, L_{50}, L_{10}, L_{90}, L_{NP}, TNI
\]

These indices also show a more or less high correlation mutually.

All of them were correlated, in 5 dB-groups, with the relevant subjective answers. A high correlation was found with nearly all the indices, the highest of 0,99 for \(L_{10}\) and \(\frac{L_{eq3} + L_1}{2}\), and 0,98 for \(L_{eq3}, L_{eq4}\) and \(L_{50}\).

Limits for urban traffic noise in front of dwelling-houses can be derived from the correlation line:

<table>
<thead>
<tr>
<th>percentage of people considerably annoyed in their home</th>
<th>10</th>
<th>25</th>
<th>50</th>
<th>100</th>
</tr>
</thead>
<tbody>
<tr>
<td>(L_{eq3}) in the street dB(A)</td>
<td>56</td>
<td>60</td>
<td>67</td>
<td>81</td>
</tr>
</tbody>
</table>
INTRODUCTION - identification and description of noise sources on a definite environmental area can be realized in form of acoustic maps which are divided into SYNTHETIC and ANALYTIC. The Polish method is to work out analytic acoustic maps of towns or town agglomerations and separated areas. Maps are made separately for 4 noise groups: traffic, railway, aircraft and industrial. Maps can refer to the existing state or predicted state for definite future. The preparation method of traffic noise map of a town, given in the paper, is an example of the Polish method.

THEORETICAL BASES - theoretically, sound level $L_k$ describing environmental acoustic climate can be qualified from the formula:

$$L_k = 10 \log \sum_{i=1}^{n} 10^{0.1 L_i}, \text{dB/A}$$

where: $L_i$ - succeeding sound level components /respective noise groups, existing in certain environmental area - the equivalent value in dB/A/. In practice, there are evaluated repeatedly component parameters of environmental acoustic climate of chosen noise groups. Representative values of traffic noise are: $L_{10}, L_{50}$ and $L_{eq} \text{dB/A}$.

EXPERIMENTAL TECHNIQUES - on base of 24-hour measurements in main streets of a town - days of a week and rush hours of day and night were established - the hours when percentage of heavy vehicles is greatest. These chosen hours are for those noise measurements.

- Measuring positions data is selected as needed; they are in the sites characteristic for the transportation town scheme;
- in every position 10 min time noise interval is recorded during maximum traffic density. Microphones are always placed 1 m from kerbs of roads at the height of 1.2 m;
- during recordings a number of various types of passing vehicles and traffic density as a whole are calculated and a detailed description of measuring sections must be done;
- all recordings are being realized during 4 summer months, no longer than 2 years in succession;
- noise recordings in measuring points, situated near main points, have to be done at the same time. Analysis of measurements data is carried on by a minicomputer, adjusted to obtain values from $L_{10}$ to $L_{eq}$ and $L_{eq} \text{dB/A}$. The $L_{10}$ and $L_{50}$ values are marked on 2 plans of a town and are graphically presented by lines of different width, proportionally to sound level values, for every class, different from others by 5 dB/A.

RESULTS AND CONCLUSIONS - In 1966-76 acoustic plans were drawn for some Polish towns /for Warsaw - for two time periods/. The maps have found approval of town-planners and other specialists who find maps useful and helpful in environmental noise control.

REFERENCES

/1/ J. Sadowski; Report about the research state and output of Poland in the domain of environmental protection against noise, Building Research Institute, Warsaw, 1976.
INTRODUCTION

Both road and railway noises are common sources of noise in our daily life. However, the fluctuation of noise level from these sources show quite different patterns. Road noise shows random and approximately gaussian pattern. Railway noise shows intermittent pattern. The unified system of assessment in these sources of noise is hardly established to express the nuisance value.[1]

This study was undertaken to find out the basic problems in the unified system of noise rating by comparing subjective response to these noises.

METHOD OF INVESTIGATION

The survey was carried out at the residential areas in Yokohama city and its environs from Oct. to Dec., 1975. The source of the noise was either road or railway at each site. Five or six housewives were interviewed at each site to find out the annoyance level before the acoustic measurement of the noise. The questionnaire was designed to take the feeling about the general outdoor noise not specifying the source of the noise. A precision sound level meter was mounted on a house garden facing the major noise source at each site. Measurement were made for ten minutes in every four hours of one complete 24 hr. period not including rainy or windy days. The interview data of 356 housewives and the results of acoustic measurements at 65 sites were thus collected.[2]

The acoustic data were analyzed through an A-weighting network and divided into five dBA classes. The noise level based on cumulative distribution function (Ln) and the equivalent noise level on the energy basis (Qg) were calculated at each site. These two indices provide a basis in unifying the assessment of noise.[3]

CONCLUSION

The data in the figures would seem to justify the following conclusion.

Both sources of noise show approximately same L10 and Q6 value. This means that L10 and Q6 value could be used as the indices for nuisance rating of outdoor noise. However looking into the detail, Ln & Q value of both sources of noise show approximately same when "n" & "g" are rather great at stronger annoyance level and when "n" & "g" are rather small at weaker annoyance level. The adjustment of these aspects will be the future study point.

REFERENCES

[2] A. Tamura & S. Gotoh, Reports of the 1976 Autumn meeting, the acoustical society of Japan, 449-450
INTRODUCTION

A survey of nuisance from road traffic noise was published by Building Research Station in 1968 (1) and the results developed (2) and incorporated in legislation (3). The work aimed at control of urban motorway noise nuisance. In 1972 a larger scale enquiry was launched to study noise nuisance from all types of urban traffic. The results have now been published (4,5) and are summarised in the present paper.

THE SURVEY

The sample consisted of 3000 London residents at sites covering both free and non-free traffic flows from 300 to 5000 vph. Façade measured noise levels ranged from 60-80dB(A)L10, with road widths from 6-25m and dwellings from high rise to single storey. Noise levels were measured continuously at each site for 48 hours. Traffic was counted and classified from 08.00-20.00 hours each day. Interviews covered noise dissatisfaction, disturbed activity and sleep, environmental conditions, and individual differences in sensitivity to noise.

RESULTS

For free flow traffic the survey confirmed the 1968 results with increased accuracy and reliability, yielding a regression equation for L10 from 06.00-24.00 hours.

\[ \text{Dissatisfaction (7pt scale)} = 0.147 L10 - 5.67 \quad (r=0.85; p<.001; n=24) \]

Where traffic is congested or disrupted no existing measure yielded useful results, the best correlation \( (r=0.43; p<.05) \), that for Lp,, being significant but too low for practical purposes. Useful results were obtained by employing the log percentage of vehicles exceeding 1525Kg gross weight as a parameter to obtain the equation

\[ \text{Dissatisfaction} = 4.8 \log \%HV - 0.2 \quad (r=0.74; p<.001; n=29) \]

Nuisance may be predicted for all types of traffic without prior classification, at a comparable level of accuracy, by combining measures of traffic composition and noise level, measured as L10, over the period 08.00-20.00. This gives the equation

\[ \text{Dissatisfaction} = 0.078 L10 + 3.34 \log \%HV - 4.5 \quad (r=0.71; p<.001; n=53) \]

Sleep disturbance was correlated with noise level measured as L10 from 22.00-06.00 hours, taking into account residents actions aimed at reducing noise by closing the bedroom windows in hot weather, estimated to reduce noise exposure by some 10dB(A). The result gives useful predictions of the percentage population undisturbed.

\[ \text{Percent sleeping undisturbed} = -1.25 L10 + 128.6 \quad (r=0.88; p<.001; n=26) \]

Detailed analysis of individual responses to all questions showed that of the 40% of total variance accounted for, only a quarter is attributable to physical variables, most of the remainder being due to differences in sensitivity, followed by variations in environmental conditions and demographic factors.

REFERENCES

BELÄSTIGUNG DURCH AUTOBAHNLÄRM

Buchta, E. Institut für Hygiene der Universität Düsseldorf
Kastka, J. Düsseldorf, Germany

EINLEITUNG

Die Belästigungswirkung des Straßenverkehrslärms auf Bewohner einer Großstadt konnte mittels einer Befragungsmethode (Fragebogen mit 38 Lärmvariablen und neu entwickelten Belästigungsskalen) erfaßt und in Beziehung zu den physikalischen Daten der Straßenverkehrsgeräusche gesetzt werden. Es konnte festgestellt werden, daß der zeitliche energetische Mittelwert sich gut zur Beschreibung der Belästigungsreaktion eignet.

METHODE

Zur Erfassung der Geräuschpegel von Autobahn- und Anliegerverkehr wurden kontinuierliche 24-Stundenmessungen gleichzeitig in mehreren Abständen von Autobahnen an Wohnsiedlungen mit geringem Anliegerverkehr durchgeführt. Ausgewertet wurden die Mittelungspegel für die Tageszeit 6:00-22:00 und für die Nachtzeit 22:00-6:00. In dieser Pilotenuntersuchung wurden an einer Autobahnbebauung mehrere unterschiedlich stark belastete Anlieger-Gruppen in deren Wohnung über ein ca. 1-stündiges Interview mit 34 Lärmvariablen befragt. Die Befragungen und Geräuschmessungen erfassen einen Geräuschbelastungsbereich von 50 dB(A) bis 75 dB(A) Außenpegel tagsüber.

ERGEBNISSE


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/1/ Buchta, E.; Kastka, J. "Relation between the annoyance of traffic noise and physical noise level data". Internoise 77, Zürich.
/2/ Kastka, J.; Buchta, E. 9. ICA Madrid 1977

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1. Environmental acoustics

B. DESIGN AND PLANNING
INTRODUCTION

The search for the best unit for environmental noise appears to be concluding with adoption, due to administrative pressure, and despair at agreeing an alternative, of the equivalent continuous "A" weighted sound level Leq. The adoption of single criteria appropriate for planning purposes is also foreseen. Practical experience with planning and architectural design for noisy surroundings suggests that adoption of this approach is unwise and should be adopted only with extreme caution. Some practical cases which illustrate the need for care are described.

DISCUSSION

The administrative value of Leq apparently is to measure all sounds with the same unit. Background noise, which has sounds wanted and those unwanted (noise), can be summated. A desirable goal can be set for environmental noise, e.g. 55 Leq, then applied for instance to train noise alone, traffic noise alone, aircraft noise alone, dogs barking alone (or in practical cases together). Persons use sound to identify processes so that separate evaluations are made of those sounds which produce the stated energy equivalent summated value. The summated value has therefore no significance to humans in describing an acoustic environment.

In an evaluation of complaints from residents near a diesel test bed facility, the Leq values from aircraft and trains were far in excess of that from the occasional test engines. However, the acceptance of these noise producers as inevitable, focused complaints on the quieter but more annoying noise.

The acoustic design of dwellings necessitates knowledge of the frequency composition of external noise and especially knowledge of the peak amplitude of external sound. In the design of dwellings in urban areas near principal traffic routes, the dominance of heavy lorry noise in determining community response places great importance on the amplitude of low frequency noise and the sound insulation at low frequencies of glazing and walls.

The use of 68 dB(A) L10% 18 hour in U.K. for separating compensatable from non-compensatable dwellings for traffic noise, has led to a lay impression that 67 is subjectively better than 69, and also to a demonic official determination to establish the precise location of the 68 contour from a road. Similarly in airport evaluation the 35 NNI contour has been used to separate clinically those to be considered and those not with respect to disturbance due to aircraft noise.

The scientific work which led to this situation clarified the real community significance of 1 - 2 changes in dB and the gigantic fluctuation in individual responses. However, these cautions soon become overlooked as will those about Leq if allowed free promulgation without severe constraints now.

SUMMARY

In areas of continuous excessive noise, it is irrelevant what unit is used. Leq, L10, L50, L90, L99, are all adequate. The area where evaluation is needed is where isolated events occur, i.e. the usual situation. Real environments are rarely continuous. Adequate care should be taken to evaluate as much of those parameters used by humans to experience sound, especially how loud, how much louder than ambient, and sound quality, at least octave bands. The real finding of acoustic research in this field may be that individual differences in response are so significant that the setting of criteria levels is basically invalid in any unit. The planning logic could be to severely limit change in acoustic situations, such that those wishing for peace can migrate to it and those who find "streets sounds nice" can live near the freeway. Planning with a quasi-acceptable level for all, may produce the quiet people depressed by noise and the noisy ones upset by the lack of external sounds!
Il est unanimement admis que tant l'architecture que l'acoustique sont en même temps arts et techniques. On peut d'autant plus faire la même remarque sur l'acoustique architecturale.

Parmi les qualités physiques des constructions (dimensions, air, chaleur, lumière) un très important rôle est joué par l'ambiance sonore, qui peut contribuer non seulement à la bonne et comode utilisation des bâtiments mais aussi à la personnalité et aux rythmes des espaces construits.

Sous ce point de vue le travail analyse divers types d'espaces clos comme annexes des salles de spectacles ou d'audition, halls, restaurants, salons, musées, expositions, bâtiments d'enseignement ou de bureaux.

En partant des deux caractéristiques qui peuvent être influencées, notamment le degré d'insonorisation et (éventuellement) le niveau du bruit de fond, le schéma ci-contre présente la manière dont en les variant on peut changer l'effet psychologique et donc la personnalité des espaces.

Le niveau du bruit de fond est difficilement modifiable dans le cadre du même bâtiment. En revanche on peut réaliser un dosage des surfaces phonoabsorbantes réalisé avec des formes et matériaux divers, et qui peut contribuer à la composition architecturale par:
- la variation des deux caractéristiques nommées en les gradant dans une suite de chambres,
- le contraste entre deux espaces voisins,
- l'accentuation ou la compensation des dimensions d'un espace. (une chambre sonore paraît plus grande en même temps qu'une chambre sourde paraît plus petite).
- la séparation acoustique de deux espaces en communication.
- la réalisation de zones différenciées ou de secteurs intimes et calmes dans des espaces ouverts si ceux-ci sont bruyants.

La principale conclusion du travail est que pour obtenir un vrai confort dans les bâtiments, les qualités acoustiques des espaces doivent être prises en compte dès le début de l'étude.

Références


This paper is concerned with an experiment to assess the effect of speech and/or ventilation system noise upon efficiency. The investigation was carried out under controlled laboratory conditions, whereby the subjects were asked to complete two tasks, one of which was an adding test and the other a checking exercise. During the experiment, which lasted one hour, the background noise conditions were varied through Quiet, Speech, Fan Noise, and Speech and Fan Noise together.

Detailed consideration of previous research work provided three points of hypothesis. These were as follows:

H.1 that speech of normal conversational level, 60 dB (A), creates distraction amongst subjects undertaking clerical tasks involving mental arithmetic or checking and gives rise to a feeling of discomfort, irritation or anxiety.

H.2 that this distraction creates a drop in efficiency by lowering output.

H.3 that ventilation system fan noise of a sound pressure level comparable to that of normal speech does not give rise to any of the detrimental effects referred to above in connection with speech.

The nature of human response patterns resulting from ambient noise can be classified under three separate categories namely, physiological, irritation and efficiency. Since this study has primarily been concerned with low level (60 dB 'A'), background noise, physiological effects such as metabolism, skin response, rate of breathing and similar indicators have not been investigated. Justification for omitting this area of investigation has been based on the findings of other studies whose results have shown that harmful physiological effects do not occur below 100 dB(A).

There is considerably less evidence of the effects of low intensity noise upon irritation or efficiency, and even less indication of how well, if at all, the level of irritation correlates with a corresponding level of efficiency under these conditions. Thus, in a study of the effect of low level noise upon human behaviour, it is necessary to consider both irritation and efficiency. However, efficiency has been given greater prominence in this study since it is from several points of view the more satisfactory. It is directly related to aspects of practical, and hence economic, importance such as clerical output or error rate, whereas irritation is related only indirectly.

Both performance and irritation were measured and assessed independently and their inter-correlations calculated.

The following general conclusions have been obtained from the research:

(a) When carrying out the adding task, the performance of subjects was found to drop by a highly significant (p = <.005) amount when tested with Speech (60 dB'A') in the background compared to Quiet (35 dB'A') conditions. No such effect, however, was found in respect of the checking task.

(b) The attitude responses of subjects indicated that whilst undertaking intellectual tasks, the presence of Speech in the background caused moderate discomfort compared to virtually no discomfort under Quiet conditions. There is, however, little evidence to suggest that the effects of Speech give rise to any more severe feelings of distress, such as anxiety or irritation.

(c) The performance of subjects on the adding task showed no significant difference in output between the Fan Noise condition and the Quiet condition. However, a significant difference was found between the Fan Noise and Quiet conditions in respect of the checking task, suggesting that the Fan Noise had provided a stimulation effect.
INTRODUCTION

The Royal Opera House, Covent Garden, London, U.K. is a major world centre for the presentation of classical opera and ballet. It is the permanent home of the Royal Ballet Company.

THE BRIEF

Some years ago it was realised that the stage surface was deteriorating to an extent that would eventually endanger the dancers and compromise artistic performance standards of ballet. As well as refurbishing the stage was required to achieve a surface more suitable for dance in that it was to have substantially constant characteristics over the full dancing area. The firm of Arup Associates, architects and engineers, (Partner, Derek Sugden) were appointed together with Grootenhuis Allaway Associates to consider the dynamic aspects of the design; both the author and Professor Peter Grootenhuis were deeply involved in the project.

THE PROBLEM

The old stage was constructed in 5 "lifts" each of which could be raised/lowered independently; each lift is about 12m x 2.4m and is supported on a pair of steel lattice arch beams spanning the 12m. Between the beams, to provide the stage surface, were heavy oak boards whose surface was splintering and uneven. Above each lattice beam the stage stiffness was very high and caused difficulties for the dancers especially in the more "athletic" ballets; between the beams the oak boards provided a lower, though still high, stiffness.

THE SOLUTION

After many tests, discussions with dancers, etc., the old surface was planed and new panels laid on top. Each panel was about 2.4m x 1.2m and comprised a selected plywood bonded to steel sheet and supported by strips of rubber-like material from the old surface; magnetic catches were used to retain the panels in position and each panel was independently replaceable. The joints between "lifts" were covered by similar, but narrow, panels spanning from lift to lift.

THE EXPERIENCE

R.O.H. is a very busy house and the stage surface has heavy wear from opera sets, etc. In the course of time damage to several panels has occurred and proposals for a new surface which would be laid only for ballet are among those under active consideration.
THE DESIGN OF AUDITORIA FOR THE OPTIMISATION OF SPEECH INTELLIGIBILITY

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INTRODUCTION

Previous work has indicated a gap in auditoria design techniques for the optimisation of speech intelligibility. The present work involves a correlation between subjective speech intelligibility and objective 'signal-to-noise' measurements in existing auditoria (1).

THE SUBJECTIVE TEST

Speech recorded anechoically is played to subjects in empty auditoria, using the Fairbanks' 'Rhyme Test' (2), testing the recognition of initial consonants of words embedded in nonsense phrases. Untrained listeners, with audiometrically good hearing, carry out every condition. The analysis determines the significance of auditorium, position, level and subject, while order is accounted for by experimental design.

THE OBJECTIVE TEST

The impulse response to electric spark (3) is recorded in auditoria and analyzed by on-line computer. Attenuation over seats and background noise are measured. From this data is derived Lochner and Burgers' Signal-to-Noise Ratio' (4), applied to whole octave bands and accounting for equivalent background noise.

CONCLUSIONS

The results signify the relative importance to design of 'signal-to-noise', reverberation time and background noise. It may eventually be possible for the acoustician to advise the architect on the location of specific surfaces for the optimisation of speech intelligibility.

REFERENCES

(4) J.P.A. Lochner and J.F. Burger, J. Sound Vib. (1964) 1, 426
The use of computerized 3-dimensional sound ray tracing in mathematical model of rooms opens new possibilities for investigating the sound energy distribution in rooms. This computer technique enables us to study, simultaneously, the space, time and directional distribution of direct and reflected sound energy on any surface in rooms.

Our laboratory has since 1967 been developing such a computer program. The computational procedure and data presentation are described in (1) (2) and (3). The program has been greatly modified during the last few years:

- An absorption factor is assigned to each room surface. The simulation of real materials in halls are thus possible.
- Sound ray impact data can be registered on any room surface (we call them "microphone surfaces"). Impact points, time delay, direction of incidence and energy are stored when a ray hits such a surface.
- We use freely hanging spheres in the room as "microphones". Impact data are stored when a ray hits the spheres.
- The impact data can be plotted on a plan view of the "microphone surface". This plotting is done using a stochastic procedure which gives ray impacts with little energy a low possibility of being plotted.
- Time-energy echograms are drawn for each "microphone". Deutlichkeit, Schwerpunktzeit or other parameters are computed, if needed.
- The sound ray reflections at the room surfaces are normally computed using geometrical acoustics. But it is possible to use a stochastic procedure to simulate diffusing surfaces.

This program has proved very valuable in our work as acoustical consultants for all types of halls:

- Multi-purpose hall in "Ibsen House" (700 seats, finished 1973).
- "Hjertnes" multi-purpose hall (700 seats, finished 1975).
- "Grieghallen" concert hall (1600 seats, will be finished 1977).

Our program is available for use on a certain basis by other consulting firms of organizations who want to investigate sound energy distribution in rooms.

(1) A. Krokstad, S. Strøm, S. Sørsdal: "Calculating the acoustical room response by the use of a ray tracing technique" J. Sound Vib. 8 (1968), nr 1, pp 118-125.
INTRODUCTION

It is said that early reflected sound has a very important role to get good intimacy. In order to know how the live end of an auditorium with rigid surface, as shown in fig.2, composes the sound field at the steady state, it is calculated with the ray-theory and geometrical acoustics.

THEORETICAL BASES

Reflection of a rigid plate, which is not extremely uneven, is calculated by the sum of the reflections of the \( n \times m \) divisions as follows (see fig.1):

\[
\Phi(p) = \sum_{i,j} \frac{k_i(k_j)}{n_i n_j} \sum_{x,y} \frac{\cos(\alpha_i \theta) \cos(\beta_j \theta)}{\sqrt{\frac{1}{U_x} + \frac{1}{U_y}}}
\]

where

\[
U_x = \frac{h}{2} \left( \frac{x-\frac{1}{2}}{S} + \frac{x-\frac{1}{2}}{P} \right)
U_y = \frac{h}{2} \left( \frac{y-\frac{1}{2}}{S} + \frac{y-\frac{1}{2}}{P} \right)
\]

The first reflection is calculated by eq. (1). Propagated sound from a point source to a receiving point, reflected by plate 2 and 3, for instance, is estimated as the second reflection by treating one plate (plate 2) infinite and another (plate 3) finite whose point source is the image of the source in regard to plate 2. When the third reflection (e.g. the source \( \rightarrow \) plate 1 \( \rightarrow \) 2 \( \rightarrow \) 3 \( \rightarrow \) receiving point) is obtained, its reflection is calculated by treating two other plates infinite.

EXPERIMENTAL TECHNIQUE

Pure tone is generated from the end of a cylindrical pipe to get an omnidirectional point source. Sound distribution is observed along the line \( A \rightarrow \) and frequency characteristics is observed at the point 1, as shown in fig.2.

RESULT AND CONCLUSIONS

On the sound distribution, measured curve is compared in fig.3 with the sum of the direct sound, first, second and third reflections. On the frequency characteristics, it is compared with measured curve in fig.4. The sum up to the third reflection seems to match well with the measured values.
Subjective preference tests of synthetic sound fields were performed to study the influence of the degree of interaural cross correlation (ICC) [1] and the pattern of the amplitude decay of the echoes (ADE) on the judgements of listeners.

As source signals two pieces of music were used: Gibbons, Royal Pavane (Motif A) and Arnold, Sinfonietta (Motif B) [2]. The synthetic sound fields consisted of the direct sound and four reflections. The direct sound was radiated from a loudspeaker in front of the listener with the horizontal angle $\xi = 0^\circ$ and the elevation angle $\eta = 9^\circ$. The reflections were presented with two different systems:

**System a:** $\xi_{1,2} = 40^\circ$; $\xi_{3,4} = 140^\circ$; $\eta_{1,2,3,4} = 27^\circ$

**System b:** $\xi_{1,2} = 0^\circ$; $\xi_{3,4} = 100^\circ$; $\eta_{1,3,4} = 27^\circ$, $\eta_2 = 45^\circ$

For each system two different patterns of amplitudes were investigated:

<table>
<thead>
<tr>
<th>time of arrival</th>
<th>0</th>
<th>32</th>
<th>52</th>
<th>68</th>
<th>80</th>
<th>ms</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADE I</td>
<td>0</td>
<td>-4.2</td>
<td>-6.2</td>
<td>-7.5</td>
<td>-8.0</td>
<td>dB</td>
</tr>
<tr>
<td>ADE II</td>
<td>0</td>
<td>-6.2</td>
<td>-4.2</td>
<td>-7.5</td>
<td>-8.0</td>
<td>dB</td>
</tr>
</tbody>
</table>

Accordingly, for each source signal four sound fields were presented to 13 experienced subjects for preference judgements.

The ICC for systems a and b are 0.53 ± 0.01 and 0.81 ± 0.07 resp. for both source signals and both ADE.

The main results are:

1. For a fixed pattern of amplitudes (ADE) the system a is always preferred to the system b, i.e., the smaller value of ICC is preferred.
2. For a fixed system (a or b) ADE I is preferred with Motif A while it is disliked with Motif B.

This behaviour can be explained by the differences in the effective durations of the autocorrelation function of the source signals [3].

La localisation des sources sonores a fait l'objet de nombreuses recherches que ce soit dans le plan horizontal ou dans le plan vertical médian. Nous n'en connaissons pas cependant concernant l'espace entier. Aussi présentons-nous une "topographie du champ de localisation auditif".

**EXPERIENCE**

Trente-sept haut-parleurs sont disposés sur une armature hémisphérique centrée sur la tête de l'auditeur. En outre douze autres sont placés au niveau du sol. Les haut-parleurs sont alimentés successivement et indépendamment l'un de l'autre. L'auditeur doit donner l'indication de la provenance du son à l'aide d'un repérage horaire et de niveau : le niveau -1 est celui du sol, le niveau 3 correspond à la verticale.

**RESULTATS**

On a constaté une très bonne localisation latérale au niveau des oreilles. Par contre celle-ci est moyenne dans le plan médian. BLAUERT remarque que cette localisation dépend de la bande de fréquence utilisée. Une étude des erreurs systématiques semble confirmer cette observation.

La conjugaison de l'élevation et du déplacement de la source vers l'arrière amène à une localisation moyenne. Les sources situées au niveau du sol sont fréquemment localisées, non pas à leur emplacement mais symétriquement par rapport au plan horizontal.

Il faut enfin préciser que les résultats sont un peu meilleurs dans le demi espace gauche que dans le demi espace droit ; ce qui n'apparaît pas sur la figure qui est symétrisée.

**CONCLUSION**

La faculté de localiser l'origine d'un son, bien que plus ou moins performante selon les secteurs ne fait jamais défaut ; et l'élevation est perçue très nettement.

Cette recherche de résultats spatiaux est un préliminaire à une étude sur l'Hologphonie (2)(3) qui est la reproduction parfaite d'un événement sonore.

**REFERENCES**

INTRODUCTION
The Christchurch (NZ) Town Hall (2650 seats) is the major hall of a complex which includes also a 966 seat theatre, conference and catering facilities. The hall was opened on September 30, 1972. It is designed primarily as a concert hall - the only compromise being a requirement for half the main floor to be horizontal for civic functions. (Fig. 1)

ACOUSTICAL DESIGN AIMS
A full reverberation (RT in excess of 2 seconds) was to be provided by a large volume and hard surfaces. Audibility of the early lateral reflections was to be assured by reflectors directing them into the audience on paths remote from grazing incidence. The large interior suspended reflectors are independent of the room boundaries which determine the reverberant volume. The theoretical argument for the importance of these reflections has been reported elsewhere. Design included the conventional acoustical aims of freedom from echo, noise control and ensemble.

ACOUSTICAL DESIGN METHODS
The paper gives an account of a reflection RT masking study based on model echograms, modifications to the design based on analogue simulation of reflection sequences in the proposed hall, and the development of a ray tracing computer programme with graphical output.

MEASUREMENTS AND RESULTS
Reverberation time, EDT and integrated early energy measurements have been carried out in the room. Echograms for both total and lateral sound have been obtained using gated tonebursts suitably filtered. In view of the fullness of the reverberation (see fig. 2) there is an extraordinary degree of definition in the room, particularly in the gallery seats (fig. 3).

INTRODUCTION

Wellington, the capital city of New Zealand, is to build a new Town Hall. Its existing hall is too small, lacks adequate facilities and is an earthquake hazard in a zone of major seismicity. The architects for the Christchurch Town Hall (described in another paper at this Congress) were commissioned in 1975 to produce preliminary designs for the new hall. Marshall appointed as consultant, invited Hyde to collaborate with him on this project.

ORIGINS OF DESIGN PROPOSALS

(a) Architectural: The undoubted success of the arena plan for the Christchurch Hall led to the adoption of a similar form for the new hall. A major improvement is the elimination of the requirement for a flat floor.

(b) Marshall's early papers on the importance of lateral reflections (1) led to a comprehensive study "The effects of early reflections on Subjective Acoustical Quality in Concert Halls" (M.F.E. Barron 1974, Ph.D. Southampton unpublished) (2) The study includes details of masking of both high low frequency orchestral sound in lateral reflections, integration time, quantitative prediction of 'spatial impression', image shifting and the difference limen. These factors have all been taken into account in the new hall.

(c) Hyde's studies in California in association with Veneklasen (3) give complementary results in listener preference for relative levels of lateral, frontal, and reverberant energy. Model studies showed that in conventional fan shaped halls preferred levels of lateral (envelopmental) sound cannot be achieved. Improvements in seat absorption characteristics are available.

(d) Acoustical experience in Christchurch Hall show that in halls of this type relative levels of lateral sound within the preferred range are achievable.

SPECIFIC DESIGN OBJECTIVES

Details are given in the paper of architectural decisions, based on the foregoing to:

1) maintain the spatial impression, reverberance, and definition exhibited by the Christchurch Town Hall
2) integrated lateral energy between 0 and -5dB re frontal energy
3) smooth RT characteristic with T60 2 sec 500-2000 Hz and rising to 2.5 sec at 125 Hz due to improvement in seat design
4) raise main floor, lower stage to allow improved projection of rear instruments
5) provide reverberant coupling behind main lateral reflector and gallery to seats below
6) provide additional lateral reflectors behind gallery seating - no reflector closer than 10ms to nearest listener to avoid image shifting, increase diffusivity of closest reflectors
7) provide additional surfaces in region of stage for ensemble and balance
8) PNC 20, freedom from echo and intruding noise.

REFERENCES

CONSIDERATIONS OF ACOUSTIC DESIGN OF MULTI-PURPOSE HALLS IN JAPAN

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INTRODUCTION
Since 1960s, the construction of civic hall has been started in Japan, and now more than 500 halls are in general use. Most of them have the facilities of "multi-purpose" and the seating area of 1500. The multi-purpose halls in Japan are used in wide range such as Japanese traditional arts, dramas, classical music, popular music, ballets, operas and meetings. Furthermore, we have more strict regulations to the building construction for fire and emergency, and the construction should be proceeded in less cost and unthinkable short terms compared with other countries. This paper describes typical problems in acoustic design and features from the viewpoint of a professional acoustic consultant.

DESIGN FOR MULTI-PURPOSE HALL
(1) Characterization of the multi-purpose hall
The types and tendencies of the performances depend on the various facts of each district where the civic hall is to be built. It is quite difficult to predict in what way the hall is used in future. Generally, the large halls are designed for popular music, and small halls for recitals and meetings.
(2) Stage facilities
Stage reflectors are often seemed as an obstruction to the drama performance. Some halls succeeded in compatible use between large and medium sized hall using movable ceilings.
(3) Room acoustic design
From our experiences, the reverberation time of about 1.3 to 1.5 sec seems most suitable, and room noise of NC 20 to 25 is our objective value to design.

NOISE CONTROL
As the peculiar problems, we sometimes faced to the airborne and solid borne sound insulation between hall and adjacent rooms, reduction of sound transmission through smoke exhausting duct and control of vibration from the subways.

ELECTRO-ACOUSTIC SYSTEMS
As the multi-purpose hall is used mostly for pop music concert in Japan, the composition and performance of the electro-acoustic system has been played the important part of the performances of the hall. As their performances, a high sound volume such as SPL of over 100 dB is required. Besides proscenium speakers, two speaker systems at the both sides of the proscenium, and stage monitor speakers are required.

REGULATIONS DUE TO THE BUILDING CODES
We have the strict regulations to the building structures, materials, smoke exhausting openings for fire and emergency. Some of these are occasionally opposed to the acoustic requirements, i.e. the wooden materials are prohibited for the interior finishings of the hall, smoke exhausting openings or ducts should be equipped at the upper parts of the stage and seating area.
Abstract

The annoyance of road traffic noise was investigated in two field studies in the city of Zürich. In the first study, the traffic noise was measured in front of the living room windows of 800 randomly chosen apartments built between 1960 and 1972. In the second study, traffic noise was recorded in four streets with different traffic densities and various types of built-up area. Here, the measuring points were chosen just at the street fronts of the houses. Noise was measured with a sound level meter and expressed in cumulative frequency levels of $L_{50}$ and $L_1$; in the second study even in the equivalent noise level $L_{eq}$.

The surveys included questions on the degree of annoyance (day and night) and on specific disturbances (sleep, speech, communication, etc.), and individual reactions were investigated. Altogether, 800 interviews (first study) and 1300 questionnaires (second study) were analyzed.

Main results:

- The percentage of "frequently annoyed" people is increasing significantly in relation to the noise expressed as $L_{50}$, $L_1$ or $L_{eq}$.

- In the daytime at the highest measured noise values of $L_{50} = 74$ dB(A) or $L_{eq} = 77$ dB(A), 70 percent of the subjects are "frequently disturbed".

- In the night at noise values of $L_{50} = 52$ dB(A) or $L_{eq} = 63$ dB(A), annoyance increased markedly; at this level 70 percent of the subjects are "frequently disturbed". Beyond this level a further increase of annoyance is less pronounced.

- The use of the balcony and the number of open windows, specific complaints such as disturbance of sleep, rest and speech communication, as well as the use of medicaments and the tendency to move away are related to the noise levels.
Introduction:

This investigation is concerned with the analysis of the data of a questionnaire about the effect of environmental noise on the citizens of Indore city. Such surveys are carried out in India by PANCHOLY et al (1)

City Samples.

City of Indore is situated in the central West zone of the Madhya Pradesh State. It has a population of 600,000. In absence of proper town planning, the educational institutions, hospitals, workshops, cinema halls, business centres are most widely and randomly located. These centres, therefore, face the problem of noise pollution. It is, therefore, of interest to sample the data on noise pollution problem. With this aim in view following questionnaire was prepared and was circulated amongst the wide spectrum of persons residing in Indore City.

1. Do you consider your environment noisy?
2. Do you feel that noise reduces your efficiency?
3. Does noise disturb your sleep?
4. Do you know that loud noise is harmful to human organ?
5. Whom do you consider responsible for producing noise?
6. Can you suggest some means and ways for reducing noise?
7. Do you think that existing rules are insufficient to check noise pollution?
8. Which is the most disturbing noise in your view?
9. Can you pay more for appliances which are less noisy?

Fig. 1 Shows the types of people selected for survey.
Fig. 2 Depicts the analysis of the survey, for question number 5.

Conclusion: From the analysis of the data it may be concluded that about 55% of the persons interviewed considered their environment as noisy. However, they do not know the implications of noisy environment. Unlike water and air pollution which can be seen and observed noise pollution can not be seen but it has equally dangerous effect on human life which can cause sickness, hearing loss and heart troubles.

References:

Die Projektierung von Wohnüberbauungen an Hauptverkehrsstrassen in städtischen Agglomerationen erfordert die gründliche Abklärung der realisierbaren Schallschutzmassnahmen. Da in städtischen Verhältnissen strassen- und tiefbaugebietige Massnahmen wie Wände, Hügel, Blenden aus funktionellen Gründen nur in sehr beschränktem Masse anwendbar sind, muss das Schwergewicht auf die - Orientierung und Lage der Hochbauten, - die Grundriss- und Fassadengestaltung sowie - die hochbauseitigen Schallschutzmassnahmen gelegt werden.


Zur Optimierung der schalltechnischen Wirksamkeit im Rahmen der bauphysikalischen und baugesetzlichen Grenzen wird ein Bewertungsverfahren beschrieben, das dem Architekten und Bauherrn erlaubt, die Kostenwirksamkeit für die Baukörpervarianten zu quantifizieren.

LITERATUR
Einleitung

Die dauernde Entwicklung des städtischen mekanisierten Straßenverkehrs führt zur Notwendigkeit einer Organisation der Schallquellen im Bereich der Stadt, um das akustische Wohlbefinden ihrer Einwohner womöglich in bestimmten Grenzen zu sichern. In Rumänien ist der Bereich der Schallpegels im freien Raum der Stadt durch die Standards nur, 10.000/75 gesetzlich festgestellt, es gibt die Grenzen des Schallpegels in verschiedene funktionelle Zonen der Stadt an. Es werden verschiedenartige Massnahmen zur Verminderung des Schallpegels genommen.

Abhängig dieser Massnahmen, Karakters und Verkehrskapazität der Straße, auf der Systematisierungsplan der Stadt Bukarest, wurden die Verkehrsschallpegels laut Standard 10.000/75 ausgeführt. Um eine hohe Einwohnerdichte zu erreichen, ist man manchmal dazu gezwungen, entlang der starkverkehrten Hauptstrassen zu bauen.

Das hiesige Referat verfolgt diejenigen Massnahmen zu analysieren, die an schallverschmutzte und in Grenzbedingungen befindliche Wohnungen anwenden muss, indem man dabei die ökonomischen Bedingungen der typisierten Wohnblocks beachtet.

Die theoretische Grundlage

Die graphische Darstellung des Schallpegels als Funktion der Verkehrsldichte und die Schallverminderungskurven.

Die Analyse der Studierten Experimente


Schlussfolgerungen

Vorheben der Einführungsnötigkeit bei den Entwürfen und bei der Verwirklichung der Strassenfronten und in die technisch - ökonomische Analyse, des Schallverschmutzungs Kriterium als qualitatives Anzeichen der architektonischen Lösungen, durch deren Abschätzung man die Qualität der akustische Verschmutzung der Wohnungen für verschiedene Varianten der Strassenfronte analysieren kann.

Das Schallverschmutzungs ist durch Unterzuchungen auf Grund statistischer Angaben festgestellt worden.

Bibliographie

INTRODUCTION

This project was sponsored by the Pan American Sanitary Bureau, Regional Office of the World Health Organisation in order to recommend a practical approach to the measurement and control of urban noise within the State of Sao Paulo. Its short term objective was to recommend instrumentation and measurement techniques and long term objective is to provide the information necessary for establishing legislation to bring the problem of noise pollution under control. This paper presents the basic measurement programmes to be made and at the Congress it is hoped to report in further detail on the progress and success of the measurement to date.

PROGRAMME

It was decided to consider the field of urban noise pollution in six sections: (a) Neighbourhood Noise (b) Vehicle Noise (c) Traffic Noise (d) Airport Noise (e) Construction Site Noise (f) Industrial Health Risk from noise.

In each case, manpower, time and measurement programmes were drawn up and outline legislation suggested on the basis of current practice in Europe and the USA. Some extracts from the programme follow, but space does not permit the wider range of details which it is hoped will be available at the Congress.

Neighbourhood Noise. Noise complaints in the Sao Paulo area are currently handled by a division of CETESB which investigates, measures, and recommends action to local councils. General guideline recommendations were set out in respect of problems from noisy animals, unrequired music, noisy street trading, noisy conduct at night, acoustic warning devices, power boats, model aircraft, fireworks, car and motor cycle racing, noise from places of entertainment or shops, noise disturbance from a factory.

Vehicle Noise. The measurement procedure was considered in two stages. Firstly for detection of offenders, measurements at any location were proposed based on limits set 5dBA higher than the proposed legislation limit to allow for possible reflection effects and bearing in mind that it is the actual noise emitted by a vehicle which offends and not its noise potential. Proven offenders are then subjected to a controlled test on the lines of ISO Recommendation 362.

Traffic Noise. This is a proposed monitoring to assess the effect of other proposed noise control measures such as traffic streamlining to reduce stop/start regularity, vehicle noise limitation etc. The programme is based on Leq records at specified measurement points.

Airport Noise. The proposal here is for monitoring in the regions of airports. Sophisticated systems are not necessary here and a method of sampling dBD level and evaluating NNI was proposed.

Construction Sites. A double proposal was made; to measure the maximum dBA level and to monitor Leq over the 12 hour periods 06.00 to 18.00 and 18.00 to 06.00. The dual measurement was proposed in order to permit contractors to control their own noise using relatively simple instruments.

Industrial Noise. In line with thinking in other parts of the world an 8 hour exposure Leq of 90dBA was proposed using equal energy averaging. Before introducing legal limits, however, extensive study is proposed to determine its feasibility without extensive disruption of industry.
COMPUTER SIMULATION OF ROAD TRAFFIC NOISE

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INTRODUCTION Using a digital computer simulation model of road traffic flows it is possible to calculate traffic noise levels. Results are reported for a free flow model and compared with those given in the U.K. traffic noise prediction method (1) which have been derived from measurements.

THEORETICAL MODEL A section of road with a single traffic stream is considered. Vehicles have random velocities and headways on entering the section and also have their manoeuvring parameters defined. The noise level at 13.5m from the stream is calculated at each second using acceleration, velocity and position of each vehicle. Single vehicle noise emission functions (2,3) are used for this purpose. For the 500 secs of the simulation the various traffic noise indices can be calculated.

RESULTS AND DISCUSSION The simulation was run with various mean velocities and flow rates and the L10 levels are indicated in Table 1. Figure 1 shows the SPL at the reception point as a function of time for a mean traffic speed of 48 km/hr and flow of 600 v/br. This simulated curve has characteristics similar to experimental curves. In Table 1 the simulated L10 levels can be compared with those obtained from the prediction method (1) for free flow traffic situations. The agreement is good, but a trend can be observed in that the simulated results are greater than the predicted results for low vehicle flows and speeds and that the reverse is true for high vehicle flows and speeds. This type of simulation program enables the effects of traffic flow restriction on traffic noise to be investigated. Restricted flow simulation models are in the course of development.

<table>
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<tr>
<th>Flow Veh/hr.</th>
<th>Mean Vol. km/h</th>
<th>L10 dBA Simulated</th>
<th>From (1)</th>
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<tr>
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<td>67.7</td>
<td>65.5</td>
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<td>962</td>
<td>48</td>
<td>71.1</td>
<td>72.9</td>
</tr>
</tbody>
</table>

Table 1.

REFERENCES
(1) Calculation of Road Traffic Noise, Dept of the Environment, (1975)
(2) P.T. Lewis, J. Sound Vib., (1973), 30(2), 191-206
(3) M. Ringheim and S.A. Storeheier, Norwegian Inst. Techn. Report LBA 533
FACTORS AFFECTING RAILWAY NOISE LEVELS IN RESIDENTIAL AREAS

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INTRODUCTION

Railway noise levels are dependent on the distance of the observer from the track, the speed and type of train and, in addition, on the presence of earthworks and buildings alongside the track. This paper presents data that allow the effects of these factors to be estimated.

EXPERIMENTAL DATA

Experiments have been carried out to determine the effect of distance from the track and of train speed and type on the noise levels alongside the tracks (1,2). The relationships are summarised in Figures 1 and 2.

A series of measurements alongside railway cuttings and embankments showed that railway noise was not reduced by cuttings as effectively as road traffic in a cutting of the same dimension. In a cutting 6m deep the noise level was reduced by about 6 dB(A). Embankments, as expected, shield areas close to the track. Perhaps the most important factor affecting noise propagation from railway lines is the presence and configuration of buildings alongside the line. Results show that detached and semi-detached houses behave as insertion losses more than true barriers and the first row gives 8 dB(A) of level reduction whilst each succeeding row gives a further 4 dB(A) of reduction. Terraced houses give about 12 dB(A) reduction for 2 short rows 150m long, whilst long terraces (300m long) give 15 dB(A) reduction for 1 row and 17 dB(A) for 2 rows.

SUMMARY

These data can be used to estimate noise levels in any residential areas alongside railway lines. The accuracy of the method is at present being tested at 40,000 houses alongside 280 km of railway lines.

REFERENCES

INTERRUPTED TRAFFIC FLOW NOISE

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INTRODUCTION

This study compares the noise levels and subjective response between free flow and flow near to light-controlled intersections.

THEORETICAL BASIS

Regression techniques have been used to establish prediction equations for free flow traffic noise by a number of authors and one of these (1) was adopted as a base from which the increments in the levels of $L_{10}$, $L_{50}$ and $L_{90}$ due to the flow interruptions could be established.

Subjective response was found by a postal social survey, which included an overall satisfaction/dissatisfaction question and 9 on specific disturbances (5 on noise and 1 each on vibration, accidents, fumes and visual intrusion), using 5 point semantic differential scales.

EXPERIMENTAL TECHNIQUES

At three intersections a total of 133 traffic noise samples were taken up to 130 m from intersections, but at 7.5 m from the main traffic stream. Flow rates were around 1000 vehicles/hr. (20% heavies).

6 free and 6 interrupted flow sites were used for the social survey. Noise levels for each respondent were predicted from 18 hr measurements at two locations per site, and the trends from the initial study.

RESULTS AND CONCLUSIONS

The noise level increments against distance $x$(m) from intersection using linear regression yielded

\[
L_{10} = 3.21 - 0.01x \\
L_{50} = 5.62 - 0.03x \\
L_{90} = 9.55 - 0.03x
\]

The 'best' measure of 'annoyance' was found to be the total of the 9 individual scores less the accident question, and with the three noise related sleep disturbance questions averaged, giving a maximum possible score of 30. Correlation with 'dissatisfaction' score was 0%

The linear regression equations for 'annoyance' score, $A$, were

\[
A = 0.38 L_{10} - 9.0 \quad (0.22) \quad 17.80 \quad 139
\]

\[
A = 0.57 L_{50} - 17.2 \quad (0.25) \quad 18.15 \quad 129
\]

\[
A = 0.97 L_{10} - 47.5 \quad (0.56) \\
A = 0.66 L_{50} - 21.3 \quad (0.48)
\]

The different gradients of 'annoyance' against $L_{10}$, suggest that $L_{10}$ is inadequate. However, $L_{50}$ has similar gradients, but with the same 'annoyance' at about 2 dBA less for interrupted flow, giving 'neutral' responses at 57 and 59 dBA.

REFERENCES

Now in Japan, the predictions of road traffic noise are carried out by applying two correction terms to the results obtained through theoretical calculations. First correction term is related with the diffraction effect. The second one was introduced empirically to realize the best fit between measured and predicted results. This correction term is given as a function of distance and height of the receiving point (Fig.1). The details of this correction term are not quite sure, but it would be reasonable to consider that the ground attenuation would play an important role, except the case of elevated road.

EXPERIMENTAL

In order to clarify the effect of ground attenuation on the propagation of road traffic noise, field experiments and scale model experiments were carried out. At first, on the sound propagation over flat ground, the excess attenuations are closely related with the sound absorption of the ground surface.

In practical road situations, the effect of ground attenuation would often be combined with the effect of road constructions. Figs.2 and 3 show the examples of the results of scale model experiments. These results show that the sound absorption characteristics of the slope would play an important role. In case of depressed road, the situations were quite similar.

CONCLUSION

In case of the prediction of road traffic noise, the distance of the receiving point from the road is rather short, within 100 m. Even in these cases, for the precise prediction of noise, it would be necessary to consider the effect of the ground attenuation in some details.
INTRODUCTION AND THEORETICAL BASES

Important factors which determine the noise level at a receiving point from a noise source are reduction by distance, reduction by diffraction and reflection by surroundings.

The concept of the Fresnel zone construction shows qualitatively that the Fresnel zone of the youngest Fresnel number on the edge of a diffracting body contributes most strongly to the sound pressure level at a receiving point behind its back. The effects of the above three factors are connected linearly. The sound field around a diffracting body then can be quantified by the theory of quantification (1). It is a multivariate analysis to predict a quantitative outsider by qualitative factors.

EXPERIMENTAL TECHNIQUE

A running motor vehicle (m.v.) noise is used as a noise source. The quantification is made for the outsider of dB(A). Path difference $\delta$, in fig.1, of the youngest Fresnel number is chosen as a parameter of reduction by diffraction.

RESULT AND CONCLUSIONS

The multiple correlation coefficient is 0.89. The category weights of two factors and their regression lines are shown in figs.2 and 3. Noise level around a building by the running m.v. is shown in fig.4. Frequency spectrums of m.v.s. are similar and close each other. The above result is used to predict the noise level variation around a building from a road with many m.v.s. Predicted and measured values are compared in fig.5. $L_{eq}$, $L_{50}$ and $L_{eq}$ of them are also compared there.
INTRODUCTION

Methods of predicting the attenuation of traffic noise due to buildings have been tested on field measurements at a number of sites. Initial tests used manual calculation methods on data for sites of single family dwellings. More sophisticated calculation methods using computer techniques were used to predict attenuations for sites with multiple residence type buildings.

PART 1

In the first part of this study, measurements of the angular view of the road from scale site maps, diffraction over the roof from barrier attenuation charts (1), and approximate corrections for simple reflections, and diffraction around vertical edges were combined to predict attenuations. The prediction accuracy was poor, as many smaller details of the sites were ignored.

PART 2

To improve the prediction accuracy, a computer ray tracing programme was developed to predict the geometrical effects, and diffraction calculations were computed point by point and summed as the source was moved along the road. The ray tracing programme traced rays at 0.5° intervals from each observation point to the road. Excess attenuation due to ground absorption was included along the path of each ray. At each reflection the sound energy was attenuated according to the chosen reflection coefficient. Diffraction calculations were performed using Maekawa's (2) method as well as an equation for double edge diffraction by Russell and Ivey (3).

CONCLUSIONS

When tested on field data from multiple residence type buildings (apartment blocks, and row housing), the standard error of the prediction of $L_{eq}$ attenuations was 1.5 dBA.

REFERENCES

(3) Russell & Ivey, Inter-Noise '76.
Introduction

Depuis l'introduction en 1969, aux États-Unis, de la loi sur la protection de l'environnement (NEPA), le concept "d'Étude d'Impact sur l'Environnement" s'est propagé sans beaucoup progresser. L'ambition initiale d'entreprendre des études systématiques interdisciplinaires de l'effet de toute action sur l'environnement s'est vue considérablement réduite. Si on examine un seul effet, celui qui relève de l'acoustique, on peut construire une méthode d'évaluation.

L'IMPACT DE PHENOMÈNES ACOUSTIQUES SUR L'ENVIRONNEMENT

Une certaine confusion règne dans ce domaine depuis que les bruits et les vibrations ont été classés parmi les "actions" et non les "effets" dans la matrice dite de Leopold(1). On peut corriger ceci facilement. Il convient ensuite de choisir les critères selon lesquels un phénomène acoustique sera jugé et d'établir une procédure d'évaluation d'impact, qui notamment, devra absoudre tous les projets qui, par exemple, par leur taille ou par leur puissance, ne constitueront que des sources négligeables de bruit et de vibration.

CRITÈRES D'IMPACT

Il ne paraît pas raisonnable de vouloir quantifier l'effet "acoustique" d'un projet par un seul chiffre. C'est par négociation entre les administrations concernées et les auteurs du projet que les décisions doivent être prises. On peut cependant, comme Schultz le propose(2), calculer un nombre représentant la "population virtuelle éprouvée à 100%" en se basant sur une échelle d'impact de 0 à 100%, qui correspond à une gêne mesurée selon les réactions de la communauté, l'interférence avec la parole, le sommeil, etc. A cet effet, les niveaux de bruit sont décrits, selon les cas, par Ldn, Leq 24, Leq8, etc. Il est possible de prescrire l'utilisation d'indices différents pour des types de projets et des périodes différents : en l'absence de conclusions fermes relatives à la gêne, les indices ci-dessus suffisent.

PROCEDE D'ÉVALUATION D'IMPACT

Les étapes suivantes sont proposées :
1. Vérifier l'absence ou la présence du type de projet envisagé sur une liste de référence. La présence sur la liste dispense de formalités supplémentaires. Certaines conditions peuvent être imposées pour pouvoir obtenir cette dispense : puissance, dimensions, coût, niveau de bruit à la périphérie, etc..
2. Calculer le niveau de bruit créé par le projet sur le voisinage et évaluer la population virtuelle éprouvée. Comparer avec les critères et choisir l'option la plus favorable.
3. Grouper les études d'impact d'environnement, économiques, etc. et négocier avec les autorités compétentes et avec le public.

The need to respect precise environmental noise limits involves notable difficulty in the case of complex industrial plants, both at the planning stage and when existing plants are being made quieter.

The problem of environmental noise has, since 1973, been the object of detailed study by the Assessorato all'Ecologia della Regione Lombardia, which has worked out a proposal for 'Regulation of noisy activities.' This defines admissible sound levels with regard to the type of noise, the characteristics of the area (residential, industrial or mixed) and the time of day.

The possible solutions to the problem are:
- modification of the layout of the plant
- sound-proofing of the noise sources
- installation of acoustic barriers.

In order to have data, on which the choice of solution rests, easily available, the authors have prepared a mathematical model, and then used it in a Fortran program, to calculate automatically the sound levels outside industrial plants.

Input data consist of, particularly:
- acoustic characteristics of the noise sources (measured where possible, in conformity with NS S31-025)
- planimetry of the plant and the possible positioning of acoustic barriers
- characteristics of surrounding terrain
- atmospheric conditions

The program prints out the values of noise levels in different octave bands and in dB(A) at the nodes of an assigned network and at particular points of special interest.

Several applications to real situations have verified the good correspondence of the model's information to the values gathered experimentally.
INTRODUCTION
Man's acoustic habitat is regarded as a dynamic system consisting of brains which are capable of learning a language and which are connected to an environment and to each other by various communication channels. Hence, the results of Colton (1) are applicable to assess plans for such a habitat; specifically, this method illuminates aspects which cause some plans to have unexpected consequences.

MATHEMATICAL LEVEL
Because the system must exist at all times, not all points are accessible from any given point. This implies that some goals cannot be realized regardless of how much authority and funding are given towards their achievement.

PHYSICAL LEVEL
Any increase in cacotropy decreases the effective value of the switching variable. One consequence of this is that the system has to work harder and harder to get less and less done while frustration increases. Plans and planning methods which add substantial amounts of cacotropy to a system are self-defeating.

TECHNICAL LEVEL
Plans which fail to consider and utilize all available technological possibilities are unlikely to control modern technology. Although the technical means are at hand to return the school and the work place back to the home few planners have considered the challenge. It is not even known what sort of living patterns are desired and desirable!

Modern construction technology offers vast opportunities in shapes, colors, spatial arrangements, economics, etc. Yet most plans hinder rather than further their utilization. The concept of the omnibuilding stays avoided. The acoustic trade-off between wall space and weight for inexpensive foam-house construction has still not been adequately investigated!

Almost no plans make allowances for technical change and maintenance although it is well known that spavined and poorly maintained highways, sound barriers, insulation, etc. result in a deterioration of the acoustic habitat.

HUMAN LEVEL
Young persons require the excitement of high noise levels as a partial antidote to (planned) boredom. Many active persons have a need to make noise to overcome their anonymity. Noise regulations which ignore the need of the young are inherently unenforceable. An adequate acoustic habitat must allow for the above essentials plus the desiderata of other persons not to be disturbed.

Planning data regarding human response are usually based on methods which require applicability of the central limit theorem of statistics. The latter implies complete interchangeability with and indistinguishability from each other of all humans except for the measured variables. Or, in ergodic terminology, it implies that many observations of a single person during a long period of time must be fully equivalent to the same number of observations of many persons during a short time interval. Since these conditions are seldom applicable to annoyance surveys a planner should realize that community acceptance of any given noise will change unpredictably with time.

Plans which result in maximizing 'noise addiction' [Colton (2)] are self-defeating in the long run.

SOCIAL LEVEL
Plans which neglect the natural social inheritance of the human species create more problems than solutions. Yet most planners are not only unaware of the pertinent results from ethology but of the very existence of this biological discipline.

Plans which ignore criminal reality can hardly be recommended. If, for instance, a subway system has to be closed at 23:00 hours because rowdy elements cannot be controlled then car ownership, traffic noise, etc. are encouraged. It also has a positive feedback towards increased crime and deterioration of public transportation. Transportation which puts its customers at the mercy of the elements, strikes, sabotage, hi-jacks, muggings, diseases, etc. plus such indignities as weapon searches and which is unreliable and unpredictable and high-priced is a punishment rather than a blessing. A human habitat in which citizens are afraid to go out (at night) may be a relatively quiet one. It is not the kind people select voluntarily. But it is the kind in which civilization declines rather than flourishes. It also deteriorates the physical vitality and well-being of its victims.

Among ignored opportunities of modern planners are the interface problems of transportation modes, the many social and economic advantages of 24-hour usage of land and facilities (shopping centers, museums, libraries, etc.).

MANAGERIAL LEVEL
The acoustic habitat cannot exist in isolation. It must be integrated with the visual, olfational, tactual, thermal, etc. habitats. It should also be part of a functional and desirable way of life. Yet one notes that in the Managed Society even sound and thermal insulations still involve separate plans and agencies!

In modern planning a number of employees become a planning staff or department. This leads to the substitution of organizational and managerial rules, values, goals, ethics, etc. in lieu of the ostensible objectives. Thus there is an inability to cope with and to correctly balance and integrate all essential factors — and increasing frustration of the planners.

REFERENCES
(1) E. Colton, 4th ICA, Copenhagen 1962, H22; 5th ICA, Liege 1965, B62; 6th ICA, Tokyo 1968, C5-12; 7th ICA, Budapest 1971, 21H1
INTRODUCTION

As a nation with fast growing air traffic, Spain is facing today the well known problem of aircraft noise disturbance around airports. The authorities awareness of the situation abroad is advising them to take the first steps towards the control of aircraft noise exposure at this stage by preventing the development of conflicting situations in present and future airports.

Application of land use planning strategies around Spanish airports has been investigated for the Planning Section of the Technical Secretariat (Subsecretaria de Aviacion Civil).

THE GENERAL STRATEGY

From the point of view of airport management and control, Spain belongs to the group of countries where all airports are run by a central body (normally Air Force Ministry) with little participation from local or regional authorities. The initial approach to the problem, previous to give that needed participation, has been nation wide, common to all airports, for technical, economical and political reasons.

Within this common approach, Environmental Protection Areas, APA, defined on the basis of distance and orientation relative to runway alignment, are established for different runway categories, utilization and traffic distribution. The lines defining those areas have been obtained from noise exposure contours, but under special instructions noise indexes or acoustical terminology should be avoided.

The novelty of the approach, though, is the building up, like a puzzle, of the total noise exposure for an airport (in terms of the APA), from different components noise exposure (also in terms of the APA). Those components are runway category, predicted utilization, traffic mix and daily periods weighted accordingly. In this way, local authorities presented with the total picture and the components forming it, can choose, with the socio-economic interest of their communities in mind, the way they want their airport to be operated.

DEFINITION OF APAs

In all, three types of idealised runways have been studied, Transcontinental, International and Domestic, and for each type traffic volumes corresponding to 100, 50, 25 and 10% utilization have been used, to avoid grossly extending the APA in certain airports. It has been assumed present traffic mix as the worst case and noise exposure has been calculated in terms of CNEL, for day, evening, night and combined traffic.

The 75, 65 and 55 CNEL contours, squared by straight lines, define four areas for any combination of runway, utilization, traffic mix and time period. The inner area will have complete restrictions for any buildings other than those related to airport activities. The following area will allow only infill building, provided with high standard of sound insulation but avoiding critical installations (hospital, school...).

The third area will still demand sound insulation particularly on those critical buildings whilst in the forth area no restrictions will be applied.
BESTIMMUNG DER SCHALLIMMISSION DER ZELLENKÜHLTÜRME VON KERNKRAFTWERKEN AN EINER IMMISSIONSSTELLE

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EINLEITUNG


BERECHNUNGSERGEBNISSE UND MESSERGEBNISSE

Die Berechnungsergebnisse mit dem Programm "SCHALL" und die Messergebnisse der Schallimmission des Zellenkühlturms (17 Zellen; 50.5 MW/Zelle; 4140 m³ Wasser/h pro Zelle; 1 Ventilator/Zelle: 160 kW, 570 m³ Luft/s) sind in Tabelle 1 dargestellt.

Tabelle 1: Oktavband-Frequenzspektrum der Schallimmission des Zellenkühlturms in einem mittleren Abstand von 1024 m (dB)

<table>
<thead>
<tr>
<th>Frequenz, Hz</th>
<th>31.5</th>
<th>63</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
<th>8000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Berechnungs-ergebnisse</td>
<td>55.49</td>
<td>51.53</td>
<td>49.75</td>
<td>46.55</td>
<td>40.59</td>
<td>38.80</td>
<td>28.36</td>
<td>20.27</td>
<td>7.78</td>
</tr>
<tr>
<td>Messergebnisse</td>
<td>60</td>
<td>50</td>
<td>42</td>
<td>43</td>
<td>41</td>
<td>39</td>
<td>31</td>
<td>15</td>
<td>10</td>
</tr>
</tbody>
</table>

SCHLUSSFOLGERUNGEN

Der Vergleich der Berechnungsergebnisse (Vorhersage) mit den Messergebnissen bei jeder Oktavmittenfrequenz zeigt, dass die Differenz zwischen den betreffenden Pegelwerten meistens im Bereich der Messunsicherheit (±2 dB) bleibt und dass die Differenz zwischen den beiden A-Schalldruckpegeln (berechnet und gemessen) kleiner ist als 1 dB. Diese Untersuchung, wie auch andere ähnliche Untersuchungen der MC ING, beweist, dass die Vorhersage der Schallimmission (Immissionsprognose) durch das Programm "SCHALL" zu zuverlässigen Werten führt.
EINFLUSS DER RICHTWIRKUNG VON FLUGZEUGGERÄUSCHEN AUF DIE FLUGLÄRMBELASTUNG IN DER UMGEBUNG VON FLUGPLÄTZEN

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Institut für Bauphysik,  
Stuttgart Federal Republic of Germany


**Berechnete Linien gleichen äquivalenten Dauerschallpegels bei je 70 Starts von älteren (1) und neuen (2) Flugzeugen**

Außerdem ist die Richtcharakteristik der Schallabstrahlung der Fluggeräusche von Bedeutung für die Wirkung von Abschirmmaßnahmen.
PREVISION DES NIVEAUX SONORES D'UNE INSTALLATION INDUSTRIELLE. DOCUMENT DE GENIE ACOUSTIQUE

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On sait que l'on entend par "GENIE ACOUSTIQUE" l'application des lois de l'acoustique à l'art de construire, afin de rendre une installation industrielle acceptable tant pour le voisinage que pour le personnel qui y travaille.

Une méthode de Génie Acoustique existe, elle a déjà été présentée par Messieurs MATTEI et FRANCOIS au 5ème Congrès International d'Acoustique en 1965 à LIEGE.

La réglementation française actuelle exigeant le respect au voisinage de limites acoustiques préalablement fixées, un programme de Génie Acoustique a été mis au point afin de réduire les temps de calculs et d'augmenter le nombre d'informations significatives. Il permet également d'étudier plusieurs dispositions afin de retenir l'implantation optimale.

Afin que ces prévisions se traduisent dans la réalité, Electricité de France a été amené à établir un document de Génie Acoustique pour chaque type d'installation de production donné.

Ce document, qui est une sorte de catalogue, permet tout au long des études et de la réalisation :

- d'inventorier les matériels concernés,
- de rappeler les clauses contractuelles et les objectifs fixés,
- d'assurer la détermination des gabarits d'insonorisation nécessaires à la protection du voisinage et du personnel,
- de fixer le planning des différentes réunions de mise au point entre le projeteur et le réalisateur,
- d'assurer une homogénéité des exigences pour les différents matériels,
- de faciliter les contrôles acoustiques et l'ajustement de l'installation au cours du temps,
- de mettre en évidence les retards éventuels des réalisations d'insonorisation.

Enfin ce document permet de déterminer, après la mise en service de l'installation, l'importance des efforts consacrés à la lutte contre le bruit.

Vor Jahren hat man in Ungarn zwei modernen Mühlen gebaut. Die ungünstigen akustischen Verhältnisse dieser Mühlen haben in der Bewohnerschaft bei den Werktätigen und den Fachleuten gleicherweise ein Missfallen ausgelöst. Es hat sich bestätigt, dass die akustische Planung der Mühlen im Interesse der Bewohnerschaft und der Werktätigen der Mühle unerlässlich ist.


Der organische Teil der Planung ist die Ausbildung des Milieus der Mühle. Wir bepflannten mit Bäumen die Umwelt der Mühle maximal. Diese Lösung wird die psychischen Reaktionen der Bewohnerschaft vermindern.


Die vorteilhafteste Lösung ist natürlich die Ausrüstung mit leisen Maschinen. Nach unseren Vorschlägen wird man für die geräuschvollsten Maschinen eine Neuplanung machen.

Die Planung der Mühle in Miskolc ist ein gutes Beispiel für die nutzbringende Kollaboration der Konstrukteuren und der akustischen Experten. Nach die ausgeführten wirtschaftlichen Kalkulationen sind die Kosten der akustischen Planung kleiner, als die erreichbare Einsparung durch die Herabminderung der Spedition.
INTRODUCTION

A model is presented to calculate the approximate costs of reducing noise emissions caused by large industrial complexes to a desired level in the surroundings.

THEORETICAL BASES

a. The conventional theory of hemispherical propagation of sound from sources plus atmospheric and ground attenuation has been programmed for a computer and combined with a plotting routine.

b. For refineries and petrochemical plant built in the open air Concawe have further developed a relation between cost and noise reduction obtained (ref.), see Fig. 1. It holds for noise reduction planned in the design stage of plant. From experience it is learned that the cost of noise reduction in existing petrochemical plant is at least twice as high.

EXAMPLE: "The Rijnmond industry noise study"

Concawe determined the source strength of the large industrial complexes in the estuary of the river Rhine, west of Rotterdam - about 2 km wide and 20 km long. With these data and the computer program an acoustic model of the whole conglomerate was set up. Its validity was checked against third party measurements. Information on cost of "in-plant" investment was obtained via a questionnaire. After incorporating a growth concept, costs of noise reduction were calculated for three particular emission levels along the surrounding residential areas and for various rehabilitation periods.

CONCLUSIONS

- A noise emission calculation programme can be used for existing and future situations (e.g. planning).

- An unrealistically low level (e.g. 40 dB(A)) at night makes the costs extremely high.

- The shorter the period of rehabilitation the higher the costs.

- Increasing the nighttime level from 40 dB(A) to 50 dB(A) would lower the costs to less than one tenth.

REFERENCE

Control of community noise from petroleum and petrochemical plant, P. Sutton, The Chemical Engineer, April 1969.
INTRODUCTION

On présente un modèle de simulation du comportement acoustique d'un trafic automobile aux abords d'un système de feux, permettant de restituer la signature acoustique temporelle à partir de laquelle tous les types d'indices acoustiques peuvent être calculés.

SIMULATION DU TRAFIC ET EMISSION ACOUSTIQUE DES VÉHICULES

Le calcul est centré sur une simulation microscopique d'une portion du réseau urbain dans laquelle chaque véhicule est individualisé. Les véhicules réagissent entre eux par une loi de poursuite, et avec les feux. Les paramètres d'entrée sont les débits en véhicules légers et véhicules lourds aux entrées du réseau, les vitesses maximales désirées, les cycles des feux de carrefours. Chaque véhicule est considéré comme une source acoustique ponctuelle et omnidirectionnelle, à laquelle on affecte une puissance acoustique instantanée qui est fonction de sa vitesse, de la charge du moteur et du rapport de boîte utilisé. Pour le rapport de boîte i, la relation est de la forme :

\[ L_{wi} = (a_i \cdot \log v) + (b_i \cdot y - \log v) + c_i \cdot y + d_i \quad (1) \]

où \( L_{wi} \) est le niveau de puissance acoustique, \( a_i, b_i, c_i, d_i \) sont des constantes, \( v \) est la vitesse, \( y \) l'accélération.

RESULTATS ET CONCLUSIONS

En additionnant la contribution sonore à chaque instant de tous les véhicules en des points privilégiés du réseau, on simule la signature acoustique reçue en ces points sur une durée de quelques cycles de feux (Fig. 1).

La comparaison des niveaux simulés et mesurés a été effectuée pour quelques sites urbains de caractéristiques proches des hypothèses de simulation, et montre avec quelques restrictions une bonne concordance d'ensemble (tableau 1).

<table>
<thead>
<tr>
<th>Indice</th>
<th>( L_1 )</th>
<th>( L_{10} )</th>
<th>( L_{50} )</th>
<th>( L_{90} )</th>
<th>Leq</th>
<th>( \sigma )</th>
</tr>
</thead>
<tbody>
<tr>
<td>mesuré</td>
<td>85</td>
<td>78</td>
<td>62</td>
<td>56</td>
<td>74</td>
<td>85</td>
</tr>
<tr>
<td>simulé</td>
<td>84</td>
<td>78</td>
<td>63</td>
<td>57</td>
<td>74</td>
<td>8</td>
</tr>
</tbody>
</table>

Tableau 1 : Niveaux mesurés et simulés 50m en aval d'un feu

En conclusion, on montre qu'il existe un "effet carrefour" caractérisé par un contraste de bruit entre l'amont et l'aval sur une longueur variant en fonction du type de trafic et de feux. Ce programme de simulation permet d'étudier qualitativement cet effet, avec application possible à la planification du trafic dans les villes.

REFERENCES :
DETECTION OF NATURAL REFLECTIONS FOR URBANISM PROJECTS

Querol, J. M. N. | Consultores Acústicos
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Relea, F. G. | INGEST
Madrazo, 14 - Barcelona - Spain

INTRODUCTION
High level noise sources, such as industries and motorways, are sometimes placed aside a hill or natural slope which may reflect the sound towards living areas.

When designing a barrier (B) close to the noise source (S) for the direct wave, it may well happen that reflected waves reduce or invalidate the barrier’s effects. (Figure 1).

An "in situ" measurement method has been developed in order to detect the existence of reflected waves and evaluate their importance at points placed in the receiver's area.

EXPERIMENTS
The equipment involved is very similar to those that measure the reverberation time with a sound pulse generation.

The experiments have been carried out by bursting detonation cartridges exactly in the future noise source location and recording the direct and reflected waves either by magnethphone or graphically, in those places where needed.

The direct wave gave in the graphic recorder a first and clear peak, after which smaller ones (Figure 2) appeared sequentially. Those were produced by reflected waves.

The peak height difference between the first and all others assessed the relevance of the reflected waves. The relative position between peaks in the graph in relation to the burst time aided the identification of the liable reflecting surfaces.

In cases when the reflecting surfaces have no defined shapes, additional measurements in auxiliary points have to be carried out.

In every case the measurements were carried at the main frequencies of the spectrum presumed for the future noise source.

CONCLUSIONS
The techniques used take into account the real structure of the studied land (Directivity and absorption).

It has revealed very useful when a election between acoustic enclosure or barrier has to be taken towards outdoors noise sources.

It has allowed a more strictevaluation of lay-out goodness in industrial plant where the buildings are placed in such a way to give an acoustic barrier effect.
Zum Schutz der Bevölkerung gegen Lärm sind in der Bundesrepublik Deutschland in Rechtsvorschriften Schallimmissionsrichtwerte festgelegt worden. Anlagen der Chemischen Industrie und andere Schallemittierende Anlagen dürfen danach nur noch errichtet oder erweitert werden, wenn sichergestellt ist, daß der von ihnen erzeugte Beurteilungspegel in der Umgebung nirgends die Immissionsrichtwerte überstreitet.

Bis vor einigen Jahren sind aber an vielen Stellen Wohngebiete neben lauten Industriegebieten entstanden; dort werden die Immissionsrichtwerte heute häufig noch weit überschritten.


Dagegen können in bestehenden Industriegebieten neue Anlagen heute meist so errichtet werden, daß sie die vorhandenen Schallpegel in der Umgebung kaum erhöhen. In Verbindung mit Maßnahmen an den bereits vorhandenen Anlagen ist es häufig sogar möglich, die Schallemission allmählich zu verringern.

Wenn das nicht möglich ist, ist es schalltechnisch sinnvoller, die zu schützenden Menschen in ruhigen Gebieten anzusiedeln, als neue Industrieanlagen, die besser nahe bestehenden lauten Anlagen errichtet werden sollten.
MODELLING OF ENVIRONMENTAL ACOUSTICS

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INTRODUCTION

The solution of environmental acoustics problems by analytical approximations and computer simulation has, to date, proved inadequate in many situations. It would seem that scale modelling is possibly the most accurate predictive tool available which has general application to difficult geometry.

BASIC PRINCIPLES

The general principles of scale model analogues have been discussed by various authors (1,2,3,4). It has been shown that accurate modelling requires that several basic principles be satisfied:

a) Geometric similarity requires generally that the parameter \( \lambda/d \) be the same in the model and prototype.

b) The acoustic impedances in the model at the scaled frequencies must be the same as in the prototype at the original frequencies.

c) The quantity \( a \lambda \) must be the same in the model and prototype if the attenuation of sound by the gaseous medium is to be modelled correctly.

In practice, gaseous absorption with its frequency dependance can present a difficulty for scale models. Accurate predictions require corrections of the model data to be made particularly when reasonably large scale factors are used. With this limitation, it should be noted that with strict adherence to points a) and b), the mechanisms of diffraction, scattering and the propagation of sound over the ground will be modelled correctly.

MODELLING MATERIALS AND GASES

Alternatives to using air in the model have been considered (4). The use of heavy gases such as Freon-12 may be advantageous in extending the limits of scaling. For this particular gas, corrections for the non-scaled absorption would still be required (5).

The methods of measuring the acoustic impedances of materials at high frequencies and oblique angles of incidence have been described (6). Suitable modelling materials, including materials for accurate modelling of ground surfaces have been found. Models have been built and tested to compare the results obtained with field measurements. Fairly accurate agreement has been found between the two sets of data (7).

LIMITS OF MODELLING

The various factors which limit the scale usable for modelling will be discussed. Briefly the limitations set by experimental technique include: size of transducers, frequency limitations of associated electronics, power of sound sources, and characteristics of model materials. The limitation imposed by the non-scaled attenuation of the gas used becomes progressively a greater problem as the ratio of frequencies in the model/frequencies of the prototype increases.

REFERENCES

(3) Nijs, L. and Heringa P.H., Institute of Town Planning Research, Delft University of Technology, Netherlands, Memorandum No. 8, 1975.
(7) Jones, H.W., Stredulinsky, D., Vermeulen, P.J., "A Study in the modelling of acoustics for the urban environment", to be published.
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Sektion für Städtebau, Geschicthe und Architektur bei der Bulgarischen Akademie der Wissenschaften - Sofia

Der Vortrag begründet den Gedanke über den sich der Verfasser bereits früher geäußert hat (1), in einem neuen Abschnitt die Akustik der Umwelt und die Probleme des Lautmediums in dem der Mensch lebt und schafft zu vereinigen. Das Schallfeld ist einer der Parameter der Umwelt, der Lebenshaltung. Mittels der akustischen Signale, die der Mensch empfängt und ausstrahlt steht er in Verbindung mit der Umwelt, im weitesten Sinne des Wortes. Für ihn sind die akustischen Signale nützlich oder schädlich, angenehm oder unangenehm. Angenehme akustische Signale sind diejenigen, die eine nützliche Information für das Leben, die Tätigkeit und die geistigen Bedürfnisse des Menschen übertragen und unangenehme, unerwünschte sind die akustischen Signale die psychologische Erregungen und physiologische Schäden hervorrufen.

Diese Teilung setzt die Lösung von zwei Grundproblemen voraus – die angenehmen, nützlichen Signale zu bewahren, zu bereichern oder zu verstärken für die allgemeine Nutzung der Menschen, und die unangenehmen zu entfernen und herabzusetzen. Vor allem ist dies mit der Kenntnis der Struktur des Schallfeldes in dem sich der Mensch befindet, sowie mit den Erscheinungen und Gesetzmäßigkeiten der Verbreitung und des Verklinges der akustischen Signale und der beim Menschen hervorgerufenen Empfindungen verbunden. Die bisherige Erfahrung zeigt, dass die Methoden und Mittel zur Erforschung, Bewertung und Regulierung der Schallfeldstruktur, bedingt durch die zwei Signalarten, allgemein sind. Deswegen erweist sich notwendig die Probleme, die bis jetzt unter verschiedene Abschnitte, wie Architektur- und Bauakustik, Schallmessung, Bewertung und Lärm- und Schwingungsbekämpfung, Raumakustik klassifiziert wurden, zu vereinigen (2,3)

Begründet werden die Probleme, verbunden mit dem vorgeschlagenen Gedanke, mittels der entsprechenden Klassifikation veranschaulicht im nachstehenden Schema.

(2) 3, 6, 7 und 8. ICA Kongresse: Lüttich, Tokio, Budapest, London
(3) BÖ Band 2 und 3.
ACOUSTIC PLANNING FOR A NEW CITY

Adamson, B.M. 11 Clifton Street, Prospect, South Australia, 5082

INTRODUCTION

Monarto will be a new city of 180,000 people, established in a currently rural area in South Australia. State Government directions give noise a high priority in planning the city. This paper describes the acoustic planning work (1) for Monarto.

NOISE STANDARDS

Following local practice, and for ease of measurement, $L_{10}$ and $L_{90}$ dB(A) were used in all noise standards for Monarto. A minimum $L_{10} - L_{90}$ of 7 dB(A) was found desirable, so standards were written in $L_{10}$ only, with this difference implied.

It was desirable and possible to plan the new city to be significantly quieter than existing cities. Major factors considered in setting noise standards were noise levels in Adelaide, (the State capital) variations in noise sensitivity within the community, and speech and sleep interference. Noise standards were recommended for three time periods, as shown in Table 1. These standards apply to each hour within each period, not to the whole period.

Limited exceedance of noise standards is permissible in 25% of residential areas. Construction noise may exceed day standards by up to 15 dB(A).

<table>
<thead>
<tr>
<th></th>
<th>Day</th>
<th>Evening</th>
<th>Night</th>
</tr>
</thead>
<tbody>
<tr>
<td>Residential Areas</td>
<td>50</td>
<td>45</td>
<td>40</td>
</tr>
<tr>
<td>Commercial &amp; Community Areas</td>
<td>60</td>
<td>60</td>
<td>60</td>
</tr>
</tbody>
</table>

Traffic noise predictions showed that the residential area noise standards would be exceeded within 300-500 metres of arterial roads, unless some form of screening is used. Where possible, noise-insensitive buffer zones will be placed beside main roads. Elsewhere, roadside barriers costing $A15,000 to $A50,000 per kilometre may be used. Building design to minimise noise intrusion is feasible for non-domestic buildings, but may be difficult in residential buildings for climatic reasons.

Large construction projects may create noise problems for up to 20 years. Estimates indicated that site boundary noise limits of 70-85 dB(A) can be achieved by improved practices and use of new quieter equipment, while adding less than 1% to project costs.

Planning of industrial areas will allow the standards of Table 1 to be met with industrial noise limits comparable to most other parts of Australia. Noise control costs would be from 0% to 3% of investment.

SUMMARY

Detailed consideration of noise at the planning stages can produce significantly quieter new cities. The planning work for Monarto has shown that this is both technically and economically feasible.

REFERENCE


Die Lärmkarte ist dabei als Kursbezeichnung für ein Rechenprogramm zu verstehen, das nach Eingabe der relevanten Daten für die Bebauung und für die Emissionsdaten der Lärmquellen unter Berücksichtigung der Gesetze der Schallausbreitung (z.B. Beugung, Reflexion) als Ausgangswerte die räumliche Schallpegelverteilung eines Gebietes in sinnvoller Genauigkeit liefert. Die Genauigkeit braucht zugunsten einer schnellen Rechenzeit nicht höher zu sein als die Genauigkeit der verfügbaren Eingabedaten. Messungen sind nur in der Erprobungsphase interessant, wenn die Genauigkeit des erstellten Programmes getestet werden soll.


Das Endergebnis des Rechenprogrammes ist die kartenmäßige Darstellung der errechneten Immissionswerte durch einen Plotter. Soll die Lärmkarte vom Anwender als Hilfsmittel für die Bauleitplanung voll akzeptiert werden, ist die Übersichtlichkeit der Darstellung, die beispielsweise eine Überschreitung der zulässigen Planungserrichtwerte sofort erkennen läßt, als eine der wichtigsten Anforderungen an die Lärmkarte anzusehen.
1. Environmental acoustics

C. BUILDING ACOUSTIC
INTRODUCTION

Author is of opinion that the objective method of investigations used in room acoustics should approach the sound perception mechanism by the human ear in order to obtain better agreement between the objective and subjective evaluations of acoustic quality of a room. One of an important element of the perception process is the spectral analysis of signals considering their dynamic character i.e. analysis of their transient states being the basic part of experimental investigations realized.

EXPERIMENTAL

The opinion presented above became the starting point for investigations, basically aimed to evaluate the changes of spectral components amplitude of sound transient states i.e. its stages of the growth and decay, depending on changes of the acoustic parameters of room. For these chosen time sections of the signal, measured at various acoustic room conditions, the power spectral density functions have been calculated by means of the FFT method. For the power spectra the following parameters, among others, have been determined:

- \( \Delta L \), \( \Delta L_p \), \( \Delta L_v \) determining the differences of sound levels of successive harmonics of spectrum for the growth and decay of the signal, obtained in rooms under investigations in relation to the levels of such harmonics for the reference sound;
- \( d_{xy} = \sqrt{\sum (L_{xy} - L_{xy,0})^2} \) expressing the \( d_{xy} \) vector length in so-called deformation space;
- \( \delta = \sqrt{\frac{1}{N} \sum (L_{xy} - L_{xy,0})^2} \) being the measure of so-called anharmonic effect noticed when changing the room acoustic properties.

CONCLUSIONS

The research which has been carried out allow to draw the following conclusions:

1. The magnitude of sound spectral structure deformation, measured in the selected point of the room, is changing depending on the kind of signal and its time section /i.e. growth or decay/ being considered.
2. It has been stated, that essential information about the harmonic and formant structure of the speech sound are contained even during the sound decay process. The amount of those information, though rather limited, nevertheless it is not decreasing significantly with the duration time of that process, i.e. with decreasing of sound energy of decaying sound.
3. In certain cases interesting anharmonic effects occur in decay process of signals having harmonic structure, especially in rooms with long reverberation time. Depending on frequency band and position of measured point these anharmonic effects reach sometimes values up to 3.5 Hz.

REFERENCES

INTRODUCTION

At low frequencies the sound field in reverberant rooms does not satisfy the assumptions for a description in terms of the superposition of a large number of plane waves with random incidence in direction and phase. Another model, which gives an adequate description of the sound field at lower frequencies, is the "eigenmode" model, in which the sound field is considered to be a superposition of a number of eigenmode fields. The eigenmodes, or natural modes, are characterized by an eigenvalue (resonant frequency) and an eigenfunction (standing-waveform).

ADVANTAGES OF NON-RECTANGULAR REVERBERANT ROOMS

Considerations based upon the eigenmode model indicate that the regular shape of the eigenfunctions in rectangular rooms is a disadvantage with respect to the statistical homogeneous spatial distribution of the sound pressure over the room, which is not the case in non-rectangular rooms. Apart from this effect, reverberant rooms with irregular shapes have additional advantages concerning the smoothness of the rate of decay of the sound pressure and the disturbance of the modal pattern by voluminous sound sources.

CALCULATIONS AND MEASUREMENTS

A disadvantage of irregular rooms is that the eigenfunctions cannot be calculated analytically. However, with the finite-element method the acoustical response of these rooms can be calculated numerically. The accuracy of this calculation decreases with increasing resonant frequency, depending on the number of elements into which the room is divided. For a number of two- and three-dimensional models of non-rectangular reverberant rooms the eigenmodes have been calculated. Analysis of the data (e.g., by calculating the standard deviation of the amplitudes with which several neighbouring modes are excited, at several points in the room) shows that a non-rectangular reverberant room tends to give a more homogeneous pressure distribution than a rectangular room. The resonant frequencies have also been measured in a non-rectangular room model (scale 1:4) by means of a Fourier analysis of the impulse response. Agreement between measurements and calculations of better than 0.5% was obtained for the first 14 eigenmodes (the highest one corresponding to 65 Hz in a 230 m³ room). An accuracy of about 1% is expected to be obtained at 100 Hz. For a number of eigenmodes the distribution of the sound pressure in the non-rectangular room has been measured. The results show very good agreement with the calculated predictions.
DEVELOPPEMENTS DE LA METHODE DES RAYONS EN ACOUSTIQUE DES SALLES

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1.- CALCUL DE LA PRESSION SONORE PAR UNE METHODE DE RAYONS

Depuis une dizaine d'années, divers auteurs ont développé la méthode des rayons qui s'est avérée très fructueuse pour caractériser l'acoustique d'une salle au niveau même de sa conception. On peut cependant reprocher à la méthode des rayons d'être un appauvrissement de celle des images : en effet, les images permettent d'évaluer la pression sonore, les rayons sont purement énergétiques. Cependant, en considérant les rayons comme issus d'images, on peut espérer connaître la pression sonore en un point de la salle considérée. On donne un exemple de calcul de l'établissement du son dans une salle rectangulaire, et on discute la validité des résultats obtenus, suivant le nombre de rayons émis et le temps compté à partir de la réception du son direct.

2.- EFFET D'UN COEFFICIENT DE REFLEXION VARIABLE AVEC L'INCIDENCE

En général, dans les calculs de rayon, on envisage un coefficient de réflexion moyen, indépendant de l'incidence, pour les différentes parois envisagées. On peut se demander si cette simplification est bien justifiée. Deux exemples montrent qu'en effet, la prise en compte d'une variation du coefficient de réflexion avec l'incidence n'amène pas de changement sensible pour la courbe d'énergie cumulée ou l'écho-gramme.

3.- CHOIX DE LA MEILLEURE DISPOSITION DES ABSORBANTS DANS UNE SALLE

Nous avons pris le cas d'une piscine en forme de demi-ellipsoïde de révolution où le niveau sonore et la réverbération sont excessifs, en l'absence de traitement acoustique. Des calculs effectués par la méthode des rayons ont montré quelle était la meilleure disposition des absorbants parmi les diverses solutions envisagées (absorbants au centre, en couronne, en secteurs circulaires).  

A titre d'exemples, nous donnerons dans le tableau ci-dessous, quelques résultats relatifs au niveau sonore reçu N par rapport au niveau sonore émis pris à 1 m de la source, ainsi qu'à la durée de réverbération Tn dans le cas où la coupole est recouverte à moitié d'absorbants. On a envisagé deux situations S1 et S2 pour la source et deux situations R1 et R2 pour le récepteur.

<table>
<thead>
<tr>
<th>Situation</th>
<th>Disposition des absorbants</th>
<th>N(dB)</th>
<th>T(s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>S1 R1</td>
<td>Zone centrale (cercle)</td>
<td>-14,5</td>
<td>2,7</td>
</tr>
<tr>
<td>S2 R2</td>
<td></td>
<td>-18,4</td>
<td>3,2</td>
</tr>
<tr>
<td>S1 R1</td>
<td>Couronne extérieure au cercle</td>
<td>-11,5</td>
<td>4,6</td>
</tr>
<tr>
<td>S2 R2</td>
<td></td>
<td>-15,6</td>
<td>4,9</td>
</tr>
<tr>
<td>S1 R1</td>
<td>Secteurs circulaires de 10°</td>
<td>-18,1</td>
<td>0,4</td>
</tr>
<tr>
<td>S2 R2</td>
<td>dont 1 sur 2 est absorbant</td>
<td>-18,9</td>
<td>0,8</td>
</tr>
</tbody>
</table>

Il apparaît nettement que la disposition des absorbants en secteurs circulaires est la plus efficace pour diminuer le niveau reçu aussi bien que la durée de réverbération.
THE STATISTICS OF DECAYING SOUND FIELDS

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INTRODUCTION

This paper presents theoretical formulae for the variance of decay rates in reverberation rooms. Two types of variance are examined. They are the variance between decay rates measured at the same microphone position (ensemble variance $\sigma^2$), and the variance between decay rates measured at different microphone positions ($\sigma^2$). Two types of signal averaging are considered. They are exponential averaging (which includes graphic level recorders and sound level meters), and linear averaging over a fixed averaging time (where the interval between the starting time of successive averages is greater than the averaging time).

THEORETICAL FORMULAE

Spatial Variance:

$$\sigma^2_s = \left( \frac{10}{\ln 10} \right)^2 \frac{12}{B} \left( \frac{D}{10} \right)^3 F \left( \frac{\ln 10}{10} \right) D$$

(1)

Ensemble Variance:

For exponential averaging device,

$$\sigma^2_e = \left( \frac{10}{\ln 10} \right)^2 \frac{12}{B} \left( \frac{D}{10} \right)^3 F \left( \frac{\ln 10}{10} \right) Y$$

(2)

For linear averaging device

$$\sigma^2_e = \left( \frac{10}{\ln 10} \right)^2 \frac{12}{B} \left( \frac{1}{BT} + \frac{2}{N} \right) \frac{\left( D^2 \frac{P}{DA} + 2 \right)}{\left( D^2 \frac{P}{DA} + 1 \right) \left( D^2 \frac{P}{DA} \right)}$$

(3)

where

$$F(x) = 1 - \frac{3}{x} (1 + e^{-x}) - \frac{12}{x^2} e^{-x} + \frac{12}{x^3} (1 - e^{-x})$$

(4)

$B$ = statistical bandwidth of bandpass filter (Hz)

$D$ = mean decay rate of room in band (dB/s)

$N$ = number of decibels of decay measured (dB)

$Y$ = ratio of decay rate of exponential averaging device to decay rate of room (-)

$I$ = averaging time of linear averaging device (s)

$N$ = number of samples of signal averaged by linear averaging device (-)

$a$ = interval between the starting time of successive averages of the linear averaging device (a > I) (s)

EXPERIMENTAL RESULTS

The above formulae have been compared with the results of a statistical analysis of variance on experimental data from four different reverberation rooms in both their empty state and with a highly absorbing sample on their floors. Good agreement was obtained, except at low frequencies.

CONCLUSION

The theoretical derivation of the formulae and the results of the statistical analysis will be published in a paper by Davy, Dunn and Dubout.
REVERBERATION TIME OF RECTANGULAR ENCLOSURES

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INTRODUCTION

This work is a generalization of the ideas which the author presented on cylindrical rooms at the IV Latin American Meeting on Acoustics (September 1975). We eliminate the concept of "mean free path" and introduce the idea of "directional extinction time" for each direction of emission from the sound source.

THEORETICAL BASES

A rectangular room is assumed as indicated in Fig. 1 with the sound source at the apex of one of the trihedrons. Each emission direction is defined by the angles $\psi$ and $\phi_1$. Specular reflection is assumed on all the walls and the analysis is based on the method of multiple images of the reflecting planes.

RESULTS AND CONCLUSIONS

For each direction defined by $\psi$ and $\phi_1$, the number of reflections produced at each of the walls of the room is computed after a given time.

Considering a uniform absorption coefficient $\alpha$ for all surfaces of the room, the final formula for the reverberation time is:

$$t_{60} = \frac{0.041X}{-\ln(1-\alpha)} \cdot K$$

where $K$ is a characteristic factor of the dimensions $X$, $Y$, $Z$ of the room which can be used as tabulated data. For a cubical room ($X = Y = Z$) we have:

$$K = 0.69$$

In this case the result is coincident with that obtained by the concept of mean free path and the ratio $V/S$ (volume/area).

An important result of this research is that it allows calculation of the reflection on each room wall separately, the "weight" or final contribution to $t_{60}$ for a given $\alpha$ can be evaluated, assuming it corresponds to any wall.

REFERENCES

(1) F. V. Hunt - Journal of the Acoustical Society (1964) v. 36 - p. 556
(2) H. Kuttner - Room Acoustics (1973) Cap. V - p. 105
INTRODUCTION

L'exigence d'une méthode subjective satisfaisante pour évaluer la qualité de l'écoute des signaux acoustiques dans les salles est largement connue et dans ces dernières années le problème a été l'objet de nombreuses discussions.

En effet une méthode subjective valide et très simple permettrait de résoudre deux ordres de problèmes :
- Exprimer l'évaluation de la qualité d'une salle à l'aide de paramètres aptes à fournir directement les informations qui intéressent pour la utilisation spécifique de la salle;
- Permettre l'effectuation d'investigations systematiques tendant à mettre en relation la qualité d'écoute avec les caractéristiques physiques et géométriques des salles, fournies par des mesurations objectives (temps de réverbération, rapports acoustiques, cohérence spatiale du champ sonore, etc.)

DEFINITION DE LA METHODE

Pour résoudre les deux ordres de problèmes surexposés on a cherché de définir une méthode qui répond aux conditions suiventes :
A) Elle doit être d'un emploi facile qui ne fait appel à des auditeurs spécialisés;
B) Les résultats qu'elle fournit doivent varier sensiblement avec les paramètres acoustiques qui influencent de plus la qualité de l'écoute, comme p.e. les caractéristiques de la réverbération;
C) Ces résultats doivent être aptes à définir la qualité de l'écoute dans la salle sans demander des élaborations sophistiquées comme c'est le cas, par exemple s'ils peuvent être traduits facilement en termes de rapport signal bruit équivalent.

A ce but on a fixé l'attention sur les méthodes d'évaluation de l'index d'Inteligibilité et en particulier sur le "Rhyme test" qui répond pleinement aux conditions A et C.

Puisque cette méthode dans sa forme originale emploie des mots monosyllabiques elle ne se prête pas à satisfaire aussi la condition B, outre à n'être pas applicable aux langues qui ne disposent pas d'un nombre élevé de mots monosyllabiques. Pourtant il a été nécessaire d'étudier des arrangements conviviaux tels que l'emploi de listes fermées de mots polysyllabiques au but d'évidencer l'effet de masque dû à la réverbération et l'adjonction d'un bruit surposé aux signaux de parole et diffusé par le même haut-parleur.

Ces arrangements sont en cours d'expérimentation dans des salles différentes pour caractéristiques physiques et géométriques, en utilisant toujours la même source sonore et le même niveau d'écoute.

Les résultats déjà obtenus semblent satisfaisants.

Les détails de la méthode, les listes des mots utilisés et les résultats des expérimentations seront recueillis dans la relation qui sera présentée en cours de congrès.
INTRODUCTION

The design of the Great Hall, (principle lecture theatre for the 8th International Congress on Acoustics), is a compromise developed from the inevitably conflicting requirements of the varied uses of the principle hall of an academic institution. Apart from two or three projected occasions in the academic year when choral works are performed, the principle use for the hall is the communication of speech. This determines the reverberation characteristic.

The space available for the hall was deemed to be of secondary importance compared to optimum use of the site for other purposes such as libraries, offices and further lecture theatres. As a consequence the extreme boundaries of the ceiling and upper walls were fixed at the time of the acoustic design. To gain appropriate volume per seat, every attempt was made to incorporate all available space, which necessarily made the hall complex in shape.

DESIGN PROCEDURE

This complexity of shape led to two major design difficulties: It was difficult to assess uniformity of sound field across the audience. Many surface materials had to be battened off from the structural fabric, making assessment of their absorption as a function of frequency difficult. In an attempt to overcome the former, a three dimensional optical model was made in which the acoustic absorption coefficients were simulated by choosing materials of the appropriate reflection coefficient and sticking them to the corresponding surfaces in the model. The audience was simulated by a ground glass plate, and the speaker by a small light source with an omnidirectional light output. The light intensity patterns on the glass plate were examined for uniformity and the model modified accordingly either in the reflection coefficients or the orientation of reflecting surfaces.

COMPLETION ACOUSTICS

As the following table indicates, considerable success was achieved in constructing the hall with the desired reverberation characteristic. These figures are for the empty hall. When full the reverberation times fall by 0.15 seconds.

<table>
<thead>
<tr>
<th>Frequency, Hz</th>
<th>64</th>
<th>128</th>
<th>256</th>
<th>512</th>
<th>1024</th>
<th>2048</th>
</tr>
</thead>
<tbody>
<tr>
<td>R.T. secs, design</td>
<td>2.8</td>
<td>2.1</td>
<td>1.6</td>
<td>1.4</td>
<td>1.4</td>
<td>1.4</td>
</tr>
<tr>
<td>R.T. secs, observed</td>
<td>2.7</td>
<td>1.9</td>
<td>1.5</td>
<td>1.3</td>
<td>1.3</td>
<td>1.3</td>
</tr>
</tbody>
</table>
THE DELIENATION OF ACOUSTICALLY INDEPENDENT SPACES IN LARGE ENCLOSURES

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INTRODUCTION

When the acoustical climate of a typical landscaped office having an absorbent floor and ceiling is analyzed, whether by the examination of attenuation curves (1-2), or by the averaging of the results of several SL recordings (3), then the $L_{10} \text{dB}(A)$ of office activities emerges as the dominant SL.

The background noise levels of these offices are however, strongly influenced by the airconditioning/ventilating system's $L_{10}$, when assuming that most offices today are airconditioned and assuming also that there is no other masking noise introduced into that space.

At a given work-station in such a space, when it is related to the ceiling outlet of the airconditioning/ventilating system, then the importance of the sound field generated by the system is evident.

EXPERIMENT

Earlier studies (T. Vass & H.G. Leventhall, to be published, Acoustic Letters) indicated that there is only a limited number of airconditioning/ventilating ceiling outlets which are capable of influencing, in the absence of any additional intrusive noise, the shape of the sound field in a landscaped office.

Those studies were performed as model experiments (scale 1:10) at 3.15 KHz and in the absence of any obstructions (furniture), produced a "cell" which was 4m in diameter, with the ceiling outlet of the airconditioning/ventilating system in the centre. (Slides 1 to 3).

In each "cell", the SPL at the same test frequency was found to be identical, provided that each outlet had the same PWL, and that each "cell" was acoustically independent of each other until the PWL of an adjoining "cell" increased by more than 5 dB.

Current research dealing with the same hypothesis and testing it also in model experiments as well as in field studies, using the same test frequency, but introducing furniture and absorbent screens into the space (slides 4 to 9), indicate that:

A) When reflective materials, (tables, chairs etc.) introduced into the office space, reach a specific % of that particular "cell" area (slide 10), then to maintain its acoustic independence from the neighbouring "cell", the distance between the airconditioning/ventilating ceiling outlets has to be increased (slide 11).

B) When the reflective condition of the floor as described in A) is present in the office space, but the criterion, that the original, acoustically independent "cell" size is to be retained, then by the placing of absorbent screens at specific locations in the "cell", this can be achieved (slide 12).

CONCLUSIONS

If work-stations in a landscaped office are related to the SL's of ceiling outlets of airconditioning/ventilating systems, then it is suggested that these outlets are delienating acoustically independent "cells" in that space, and that the size of these are strongly influenced by the reflective materials present in these "cells", and/or by the type and location of absorbent screens.

REFERENCES

INTRODUCTION

Beginning a new phase in its development Radio & Television Beograd (Radio: 1,650,000 minutes - 6 Programmes; TV: 364,260 minutes - 2 Programmes) is building the House of Radio and Television in Beograd - Kosutnjak. Its first construction phase is the TV Production Centre with two TV studios (970 and 380 sq.m.) and other facilities totalling 12,500 sq.m. An existing film studio (1,500 sq.m.) is used as a central structure containing both TV studios. Other buildings are erected around it. In final phase the House of Radio and Television Beograd (Architects: Uglješa Bogunović and Slobodan Janjić) shall have some 100,000 sq.m. of useful area.

ACOUSTIC DESIGN

Most problems encountered were caused by the use of the existing film studio and its being divided into 2 TV studios. (New partition whose limiting sound insulation value was determined by the existing flanking walls; the weight of the roof insulation severely limited the existing and new lattice web girders of the roof construction etc).

On the other side there was a strong need to modify well known ways of attaining high sound insulation to the conditions specific for the building practice and possibilities in Yugoslavia. Foreign experiences (Dr. K.H. Zehm, several communications 1971-1975) pointed out, that it is much easier to design complicated high-insulation structures than to put them into practice.

To attain the set goals the existing building is separated from all new buildings surrounding it with the gap starting from the foundations and going all the way up. Similar gaps are also separating new buildings from each other. All sources of noise (boilers, pumps, chillers, fans, transformers etc) are installed in the new buildings.

RESULTS

The paper contains numerous results of the sound and vibration insulation measurements on building structures of the TV Centre. The basic feature of the applied solutions is its simplicity and easy control during construction works. Some details are also given about the applied wide-band sound absorber used for the acoustic cladding in both television studios.

Fig.1 represents an example of the experimental results for the vibration insulation of the "floating" transformer cubicles.
INTRODUCTION

It is a contradiction of rationality that we have poorer documentation of small auditoria than of large, when not only are there many more of the former than of the latter, but smallness does not avoid acoustical troubles. By "small" it is meant here spaces accommodating a range of numbers from 50 to 100 for large committees and small conferences, through recital and lecture halls for 175 to 250 people, to theatres for say 500 to 800 seats. It is a subject area which calls urgently for concentrated acoustical attention.

LARGE COMMITTEE AND SMALL CONFERENCE ROOMS

In the smallest range of size, rooms for multi-language meetings are in one sense less of a problem than for a single language because of the usual use of electro-acoustic aids for the former, but the size and shape of table and of room then become critical, as do the boundary surfaces and the background noise levels. There is no margin for a careless approach. The size and shape of table may now also have political constraints. Guidance for reverberation is mostly speculative.

RECITAL AND LECTURE HALLS

Rooms for upwards of 150 people range in use from lecture theatres to chamber music. The shape requirements begin to be different. Audiences for lectures usually need an unimpeded view, and direct signal, implying a fairly steep rake for seating, which may relate well to the suitability of low volume, short reverberation, and a fan-shaped plan, but for music a smaller slope for seating and a larger volume per seat begin to be desirable. In computing volume it becomes rational to assume that the audience is an absorbent of about 1 m thickness and not to treat it as part of the volume. The fan-shaped plan becomes risky and the shapes around the sound source begin to be important, as does diffusion.

THEATRES

In the range of auditoria for 500 to 800 people we have the typical small theatres for plays, stage musicals, opera and sometimes concert music. There will be an orchestra pit and the ceiling will have to accommodate at least one lighting bridge, with a vertical bank of lights on either side. A fan-plan is so deficient in reflections that quality of sound can be damaged beyond repair. The rectangle, with good reflecting surfaces either side of the stage opening and additional diffusion is good policy. The ceiling may have to be shaped carefully over the orchestra pit to produce useful rather than damaging reflections, and also to prevent the lighting bridge from destroying useful reflection. The seating rake should not be too steep, for the direct signal can become too strong for the limited reverberation, and closing up to a reflecting ceiling can cause excessive loudness at the rear. An orchestra shell is a desirable provision but need not totally enclose.

At 2500 m² the reverberation should be about 1.2 or 1.3 secs at 500 Hz, but it can be worthwhile to consider an assisted resonance installation to increase it for music.
INTRODUCTION

Education, communication, culture (and, perhaps, propaganda) represent strong reasons for new studio developments in developing countries. Studio acoustic design acquires an added dimension of uncertainty if the unknown properties of indigenous materials are compounded with the other inexplicable effects so often found. Three possible solutions to the problems are described.

1. MOBILE STUDIOS

The mobile studio, designed supervised and constructed within the designers' own sphere of operations, represents one possible answer that may serve for small talks or interview studios up to 15m² in floor area. The comparatively lightweight construction of the vehicle while it contributes low frequency absorption also permits external noises to enter the studio. Middle and high frequency absorption can to some extent be integrated with the multiple skins required for sound insulation and will also provide thermal insulation.

2. CONVENTIONAL CONSULTANCY

Through discussions of the specific requirements with the local personnel, the consultant team develops a design defined by a set of architectural and services drawings, a detailed specification and a bill of quantities. Following a tender stage, a contractor is appointed and, with the necessary supervision, constructs the studio(s). Suspended ceilings (with or without acoustic properties) and carpets of various types are now available in most parts of the world but these cannot and should not provide all the acoustic absorption. One is left with the necessity to import materials and one excellent solution draws upon a range of modular absorbers [1]. These are now available in a knock-down form with a consequent saving of space and expense in transport and are suitable for assembly on site with a minimum of supervision.

The absorption characteristics of such absorbers are shown in the Figure.

3. "STRUCTURAL" ABSORPTION

The construction of studios from building blocks which incorporate the required absorption within the structural element is an attractive possibility for inexpensive construction. Concrete blocks containing resonator necks and volumes, structural louvres incorporating fibrous material or waffle slabs faced with micro-perforated surfaces represent possible approaches to achieve such an integrated solution.

SIMULATION OF THE LOCAL ACCOUSTIC FIELD IN A LANDSCAPED OFFICE USING FAST FOURIER TRANSFORMS OF IMPULSIVE SOUNDS

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In an earlier paper (1) the concept of a total model comprising two separate models, one for the background speech and one for the local speech, was discussed. The background model functions adequately with input data from steady state attenuation measurements and can therefore operate in a simple analogue manner. The local speech simulation requires a greater degree of sophistication, since this speech is listened to more intently by the office occupant. The simulation uses the transfer function between the source (S) and the receiver (R) locations of interest. This function may be taken as the dB difference of the FFT's of the measured signals at S and R, the impulsive source being close to S. In addition to the usual precautions taken when performing FFT's of short signals, great care must be taken to avoid non-linearities in response of the various structures which lie in the acoustic paths between S and R. When these do occur it is necessary to use the well known technique of adding low level signal responses.

Using the transfer function, the simulated signal can be produced simply by multiplying the original signal's FFT with the function and performing an IFT. Once again the precautions are applied.

STEREOPHONIC SIMULATION

In (1) the subjective tests were performed monaurally by mixing the digitally reconstructed signal with the background. The results of these tests indicated that subjective performance was much lower than would be expected from experience in real offices of subjective ability to decipher direct signals. This was probably because of the stereophonic character of the real field. A first attempt at introducing stereophonic characteristics used a modification of a device (2) which simply introduces a phase difference between the left and right headphones. Tests with this simulation indicated that a subject's ability to decipher the local signal are much improved. It is interesting to note that the subjects wish to move their heads in an attempt to optimise the signal to noise ratio. This suggests that a variable phase local signal (2) and a fixed stereophonic background would be useful additions to the model.

REFERENCES

INTRODUCTION

It is generally recognised that early decay time (EDT) based on sound energy decay over the first 100 or so milliseconds determines the subjective impression of reverberation and of sound quality in an auditorium. EDT measurements usually show some variation from one seating position to the other causing a variation in subjective quality. EDT has thus become an important parameter in the subjective correlation between physical measurements and subject judgement. The reason for the existence of EDT in auditoria is attributed to the uneven degree of directional distribution of energy in the early reflections. Consequently, steps to ensure more uniform reverberant decay leading to an improvement in the acoustics, entail the provision of properly oriented reflecting surfaces in the hall.

PRESENT EXPERIMENTS

As an alternative approach the present experiments were undertaken to examine the distribution of sound energy in the normal modes of an enclosure during the reverberant decay and its relationship with EDT. A pistol shot was used as the source of excitation in an empty hall with a reverberation time of approximately 2 seconds at 2 kHz. The resulting frequency response shows variation from position to position with a corresponding variation in EDT.

Frequency irregularity per octave in the decaying state is proposed as an index for diffusion in the reverberant field. It is defined as:

$$ F = \frac{L_{\text{max}} - L_{\text{min}}}{f_2 - f_1} \text{(dB/octave)} $$

the summation for $L_{\text{max}}$ and $L_{\text{min}}$ extending over all maxima and minima in an octave band ($f_2 - f_1$). For the two positions (Figs 1 and 2) the decibel difference between the frequency irregularity in consecutive octave bands (1 - 2 kHz and 2 - 4 kHz) is close to 1 for position 1, and 3 for position 2. Position 1 gave a linear decay whereas position 2 gave a noticeable initial part and an EDT different from the reverberation time over the entire 60 dB fall.


Source: Pistol shot
Decay = 50 ms

Fig. 1 Decaying state freq. response in position 1.
Fig. 2 Decaying state freq. response in position 2.
INTRODUCTION

There exists a widespread tradition of using reverberation times (T) in quantifying the quality of auditoria in terms of speech intelligibility. For reverberation curves departing from perfect linearity, the definition of an effective T implies a type of data reduction which does not obviously bear a direct relationship to speech intelligibility.

Recently, an alternative approach has been proposed based on measuring the Modulation Transfer Function (MTF) of an auditorium (ref. 1). This function quantifies the degree of reduction of the modulation depth of a test signal (a noise carrier with 100% intensity modulation), as a function of modulation frequency (Fig. 1). Generally, this function behaves like a low-pass filter which can be characterized by its cut-off frequency (F). For MTF's departing from a perfect first-order low-pass filter, the definition of an effective F implies a specific type of data reduction which, also, the relevance to speech intelligibility is not clear a priori. This contribution considers the potential of T and F in quantifying the speech intelligibility of a sound transmission path.

THEORETICALLY

A reverberation curve and a MTF are related mathematically. Consequently, T and F are not independent. In the limit case of a perfectly linear reverberation curve, the relation is

$$F = \frac{2.2}{T} \quad (1)$$

Generally, however, the relation between F and T is rather weak and a rating of sound transmission paths in terms of F or T may differ considerably.

EXPERIMENTALLY

For a number of (artificial) sound transmission paths three type of data were obtained: (1) intelligibility scores from listening tests, (2) effective reverberation time T and (3) effective cut-off frequency F. The relevance of T or F for speech intelligibility can be judged from the degree to which they rank-order the different conditions in accordance with the intelligibility scores.

REFERENCES


Fig. 1. Illustration of the Modulation Transfer Function. A noise carrier with 100% intensity modulation is used as test signal. The modulation depth m of the received signal is determined and converted to Modulation Transfer MT. This, as a function of modulation frequency, constitutes the MTF.
INTRODUCTION

Monumental churches (as there are many in the Netherlands) from the eleventh and following centuries must withstand the influence of time.

Many of them have been or will be restaurated in the next future.

Since the acoustic circumstances are of main importance for optimal functioning of such buildings, a restauration needs a thorough acoustical accompaniment, concerning both music and speech.

INVESTIGATIONS

Our company has done some thorough investigations concerning the acoustic situation in some two dozens of churches, some already restaurated, some still in their old condition.

The investigations included: reverberation-time measurements, reflectogram-measurements, speech intelligibility, nonsense syllables, musical experiments with organ and brass instruments.

Generally restaurations bring the churchrooms back in a rather bare state, with as a result very long reverberation times, especially in the low frequencies.

FACTORS

Important acoustical factors for restaurations are: materials (like plasters on the walls), diffusion (mostly determined by the original architecture), number of listeners, types of seats, place of the pulpit, reflectors above the pulpit, the use of wooden systems like floors and wainscots (resonating systems, resp. slit resonators).

Because of volume and other influences a minimum of textiles should be introduced (high frequency reverberation).
INTRODUCTION

The basilicas of Acheiropoietos and St. Demetrius are two of the earliest existing monuments of early-christian architecture, both build in the centre of the old town of Thessaloniki.

DESCRIPTION

St. Demetrius basilica was build during the first quarter of the 5th century and has been destroyed twice by fire. The now existing building is the result of a recent total reconstruction. The church is divided in five aisles.

The volume of the church is 22,100 m$^3$ and the total internal surface is 6,500 m$^2$.

Acheiropoietos basilica was build during the second quarter of the 5th century and has never been destroyed or altered. The church is a three aisled basilica of similar construction as St. Demetrius.

The total volume of the church is 19,250 m$^3$ and the internal surface is 5,900 m$^2$.

MEASUREMENTS

Both churches have been measured with the aid of an instrumentation tape recorder and a shotgun, and the tapes were analyzed with standard laboratory equipment.

The reverberation times of both churches (empty) are given in the included diagram (Fig. 1, upper curve St. Demetrius, lower curve Acheiropoietos).

CONCLUSIONS

Both churches are showing much lower than expected reverberation times. The actual calculated absorption coefficients for both churches (> 0,10) are highly improbable, because most of the surfaces are plaster or marble. It is evident that the low reverberation time values, are resulting from the fact that each room total volume is divided in a number of smaller, coupled-together volumes.

The presented results are in agreement with measurements made in basilicas in Rome (1) and with measurements, reported earlier by the author, in an other large-volume monument (2).

REFERENCES

1. Introduction
Measurement in room and architecture acoustics is made awkward by heavy apparatus and is subject to several errors. Errors due to distortion of individual measuring results by sporadically occurring noises are avoided at present by applying more power than is really necessary.

2. Measurement with dynamic signals
By using amplitude modulated signals, e.g., with square-wave amplitude modulated terz-octave band noise (switching on and off in periodic sequence) of frequency F, and using a low-frequency bandpass filter of frequency F in the receiving section (additional filter in the noise level meter) it is possible to separate and differentiate random noises and background noises from the signals used.

The applicability of pulses for measurement in architecture acoustics was demonstrated in theory and practice /1-3/.

An application of amplitude modulated signals for measuring the reverberation time which affords a higher degree of accuracy and which can do with less power was described /4-5/.

3. Measurement with controlled integration
By this method it is possible to measure the reverberation time and the transmission loss directly by means of a controllable noise level integrator.

This method utilizes the fact that in the case of amplitude (square-wave) modulated noises the mean noise level in the transmitting and receiving periods not far from the loudspeaker depends directly on the reverberation time. In addition, measurement over a long period, in each case over the transmitting or receiving periods, makes it possible by comparison to eliminate random, disturbing background noises. Details of the layout of the measuring equipment and its possibilities and limitations are discussed.

4. References
/2/ J. Mantel: "Vereinfachte Methoden zur Messung der Schalldämm-Maße" Kampf dem Lärm (17), pp. 80-81, Juni 1970
/5/ J. Mantel: "Direct and Accurate Methods of Measuring Reverberation Times" Inter-Noise 77, Zurich, March 1-3, 1977
TYPICAL NOISES IN AUDITORIA

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The determination of noise influence on the speech intelligibility in auditoria of different type was initiated in 1976 by our centre as a research project.

The first stage of the work concerned typical noises encountered in cinemas, theatres, opera houses, concert halls and multipurpose halls. Investigations were carried out in 20 different rooms.

The noises inside the hall, which derived from the building, where the hall was situated, and the street traffic and transport, were measured and designated as the "specific noise level".

Furthermore, the noises caused by air-conditioning and ventilating systems, film projection and stage lighting equipment were considered.

As an example, frequency spectra of the specific noise levels for cinemas, theatres and opera houses are shown in Fig. 1.

It has been stated that the minimum levels of the specific noises are almost the same. The maximum levels, however, are different with the highest values for cinemas and the lowest for operas. The outlines of frequency spectra are very similar for these three types of auditoria.

In the next stage of our research work, subjective assessment of the speech intelligibility, when masked by noises discussed above, have been made.

REFERENCES

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CONTROL OF SOUND INSULATION BETWEEN DWELLINGS IN THE UK

Scholes, W.E. Building Research Establishment
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INTRODUCTION

Regulations (1-2) to control the sound insulation of walls and floors between dwellings were first introduced in the UK in the 1960s. They were based on research into noise nuisance at different levels of insulation, eg (3-4). Briefly the present regulations allow the choice of either designing to meet functional/performance requirements or of incorporating into the design specified constructions for party walls or floors together with specified flanking constructions. The performance requirement corresponds approximately to the airborne insulation of a 225mm solid brick wall, plastered both sides, or to the airborne and impact performance of a concrete floor with a floating finish. The mean measured field performance of four examples of the proposed construction is used to determine compliance. The responsibility for enforcement of the regulations lies with the Local Building Inspector.

THE PRESENT POSITION

Recently BRE has been carrying out two investigations aimed at assessing the effectiveness of sound insulation control. The first has involved a survey of sound insulation performance between dwellings constructed since 1971 (5). To date some 2000 individual party walls and floors have been measured. Analysis of the data from the 1270 party walls measured shows that some 55% of the walls tested fail to meet the performance standard. Furthermore some 11% of the walls have very poor performance and their insulation values are not much above those which gave rise to serious and concerted complaints in the 1950s. A separate social survey (Langdon and Buller, in course of publication) has indicated that some 8% of people living in attached dwellings are bothered by noise from their neighbours and that some 30% report hearing their neighbours.

CONTROL IN THE FUTURE

It is considered that the regulations have been successful in that they have prevented a much worse situation than we have at present. There are still complaints about sound insulation, but these are far less numerous or serious than they were in the 1950s. For the immediate future it is considered that the situation can be improved by more scientifically based enforcement of present regulations and by wider publicity of design features known to adversely affect sound insulation performance. For the longer term it is envisaged that regulations will need to take account of variability in performance, as well as of mean performance, so that the occupants of dwellings will have a reasonably high expectation of freedom from annoyance due to noise from their neighbours.

REFERENCES

INTRODUCTION

Building regulations in many countries now contain acoustical requirements, mainly to ensure adequate sound isolation of one dwelling from another attached to it. These regulations are usually based on the transmission loss of the common wall between the two dwellings and not the sound isolation. Even if these regulations were strictly enforced, there is considerable doubt as to whether they would be effective in achieving adequate privacy between dwellings.

OCCUPANT SATISFACTION

Social surveys have been undertaken to determine the level of satisfaction attained using various forms of construction. The surveys completed indicate resident reaction to sound transmission through three thicknesses of concrete floors having sound transmission classes of 48, 51 and 53. There is a significant difference in the responses of occupants to airborne sound transmission for the three floor types. The results indicate that about 60% of occupants of lower socio-economic status are satisfied with a floor having a STC value of 50.

Other surveys are being carried out in "middle class" and "upper middle class" areas to determine the acceptability of various wall and floor constructions and to compare the acoustic privacy attained with that of people living in detached dwellings.

Transmission loss alone is not a good indication of sound isolation. Noise Reduction is a better indicator because it accounts for flanking transmission paths as well as the size of the wall and the receiver room absorption and Schultz's (1) "Privacy Index" which accounts for background noise as well, is even better. There is however, still a strong case for using transmission loss.

SOUND ISOLATION TESTING

If acceptance testing of buildings for acoustic performance is to be undertaken, the method used must be simple, quick, reliable, and require little instrumentation. The method currently being investigated is based on the dB(A) difference single value rating system. In the present method a rapidly swept discrete frequency signal is used. It has been found that there is excellent agreement between the results of transmission loss obtained in this way and with the standard technique. There are a number of advantages of the swept frequency system (and there are further simplifications which can be made), over other systems, for determining the acoustical privacy between two rooms.

CONCLUSION

For a building code to be effective in regulating the sound isolation of dwellings it is essential that the level of user satisfaction be determined and that a more realistic method of specifying or testing constructions be used.

Schultz's method ($\Delta L + L > 75$ where $\Delta L =$ sound level difference in dB(A) between rooms and $L_A =$ background noise level in dB(A)), or a variation on it, e.g. $L_2 \leq L_A$ (where $L_2$ is the sound level in the receiving room from a standard sound source), would be appropriate.

REFERENCE

(1) T.J. Schultz, Internoise '75, 1975, Sendai, Japan.
THE ATTENUATION OF ROAD TRAFFIC NOISE BY BUILDING FACADES

Lawrence, A. University of New South Wales, Kensington, Australia

INTRODUCTION

The most effective long-term solution to the problem of road traffic noise is to reduce the noise emission from individual vehicles, however, a more immediate solution is to improve the attenuation provided by the facades of affected buildings. The occupants of domestic buildings in warm climates usually rely on open windows for ventilation and cooling purposes, which severely limits the theoretical noise reduction that can be obtained.

THEORETICAL BASIS

The attenuation provided by a composite form of construction, e.g. a brick wall including a window, may be readily calculated, provided that the sound transmission loss (STL) of each element is known. When one of the elements is an air path such as an open window, the STL of that element tends to zero and it can be shown that the overall STL cannot exceed 10 dB if the air path area exceeds 10% of the whole.

Generally, the effective attenuation, in dB(A), provided by a facade without air paths depends on the band-spectrum levels of the traffic and on the STL/frequency characteristics of the building elements.

EXPERIMENTAL TECHNIQUE

Simultaneous recordings of traffic noise are made outside and inside typical living rooms and bedrooms with the windows either open or closed. The noise reduction provided by the facade is determined, either by comparing the rms one-third octave band levels sampled over a complete 10 minute recording period, or by selecting individual vehicle peaks (10 dB(A) above the general traffic noise level) and determining the rms one-third octave band levels over the duration of the vehicle pass-by (usually 4 seconds).

RESULTS AND CONCLUSIONS

Figures will be presented showing typical results obtained. Generally, as expected, the performance of the windows dominates the attenuation. With windows open, the average noise reduction is 10 dB(A) as predicted; with windows closed, values of 20 to 30 dB(A) have been measured.

Although most of the sound energy radiated by a traffic stream in urban conditions is in the lower frequency bands, when the A-weighting is applied it is found that the most important components occur in the 250 to 4000 Hz range. With massive forms of construction, such as cavity brickwork or dense concrete, the 'mass law' governs the effective attenuation in dB(A), but in the case of windows and other lightweight construction, coincidence dips in the 1000 to 2000 Hz range limit the dB(A) reduction. Even double glazed windows with significant air spaces are of limited effectiveness unless different glazing thicknesses are used to smooth the coincidence dip.

As predicted, field measurements have shown that the maximum reduction of road traffic noise provided by domestic building facades is limited when windows or doors are open, to approximately 10 dB(A). Alternative means of ventilation and/or cooling are required if greater noise reduction is necessary.
INTRODUCTION.
In this paper we try solve the transmission loss in double walls problem by means the classic approach method.

THEORETICAL BASES.
The treatment of calculation following is based from electrical circuits theory solved by (1) for equal walls.
The walls considered here are different. They have distinct mass, critical frequency, and losses.
We associate impedances to walls and air cavity. The walls are considered of infinite size or dimensions.
How (1) we solve the problem by means a process for reduce the final integrals a simple forms tractables for analytical solution or in the worse cases by planimetric or another methods. We present something formal discrepancies lights to (1) in the solution problem.

RESULTS.
Is deduced from our final expressions that in the curve representing the transmission loss, there is two valleys belonging respectively to critical frequencies of the two walls depending of its losses.
For upper resonance frequencies and for under of the most low critical frequency, corresponding to something wall, the TL line follows: 12 dB/octave inclination law. For above critical frequencies, the TL line oscilate between 12 to 20 dB/octave aproximately.

REFERENCES.
(2) A. LONDON. "Transmission of Reverberant Sound through Double walls". 1950 JASA 22.
INTRODUCTION.

The exact solution is obtained for the problem of the transmission loss of a sound wave through a double partition. We assume that double partition separates two rooms.

To explain this phenomenon we base our calculations on the physical principles of the ondulatory and elasticity theories.

THEORETICAL BASES.

In the problem resolution we make the following assumptions:

a) The panels of the double partition obey the elasticity laws of the rectangular plates.
b) Any another wall of the cavity or the receptor room are considered perfectly rigid.
c) Finite dimensions walls.
d) For simplicity, losses are disregarded.
e) Assuming no flanking.

The elasticity problem of walls is solved by means of the Navier's solution of the rectangular plates (1-2), under the hypothesis that there is a load distribution q(x,y,t) depending periodically with time, produced by pressure differences, and one function s(x,y) representing its distribution along the plate.

The sound propagation in air cavity and receptor room is studied from the wave equation with its geometrical boundary conditions.

RESULTS.

Results are independent by hypothesis, of the generation of sound in the source room and its resonance modes. We considerate only that there is a incident pressure sound generated by any form above double partition.

The final solution involves the resonances modes of each wall and receptor room, the coupling between walls with air cavity and with the receptor room.

In the next time it will be developed a computer program to check this theory.

REFERENCES.

APPAREILLAGE DE MESURE SIMPLIFIEE DE L'ISOLATION, DE L'ABSORPTION ET DE LA PUISSANCE ACOUSTIQUE

Pujolle, J. Telediffusion de France

1-BUT DE L'ETUDE PRESENTEE

Trouver des méthodes de mesure simple et rapide de l'isolation et de la puissance acoustique sur place et en laboratoire, développer l'appareillage permettant d'exécuter facilement ces mesures, faire servir enfin le même appareillage à la mesure du facteur d'absorption en salle réverbérante.

2-BASES THEORIQUES

L'isolation recherché est l'isolation global pondéré A tel qu'il est imposé par la réglementation française actuellement. Et c'est un isolement normalisé sur le chantier. Le facteur de correction 10 log (7/0,5) est atteint non par l'intermédiaire de la durée de réverbération T qui impose un autre appareillage pour être mesuré, mais par l'évaluation de la différence de niveau de puissance Lw et de pression acoustique réverbérée Lp dans le local réception. Ce calcul s'appuie sur la théorie des sources images que nous avons développée dans une série d'articles (1 à 7). C'est aussi à l'aide de la différence Lw - Lp que l'on peut mesurer la puissance acoustique globale et le facteur d'absorption en salle réverbérante spéciale.

3-PRINCIPE DES APPAREILS UTILISES

La source sonore doit avoir un spectre mis en forme pour être soit rose, soit rose avec une pondération type courbe A au point où se trouve le micro- phone de mesure dans le local émission. La source est petite, omnidirectionnelle et peut être positionnée n'importe où, avec grande facilité dans un local. Les niveaux de pressions sont comparés 2 à 2 à l'aide d'un appareil appelé compararophone. Ce dernier comprend essentiellement 2 chaines d'amplification dont les sorties sont opposées après réglage du gain de l'une d'entre elles. La lecture du gain indique la différence des 2 niveaux de pression mesurés.

La source sert aussi comme source de référence de puissance acoustique connue à l'aide d'un microphone en position fixée par rapport à elle. La mesure de la puissance acoustique globale pondérée A d'une source est faite par la méthode de substitution. Source de référence et microphone peuvent être déplacés simultanément de telle sorte que le rapport de l'intensité directe à l'intensité réverbérée reste constant.

Le même dispositif est employé pour la mesure du facteur d'absorption de manière simple et rapide. On y ajoute un filtrage par bandes de tiers d'octave ou d'octave si on veut connaître la variation du facteur d'absorption en fonction de la fréquence.

Les appareils décrits ont fait l'objet de plusieurs brevets.

Références :

(1) J. PUJOLLE - Revue d'acoustique, 18, 21-25 (1972)
(2) J. PUJOLLE - Revue d'acoustique, 19, 107-113 (1972)
(3) J. PUJOLLE - Revue Radiodiffusion-Télévision, 25, 254-259 (1972)
(4) J. PUJOLLE - Revue d'acoustique, 27, 265-267 (1973)
(7) J. PUJOLLE - Revue d'acoustique, 36, 44-50 (1976)
UEBER DIE AKUSTISCHEN PROBLEME BEI DER ERRICHTUNG VON T.V. STUDIO EINS AUF DEM ANDEREN GELEGT

Petrov, K. Bulgarisches Fernsehen und Funkwesen
Sarkov, N.

Der Vortrag gibt eine Projektlösung der neuen Studioskomplexe im Gebäude des bulgarischen Fernsehens, mit aufeinander gelegten Studios, sowie die Ergebnisse der akustischen Messungen.


DER EINFLUSS VON FREIEN WELLEN AUF DIE SCHALLDÄMMUNG
VON WÄNDEN ENDLICHER FLÄCHE (FILM)

Heckl, M.  Institut für Technische Akustik der TU Berlin


Literatur
Cremer, L.  Die wissenschaftlichen Grundlagen der Raumakustik Band III  Hirzel Verlag Leipzig 1948
Heckl, M.  Die Schalldämmung von homogenen Einfachwänden endlicher Größe  Acustica 10 (1960) S. 98

* The oral paper and the film will be given in English.
INTRODUCTION

Under the pressure of incident wave wall is set in vibration, which radiates in another side the transmitted energy. This must be the mechanism of transmission of sound through wall. In this paper the radiation efficiency of rectangular finite wall is calculated and applied to wall transmission.

THEORY

Total radiated power $e$ is expressed with wall velocity and area, $U_w$ and $S_w$, specific impedance of air and radiation efficiency:

$$e = U_w^2 x \rho c x S_w x \sigma'$$

(1)

where, $\sigma'$ radiation efficiency, written in the form of spherical integral, is expressed as follows, in case of rectangular wall which dimension $(2a \times 2b)$:

$$\sigma' = \frac{(\mu a)(\mu b)}{k^2} \int_{W} \int_{X} \left( \frac{\sin X \sin Y}{X Y} \right)^2 x dW W ; \text{ spherical angle}$$

(2)

$k = \omega / c$, and $X$, $Y$ are functions of incident and radiation angles. Radiation efficiency must be calculated numerically but, able to be integrated analytically until half-way in case of axial incidence, the calculation is not so cumbersome even in large dimensions $(ka)$ and $(kb)$.

CALCULATED RESULTS AND CONCLUSION

Fig.1 shows Radiation efficiency, which tendency converges to $1/\cos \theta$, $\theta$ incident angle, when $(ka)$, $(kb)$ become infinity. This corresponds to Abstrahlfaktor by Dr. Gösele but never diverges to infinity even when $\theta = 90$. When $(ka)$, $(kb)$ small, it loses gradually directivity until equidirectional as limite.

Fig.2 shows calculated example of Reduction Index of wall between diffused rooms. (wall mass = 25 kg/m$^2$) Dotted line is conventional Mass-Law. Although neglected the effect of flexible vibrations, its tendencies seem plausible because ordinary reported values of Reduction Index fall nearer to our calculation.

Mr. Rindel in Denmark has introduced our theory kindly in his paper, with his own theoretical and experimental works, recently in English.

REFERENCES

INTRODUCTION

An investigation has been done on the directivity effects in the far free field of a vibrating rectangular wall, at frequencies lower and higher than the critical frequency \( f_\text{cr} \).

Results of measurements on a real wall with unknown boundary conditions have been compared with calculations on two models of a simply supported panel.

THEORETICAL BASES (see fig. 1 for the used symbols)

From RAYLEIGH's formula an expression can be derived for the sound pressure \( \bar{p} \) averaged over a circle (characterised by \( r_0 \) and \( \theta \)) in the far free field of a rectangular wall vibrating with a single frequency:

\[
\bar{p}(r_0, \theta) = j \omega \frac{\exp(-j kr_0)}{r_0} \int_0^{2\pi} \tilde{v}(r_1) \frac{j (kr_1 \sin \theta) r_1}{r_0} \, dr_1,
\]

where \( \tilde{v}(r_1) \) is the normal velocity averaged over a circle with radius \( r_1 \) in plane \( S \) which comprises the wall surface. If the wall is mounted in a heavy frame where \( v_p = 0 \), \( v_p \) has to be measured or calculated only for circles comprising part of the wall surface and integration has only to be performed from 0 to \( r_1 \).

For a simply supported panel, \( \tilde{v} \) can be calculated from the following equation:

\[
v(\phi, r_2) = \tilde{v} \sin \frac{n\pi}{a} (r_2 \cos \phi - 0.5a) \sin \frac{n\pi}{b} (r_2 \sin \phi - 0.5b).
\]

MAIDANIK [1] gives a model of a simply supported panel in which the panel is divided into phase cells, which for some frequencies are coupled and thus partly cancel out by interference. From this model we also can calculate \( \bar{p}(r_0, \theta) \).

EXPERIMENTAL TECHNIQUES

Measurements have been done on a 7 cm thick gypsum wall, mounted in a rigid frame between a reverberation chamber and an anechoic chamber. The wall was excited by a pure tone driven shaker or by a pure tone reverberant sound field and \( \tilde{v}_n \) was measured by means of a velocity sensitive microphone rotated around the center of the wall, just free from its surface. Calculations according to eqs. (1) and (2) and to MAIDANIK's approach have been carried out on a HP2100 and an IBM1130 computer.

RESULTS AND CONCLUSIONS

For \( f > f_\text{cr} \), the results of eqs. (1) and (2) have a good agreement and show a rather sharp directivity lob according to the coincidence effect. For \( f < f_\text{cr} \), discrepancies have been found between the results of the three methods, which partly can be explained by the inhomogeneity of the gypsum wall and its undefined boundary conditions.

REFERENCES

INTRODUCTION

Nous présentons une méthode de calcul de la transparence acoustique $\mathcal{T}$ des plaques orthotropes multicouches que nous avons étudiées en référence (1). Une étude numérique montre ensuite l'influence des différents paramètres ; on détermine une plaque légère à haut pouvoir isolant. Les expériences de vérifications sont conduites en chambres anéchoïques.

THEORIE

Notre étude des vibrations des plaques multicouches rectangulaires, appuyées, a montré l'existence de familles de modes propres, c'est à dire de groupements de modes de fréquences propres différentes, présentant des déformées modales identiques dans le plan de la plaque; on a les relations (pour les modes de la famille $(n,m)$):

$$ k_x = k \sin \varphi = \frac{n\pi}{l} \quad k_y = k \sin \varphi = \frac{m\pi}{b} $$

$k$: nombre d'onde, $\varphi$: angle du plan d'onde

La transparence de la plaque qui s'obtient par sommation sur tous les modes peut se grouper par famille :

$$ \mathcal{T} = \sum \sum (\sum \mathcal{T}_{nm}) = \sum \sum \mathcal{T}_{nm} $$

Approchons (2) par une double intégration :

$$ \mathcal{T} \approx \int \int \mathcal{T}_{nm} \, dn \, dm $$

Cette opération revient à considérer une transparence moyenne de la famille $(n,m)$ sur le domaine $[m-\frac{1}{2}, m+\frac{1}{2}] \times [n-\frac{1}{2}, n+\frac{1}{2}]$

En changeant de variable dans (3); $(n,m) \rightarrow (k, \varphi)$ il vient :

$$ \mathcal{T} \approx \int \int \mathcal{T}_{k\varphi} \, dk \, d\varphi $$

Cette expression (4) a l'avantage de ne plus nécessiter la connaissance des fréquences propres de vibrations.

RESULTATS-CONCLUSION

Le calcul de (4) a été effectué dans le cas d'une onde incidente plane et oblique. Nous montrons l'importance du cisaillement transversal pour les plaques anisotropes et le peu d'influence de l'amortissement interne des couches molles de la plaque. Nous mettons aussi en évidence une plaque multicouche légère, à haut pouvoir isolant, que l'utilisation des matériaux composites fibrés permet de réaliser. L'expérimentation, conduite en chambres anéchoïques, concorde correctement avec les prédictions théoriques.

REFERENCE

UEBER DEN LUFTSCHALLSCHUTZ BEI RAUMLICHE FERTIGBAUTEILEN

Focsa, V.  
Veres, A.  
Biborosch, L.  
Poppel, M.

EINLEITUNG

Über die moderne Mess- und Bewertungsverfahren des Luftschalldämmass- ses bei Wänden und Decken hatte, im Rahmen des 8-ten ICA-Kongresses, L. Cremer einen zusammenfassenden Bericht vorgetragen (1).

In enger Übereinstimmung mit den daraus folgenden Rückschlüssen die besonders die Erfassung räumlicher Bedingungen betreffen, stehen auch vorliegende an Fertigzimmern gewonnenen Schalldämmasswerten \( D_n \).

EXPERIMENTELLE ERGEBNISSE UND DISKUSSION

Die Messungen wurden, gemäß ISO-Recommendation 140, nach den übli- chen Verfahren durchgeführt. Wie teilweise aus Abb. 1 ersichtlich, lagen die zimmergrosse Fertigräume über- und nebeneinander beinahe vollständig durch eine Luftzwischenschicht von 2 bis 4 cm getrennt. Obwohl die Gesamtdicke der schweren \( g = 2400 \text{ kg/m}^2 \) Stahlbetonwände etwa 14 cm betrug, sind die ermittelten Luftschalldämmasswerten \( D_n \) (ausgezogene Kurve) wesentlich kleiner als die zumutbaren Soll- werten (dicke Kurve).

Überraschenderweise, sind sogar die für einer Ähnlichen, ebenfalls 14 cm dicken aber einschal- ligen Betonwand, erhalenenen \( D_n \)-Werten für 3 - 4 dB besser (gebrochen Kurve).

Ähnlich schlechte, die Soll- kurve weit unterschreitende Mess- ergebnisse wurden auch für die "zweischaligen" Decken erhalten.

SCHLUSSFOLGERUNGEN


Von dem bekannten Unterschied zwischen direkten (In Abb. 1 Weg Dd) und durch den flankierenden "Bauteilen" (Weg Pp) oder Eck statt- findende Schallübertragung kann kaum die Rede sein. Vielmehr konnte eine sehr grosse Trennfläche zwischen den untersuchten Fertigräumen angenommen werden, was eine derartige Abnahme der \( D_n \)-Werte erklären wird. Anderseits, wenn man den Korrekturterm zur Massengesetz welcher die innere Dämpfung berücksichtigt betrachtet so kommt man wegen der vermutlichen langen Nachhallzeit der Biegelwellen in der räumlichen Struktur, zu einer ähnlichen Abnahme des Luftschalldämmmasses.

LITERATUR

In situ measurements are essential for achieving the noise control asked for. The following is restricted to airborne sound insulation between rooms in adjoining dwellings (apartments).

The sound insulation really achieved will usually deviate from that mentioned in catalogues or books for the walls and floors used in a specific case. Apart from sound leaks, this is due to flanking transmission (sound energy travels from one room to an adjoining one not only through the party wall or -floor, but also along "flanking constructions") and to energy loss (the party wall or -floor and the flanking constructions do not transmit sound energy from one room to an adjoining one only, but part of the energy is distributed to other building construction elements). Both effects work in opposite directions.

These effects can, of course, be taken into account during the design stage, but in view of the bad experience in several countries with actually achieved sound insulation, control measurements are very desirable. If used in combination with effective enforcement measures one can hope that in the future designers will make use of the existing knowledge in a more efficient way.

Control measurements should preferably be as less expensive as possible. The simpler the formulation of the requirements, the simpler, and therefore the cheaper the control measurements. In The Netherlands, airborne sound insulation requirements are expressed in 5 octave bands. Compared with the use of 1/3 rd octave band we economize a factor 3 in the number of measuring frequencies. Moreover the use of octave bands guaranties a more homogeneous sound field so that less microphone positions are necessary. Even so, however, it turns out that, everything taken into account, a saving in costs of about 25% to 30% is possible if a proposed screening method is used, as follows. Pink noise is used (great precision not necessary) and the difference between the measured overall (lin.) level in the transmitting room and the measured A-level (corrected for reverberation time) in the receiving room is taken as an indication for the airborne sound insulation. The reverberation time is measured or through the A-filter or through a 500 Hz octave band filter; this has still to be decided, the latter method gives slightly more accurate results. Compared with the standardized method, using the screening method means a loss in accuracy of about 1,5 to 1,7 dB, which may be acceptable for this type of measurements.
INTRODUCTION

Although many studies of noise disturbance in office buildings have been carried out much of the information produced suffers from two important weaknesses. First, the assessment of people's attitudes has generally been based on the use of questionnaires, a technique whose validity is in some doubt; and second the measurements of noise level are carried out so as to describe conditions in the office rather than the conditions experienced by one person in an office may differ substantially from that experienced by another. These considerations led to the adoption of a different approach in this project.

THEORETICAL BASIS

The general aims of the study was to construct for each person in an office, an assessment of his attitude to the acoustic environment using a range of psychological techniques and to measure as accurately as was reasonably possible, the nature of the individuals' acoustic environment. This technique takes advantage of the differences that occur within an office rather than ignoring it.

EXPERIMENTAL TECHNIQUE

(a) User reaction. This was assessed by three different methods. In addition to a questionnaire containing the usual range of multi-choice questions, single open ended questions were distributed separately from the questionnaire and focussed interviews were carried out.

(b) Acoustic Environment. Each office was observed for 2-3 days to identify acoustically similar working zones, that is, areas in which each person experiences a similar acoustic environment. The noise climate was then recorded for each zone over a representative period (2-4 days, depending on the variability of the situation). These recordings were analysed to produce various standard descriptions such as Leq and in addition the nature and frequency of major sources of distraction.

RESULTS AND CONCLUSIONS

It was found that the multidimensional approach to subjective reactions produced information on individual attitudes to noise environments which was more detailed than that by questionnaire alone and whose validity could be tested when these subjective results were related to description of the physical environment. The results demonstrated that statistical descriptions of sound levels are too general; people tend to be annoyed by particular identifiable intruding sources. It was found that different buildings and furniture layouts could be rated in terms of their potential for controlling noise from such sources of noise distraction. Those buildings would seem to confirm the need for detailed rather than general studies of noise disturbance.

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THE BEHAVIOUR OF PULSED SOUND IN OPEN PLAN SPACES

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INTRODUCTION

Most studies of the acoustics of open plan spaces have been carried out with sources of continuous sound. This approach suffers from a number of disadvantages: the sources are not typical of many distracting sounds in offices e.g., speech, they create a steady state not a transient acoustic situation, and they do not allow the composition of the sound field to be determined.

In response to these limitations, the authors have developed a method for analysing the transmission patterns of pulse sound in buildings.

EXPERIMENTAL TECHNIQUES

The method used is to generate a gunshot in a room by means of a device which provides a free field combination zone, and to record the subsequent train of pulse through a series of microphones, onto a tape recorder, and to replay the result through a Digital Event Recorder. A sound level/time trace is then produced which allows individual reflections to be identified and the contribution each makes to the final integrated energy to be estimated.

RESULTS AND CONCLUSIONS

The technique has been used to study the effect of different types of ceiling, floor finish and furniture types and layout on the propagation of pulse sound through a number of open plan offices.

The results indicate that (a) the transmission of pulse sound is different from the transmission of continuous sound, due to non linear effects in the building materials; (b) simple multiple image theory of sound transmission through open office does not hold in practice; (c) the design and layout of furniture is more important than has been suggested in some previous studies.

This work is supported by the Science Research Council Grant No. B/RG/85965.
A room with rigid reflecting walls loses sound energy by radiation if it has an open window. The decay rate of sound energy defines implicitly a reverberation time and hence, also, an equivalent absorption coefficient. It is an important parameter in the acoustical design of lecture halls, conference rooms, etc whenever the environmental conditions permit open windows.

We make a normal mode analysis of a rectangular room with an open window, Fig. 1. The sound pressure \( P_n^m(x,y,z) e^{i\omega t} \) of the \( N\)th eigenmode satisfies the boundary value problem

\[
\nabla^2 P_n^m + k_N^2 P_n^m = 0, \quad k_N = \omega / c \tag{1a}
\]

\[
\frac{\partial P_n^m}{\partial n} = 0 \quad \text{on} \ S \tag{1b}
\]

Here \( N \) is the triplet of integers \( n_x, n_y, n_z = 0,1,2,\ldots \), \( \omega_N \) the corresponding eigen-frequency, \( Z_N(x,y,z,\omega_N) \) the input specific acoustic impedance of the window and \( \rho c \) the characteristic impedance of air. We consider eq(1) as a perturbed version of the boundary value problem of an identical room but without the window:

\[
\nabla^2 P_n^m + k_N^2 P_n^m = 0 \tag{2a}
\]

\[
\frac{\partial P_n^m}{\partial n} = 0 \quad \text{on} \ S + W \tag{2b}
\]

A boundary perturbation technique gives the eigen-values of (1) to 1st order as

\[
k_N^m = j \alpha_N \pm \sqrt{k_N^2 - \alpha_N^2} \tag{3}
\]

where

\[
\alpha_N = (\rho c/2\gamma_N) \int W P_N^2 / Z_N \, ds \tag{4}
\]

\( \gamma_N \) being the normalization constant of \( P_N \). The complex nature of the eigen values in (3) shows that the sound pressure of the \( N \)th mode decays as \( e^{-\alpha_N t} \). Eq.(4) shows the dependence of the decay rate on the mode and position of the window. By an averaging procedure the decay rates of sound energy is calculated for particular groups of modes.
EXCITATION OF REVERBERATION ROOMS BY ROTATING SOUND SIGNALS

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INTRODUCTION

Measurements in reverberation rooms below the critical frequency have an increased uncertainty because of insufficient diffusivity. A lot of efforts are made to improve the results by special room constructions, fixed diffusers or rotating vanes. The large surfaces required in the low frequency region for fixed or rotating diffusers are a disadvantage and cause mechanical problems.

Another means of influencing the sound field in reverberation rooms would be the sound source. At the PTB it has now been tried to generate a sound field rotating in the room by means of four loudspeakers placed on fixed locations. The signal of each speaker is amplitude-modulated by a triangle wave with a phase shift of about 90 degrees between the individual speakers. In this way the total signal energy emitted into the room remains constant whereas the virtual centre of the sound source is circulating.

EXPERIMENTAL TECHNIQUES AND RESULTS

Tichy and Baade (1) have shown that a signal radiated from a steady, single-frequency source will be transformed by a rotating vane into a frequency-modulated sound field because of the continuous change of the image-source configuration. A similar effect may be expected for the moving signal. The sound pressure generated in this way in a 237 m² reverberation room was received by a microphone and small band-analysed. Indeed sidebands appear at the expected frequencies. It was then tried to investigate the spatial distribution of the sound pressure level in the same room for a rotating sinusoidal signal. For this purpose, 20 points separated by more than λ/2 were selected. In a first set of experiments the SPL at these points was compared for stationary and rotating excitation of the room. Fig. 1 shows the results in terms of the coefficient of variation s/L (s=standard deviation, L=mean of SPL) as a function of frequency. It can clearly be seen that s/L for the rotating signal is always smaller than in case of stationary excitation.

Further experiments concern the reproducibility of the procedure and the development of a source array with incoherent signals for use in measurements of absorption coefficients.

REFERENCES

Le problème principal de l'acoustique des salles est la prévision des caractéristiques à donner à une salle pour que son acoustique soit adaptée au mieux à sa destination. En particulier, la prévision de l'intelligibilité est réalisée de manière assez satisfaisante à l'aide de la méthode des rayons ou des sources-images, appliquée dans le cadre de l'acoustique géométrique (où l'on néglige le caractère ondulatoire).

Dans le but de chercher une évaluation plus précise du champ sonore, notamment les variations de la pression en fonction de la fréquence ou de la position de la source et du récepteur, nous avons entrepris de tenir compte de l'aspect ondulatoire. Et comme la méthode modale classique ne s'applique qu'à certaines formes géométriques bien particulières, c'est dans la méthode des images que nous introduisons cet aspect ondulatoire ; cela nous permettra par la suite d'étudier toutes sortes d'enceintes. Le présent exposé se propose de montrer dans quelle mesure et avec quelles restrictions la méthode ondulatoire des images est licite et possible.

Le problème que cette méthode permet de traiter est celui assez général de la propagation en milieu fluide parfait avec conditions aux limites locales. Ce milieu limité (pas forcément fermé) est excité par une source ponctuelle délivrant un régime sinusoidal permanent.

La pression au récepteur est la somme des pressions émises par chaque image de la source (y compris la source), chacune étant affectée d'un facteur de réflexion complexe.

Nous établissons analytiquement que :
1) Quand le milieu de propagation a pour limite un plan, ou deux plans parallèles, ou 4 ou 6 plans formant une enceinte rectangulaire ou parallélépipédique, l'expression de la pression ainsi obtenue peut vérifier le système d'équations de propagation, de manière approchée et dans certaines conditions restrictives.
2) Quand la limite est constituée de 2 plans parallèles, toujours dans certaines conditions restrictives, l'analyse fait apparaître les fréquences propres, les nœuds et les ventres de vibration déduits par ailleurs de la théorie modale.

Nous comparons la méthode des images à la méthode modale sur le plan des résultats :
Les deux méthodes sont appliquées à une enceinte parallélépipédique. Elles sont numérisées et leurs résultats sont comparés entre eux.
Des expériences sont effectuées, soit dans une salle de dimensions ordinaires, soit sur maquette. Chacun des deux calculs est confronté aux mesures expérimentales. On étudie ainsi la variation de l'intensité avec la fréquence, ou avec la position du récepteur, pour différentes conditions d'impédance sur les parois.
Dans le cas d'un guide (2 plans parallèles, ou bien tube rectangulaire), on sait que la méthode modale décrit la réalité de manière fort satisfaisante. Nous nous limitons donc cette fois à la comparaison des 2 méthodes théoriques entre elles.
GROUNDBORNE NOISE AND VIBRATION FROM UNDERGROUND RAIL SYSTEMS

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INTRODUCTION

In modern transit systems, structureborne noise due to the transmission of vibration through the ground is a major source of environmental impact. This paper summarizes a method (Kurzweil and Lotz, Handbook of Noise Control, 2nd edit., C.Harris-editor, Chap.33, to be published), for predicting A-weighted sound levels as well as noise and vibration spectra due to ground-transmitted vibration in buildings near subways.

DISCUSSION

A simpler relation for estimating the A-weighted sound level, \( L_{AR} \), in cellar rooms between 1 and 20 m from a subway wall is \([1]\):

\[
L_{AR} = 59 - 20 \log R + 10 \text{ dBA}
\]

where \( R \) is the distance in meters from the tunnel wall to the building. The data upon which eq.(1) is based represents a range of vehicle types, conditions, and speeds; track types and conditions; tunnel and building constructions; and soil types. A procedure for predicting groundborne noise, which accounts for some of the above parameters, is represented by eqs. (2) and (3). First the maximum octave-band vibration acceleration levels, \( L_a \), in a room of a nearby building are estimated using eq. (2), then the resulting octave-band sound pressure levels in the room, \( L_{pR} \), are predicted:

\[
L_{AR} (\text{dB re } 10^{-6} \text{ g rms}) = L_a(\text{tunnel wall}) - C_g - C_{cl} - C_b
\]

\[
L_{pR} (\text{dB re } 20 \text{ Pa}) = L_{AR} - 20 \log f + 37
\]

where \( L_a(\text{tunnel wall}) \) is the maximum octave-band acceleration level (in dB re \( 10^{-6} \text{ g rms} \)) on the subway wall during a train pass-by, \( C_g \) is the vibration attenuation due to propagation through the ground, \( C_{cl} \) is the vibration attenuation (coupling loss) between the ground and the building, \( C_b \) is the vibration attenuation due to propagation in the building, and \( f \) is the octave-band center frequency (in Hz). For groundborne noise, the octave bands from 31.5 to 250 Hz are important.

The tunnel wall vibration spectra from a number a concrete, earth-based (as opposed to rock-based) tunnels with trains traveling at 60 km/h on continuous welded rail have been compiled. The effects, on tunnel wall vibration spectra, of train speed, axle load, vehicle suspension, resilient wheels, wheel flats, rail joints, rail corrugation, resilient rail fasteners, floating slab track, ballast mats, tunnel construction type, and supporting soil type (soil or rock) are discussed.

Typical values for the ground vibration attenuation factor, \( C_g \), are given as a function of frequency and distance for an "average" soil and for rock. The ground to building coupling loss, \( C_{cl} \), is essentially zero for lightweight frame buildings and slab foundation on grade; while for heavy masonry buildings on spread footings or piles it varies between 10 and 20 dB. Finally, the vibration attenuation within a building, \( C_b \), varies from about 4 dB per floor for heavy (masonry) buildings to zero for lightweight buildings.

Eq. (3) provides a reasonable estimate of the sound levels in a room resulting from vibration of the floor and walls but does not explicitly account for changes in sound level due to changes in the room's absorption characteristics.

REFERENCES

INTRODUCTION

La méthode d'analyse statistique par l'énergie se prête bien au calcul de la transmission du bruit dans les bâtiments. Une structure compliquée est alors représentée par des lots de résonateurs simples. Dans le cas d'un bâtiment chaque lot correspond soit à une paroi, soit à un local. En écrivant l'équilibre énergétique pour chaque lot on obtient un système d'équation linéaire traduisant l'équilibre énergétique entre les différents lots.

Ce système d'équation s'écrit :

$$\frac{n_i}{\omega} = (n_i + \sum_{j \neq i} n_{ij}) E_i - \sum_{j \neq i} n_{ji} E_j$$

Les $n_i$ sont représentatifs de la dissipation d'énergie dans les lots $i$, les $n_{ij}$ des échanges entre lots et $E_i, E_j$ des énergies.

La résolution de ce système, facilement réalisée sur ordinateur (1), permet de prévoir les isolalements entre locaux d'un bâtiment. Cependant, en pratique, la difficulté est de connaître les valeurs de $n_i$ et $n_{ji}$ à entrer dans le calcul.

Nous présentons ici la méthode utilisée pour mesurer en laboratoire les facteurs $n_{ji}$ sur des jonctions de paroi en Té.

METHODE DE MESURE

Le laboratoire est constitué par une salle ouverte sur une de ses grandes faces. Cette salle est fermée par une paroi et divisée en deux par une seconde paroi. Ces deux parois forment ainsi la jonction en Té étudiée. Toutes les parois ne faisant pas partie de la jonction sont doublées.

L'expérimentation consiste à :

- mesurer les facteurs de pertes apparents des parois de la jonction et les durées de réverbération des locaux.
- exciter successivement les parois de la jonction en mesurant dans chaque cas la vitesse quadratique moyenne des parois et le niveau de pression sonore dans les locaux.

On peut ainsi écrire des équations d'équilibre indépendantes où les inconnues sont les paramètres $n_{ji}$ cherchés.

RESULTATS ET CONCLUSIONS

Cette méthode a été appliquée pour une jonction constituée d'une paroi lourde de 18 cm d'épaisseur et d'une paroi latérale en carreaux de plâtre de 7 cm d'épaisseur.

En portant des valeurs mesurées $n_i$ et $n_{ji}$ on a calculé l'isolement aux bruits aériens entre les deux locaux, puis cet isolement a été mesuré suivant la méthode classique. La comparaison des résultats montre une concordance satisfaisante entre les valeurs mesurées et calculées.

Les premiers résultats obtenus sont encourageants mais il est nécessaire d'étudier d'autres jonctions pour les confirmer.

REFERENCES

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FIELD MEASUREMENT OF STRUCTURE-BORNE SOUND IN BUILDING

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INTRODUCTION
Structure-borne sound propagation is one of the most important problems in building acoustics, but there is not yet sufficient knowledge to predict its behavior when preparing the acoustical design of a building.

Fortunately we had an opportunity to measure its propagation in a ten storied steel frame reinforced concrete building. This building was not yet completed and had no finishing on the floor, wall or ceiling. Therefore it was possible to excite vibration or to place vibration pick up at any point we chose.

EXPERIMENT
Being a hotel, the building had many of the same rooms from the 3rd floor to the 10th. The vibration excitation point was the center of the floor slab on the 4th floor. Measuring points were the center of the floor and wall, distributed vertically and diagonally on every floor up to the 10th and horizontally on the 4th floor as shown in Fig.-1.

The vibration machines were a wooden hammer and a moving-coil type exciter. When the moving-coil type exciter was used, octave band noise modulated by M-sequence signal and cross-correlation method were used experimentally to obtain higher S/N ratio. When hammering was used as an impact vibration source, the output voltage of vibration pick up was squared and integrated to obtain a result equivalent to steady state distribution.

RESULTS
The results of the measurement are shown in Fig.-2 and 3, and the conclusions are as follows.

(1) Reduction of the vibration at the same distance from the source is higher in high frequencies than in low frequencies.

(2) The reduction of vibration in a vertical direction is about 1.5dB/floor at 63Hz and 4.5dB/floor at 2kHz.

(3) The reduction of vibration in a horizontal direction is nearly equal to reduction in a vertical direction at the same distance from the source.
INTRODUCTION

The sound pressure levels in accommodation spaces on ships are mainly determined by structureborne sound excited by propeller, main- and auxiliary engines. In this paper the propagation of structureborne sound in superstructures is discussed.

MODEL

Two cases have been considered: The power flow in the vertical sections is in the form of (i) flexural and (ii) longitudinal waves. In the first case it is assumed that the displacements of the junctions are negligible. The coupling between the plates is due to rotation of the junctions. The resulting bending moment acting on a plate element can be written as:

\[ M_{yn}(0, y) = \gamma_n(O) F_1(\kappa_n L_x) + \gamma_n(L_x) F_2(\kappa_n L_x) \cdot g(y) \]

where \( \kappa_n \) is the wave number for flexural waves. See Fig. 1. The equations of equilibrium of bending moments at each junction result in a system of equations from which the ratios \( \gamma_n/\gamma_1 \) can be solved. The velocity level differences between the decks can thereafter immediately be calculated. A similar method can also be used in order to describe the second model (longitudinal waves).

RESULTS

Measurements have been made on two tankers (130000 tdw). In Fig. 2 the measured (○○○) and calculated (- - -) velocity level differences between the decks 1 and 3 in one ship are shown. The calculations are based on the flexural wave model.

Measurements and calculations indicate that the attenuation of structureborne sound is a function of frequency and also depends on material parameters and dimensions. A model in which only flexural waves are considered gives very satisfactory results. In the second case, using a longitudinal wave model, the velocity level differences become far too small.
A SIMPLE FORMULA FOR ESTIMATING VIBRATIONAL NOISE LEVEL IN BUILDINGS

INTRODUCTION

Calculation of a noise level produced by vibratory movements of a wall, ceiling and particularly of a floor appears to be subject to considerable simplification. Our subjective is to calculate the noise level in a room distant or next to the vibrating partition. The calculation yields to approximate results; however, a quick and a rough estimate of the noise level is often necessary in many practical considerations.

ASSUMPTIONS AND SIMPLIFICATIONS TO THE FORMULA

The sound pressure \( p \) in a room may be calculated from the radiated power of vibrating surfaces \( S_i \), their velocities \( v_i \) and radiation factors \( r_i \), and from the total absorption \( A \) of the room:

\[
\frac{p^2}{2} = \frac{(2p_0)^2 \sum v_i^2 r_i}{A} \quad (1)
\]

Assume a rectangular room with all partitions alike, so that each \( S_i \) may be substituted by an average surface \( S = \frac{\sum S_i}{6} \), and each \( r_i \) approximated by a constant \( r \). Thus we may place \( S \) and \( r \) out of the summation. Further, in massive and rigid structures, the velocity level decreases approximately by \( N \) dB per room (in horizontal or vertical direction). We shall take the floor to be the vibrational source. Therefore, the floors in the next storey will have a velocity level attenuated by \( N \) dB. We may further assume that the surrounding walls would vibrate with a velocity by \( N/2 \) less than the floor. In this way the sum of the squared velocities of all partitions of the room on the \( n \)-th storey above the source floor will be:

\[
\sum v_i^2 = v_0^2 10^{-N/10} (1 + 4 \cdot 10^{-N/20} + 10^{-N/10}) = v_0^2 \cdot z^2 \quad (2)
\]

where \( v_0 \) = velocity of the source floor. Eq. (1) is thus simplified to:

\[
\frac{p^2}{2} = \frac{(2p_0)^2 \cdot v_0^2 \cdot z^2 \cdot r}{A} \quad (3)
\]

Approximately, for massive and rigid structures \( N = 7 \) dB, i.e. \( z^2 = 0.2^2 \cdot 2.98 \) and for usually equipped rooms \( S/A = 1 \). Also, the radiation factor may be approximated by 1. Finally, after introducing the value for \( p_0 \) in Eq. (3), the sound pressure level is obtained as:

\[
L_p = L_v - 7 \cdot n - 2,7 \quad (4)
\]

where: \( L_p \) = sound pressure level (in dB re 20 \( \mu \)Pa) in the room at the \( n \)-th storey above the source room, and \( L_v \) = velocity level (in dB re \( 10^{-6} \) m/s) of the floor in the source room.

CONCLUSION

At a certain room distance the noise produced by vibratory movements of a floor in a building may be approximately calculated by a very simple formula. The formula, however, should be used with care and only when the above assumptions are fulfilled and approximations checked. Modern light structure buildings are far from following the linear law of attenuation, they often exert resonances along their height.
VALIDITE D'UNE NOUVELLE METHODE DE TEST DE LA SONORITE DES PLANCHERS, AUX IMPACTS

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INTRODUCTION

A l'occasion de Inter-Noise 75, T.J. SCHULTZ (1) a proposé l'utilisation d'une nouvelle machine et d'une nouvelle méthode de mesure pour caractériser la sonorité des planchers aux impacts. La machine n'a qu'un seul marteau, plus léger que ceux de la machine normalisée, dont la tête est munie d'un revêtement élastique. Au lieu du niveau de la pression acoustique moyenne, c'est le niveau de crête qui est mesuré. La méthode ayant paru intéressante, le Ministère de l'Equipement et du Logement a chargé le C.S.T.B. d'en vérifier le bienfondé et de la mettre à l'épreuve.

CONTENU DE L'ETUDE

Nous avons vérifié le bienfondé de la machine en mesurant l'impédance mécanique de marcheurs, en début d'impact contre le sol, et en la comparant à celle du système proposé : masse de 200 g avec tête élastique de rigidité 7,1.10^6 N/m. Le contrôle a porté aussi sur la composante verticale de la vitesse : 0,55 m/s. L'impédance a été obtenue à partir de la mesure de la force appliquée au sol, au cours de l'impact. La vitesse a été mesurée par un procédé photographique. Par ailleurs, nous avons modifié une machine normalisée pour la rendre conforme à la proposition et nous avons comparé les niveaux de bruit produits par cette machine sous deux types de planchers couverts de différents revêtements de sol à ceux produits par des marcheurs réels.

RESULTATS ET CONCLUSIONS

Les mesures effectuées au cours de la marche de 7 femmes chaussées de chaussures à talons hauts montrent que, jusqu'à 1,5 kHz (domaine de validité de nos mesures), l'impédance des marcheurs, au moment de l'impact du talon de leurs chaussures, est bien celle d'un système masse-ressort. Toutefois, en moyenne, la masse est environ la moitié (110 g) de celle proposée, avec un écart type de 60 g. La rigidité du ressort est, elle, très variable d'un marcheur à l'autre (écart type de 1,9.10^6 N/m), alors que la valeur moyenne (1,4.10^6 N/m) est inférieure à la valeur proposée. La composante verticale de la vitesse d'impact est aussi plus faible que celle proposée : 0,30 m/s au lieu de 0,55 m/s avec un écart type de 8 cm/s. En moyenne les caractéristiques de la machine proposée semblent donc être trop sévères pour bien représenter la marche de femmes. Cette trop grande sévérité a été confirmée par les mesures des niveaux de bruit qui, sous un plancher nu ou couvert par un revêtement de sol d'efficacité moyenne, sont plus élevés avec la machine qu'avec les marcheurs.

Il semble donc que la machine proposée se situe en position intermédiaire entre la machine à chocs normalisée et la marche réelle de femmes équipées de chaussures à talons hauts, considérées comme étant les marcheurs les plus sonores. Pour la faire correspondre mieux à la réalité, il y aurait lieu d'en modifier les caractéristiques pour les amener à être proches de celles mesurées.

REFERENCES

(1) T.J. SCHULTZ : A proposed new method for impact noise tests Inter-Noise 75
INTRODUCTION

This paper describes an ongoing study into the transmission of sound between two rooms separated by a concrete wall but otherwise connected by a ventilation duct. The investigation examines the problem of flanking transmission along the duct but does not consider the airborne path via grilles etc. The preliminary results of experiments carried out under no flow conditions have already been reported (ref. 1). This work has been extended to examine the effect of air flow in the duct.

ANALYSIS

Statistical Energy Analysis was used for this work as far as possible and good agreement has so far been obtained between the measured and theoretical results except at the fundamental cross mode of the duct where modal energy is underestimated.

RESULTS

The nature of the transmission has been investigated and found to be that of two sets of linearly coupled oscillators (room-panel-cavity) buffered in series (cavity) and connected jointly in parallel with a third set of linearly coupled oscillators (room-wall-room). The general solution to this is analogous to a window in wall situation.

The study into the effect of changes in shape has shown the equilateral duct to have enhanced resonance peaks but to be otherwise similar to the rectangular duct. The cylindrical duct has been found to have quite different radiation characteristics from that of rectangular shape. The transmission at the ring frequency has been found to be considerable. Increasing the duct wall thickness has the effect of raising the coincidence frequency but otherwise transmission is similar since increased radiation efficiency is more or less cancelled by the additional mass.

Work on the effect of air flow in the duct on transmission of sound is still in progress but preliminary work suggests that air flow in the duct makes only minor changes in transmission loss. The transmission loss of the system with air flowing is in general less than 1dB below the transmission loss of the same system with no air flowing although differences as large as 2.2dB have been recorded under specific circumstances.

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ON THE ATTENUATION OF THE SOUND LEVEL COMING IN AN OPEN WINDOW

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INTRODUCTION

In order to solve noise control and speech privacy problems between two rooms at an open window condition, the attenuation of the sound level when coming in an open window must be known. This paper presents the experimental investigations which were made for this purpose and their results.

THEORETICAL BASES

The attenuation of sound striking a plane surface at an angle can be expressed by the following formula:

\[
\Delta L = L_0 - L_\theta = 10 \log \frac{I_0}{I_\theta} = 10 \log \frac{10 \cos \theta}{10 \cos \theta}
\]

where:
- \(L_0\) = sound pressure level on axis
- \(L_\theta\) = sound pressure level at angle \(\theta\)
- \(I_0\) = sound intensity on axis
- \(I_\theta\) = sound intensity at angle \(\theta\)
- \(\Delta L\) = attenuation of the sound level at direction \(\theta\)

Scale model investigations were designed and carried out based on the supposition that the attenuation of the sound level is affected by the window opening ratio (width/height) by the sound frequency, as well as by the direction of the sound.

EXPERIMENTAL TECHNIQUES

The experiments were made with 1/10 scale models in an anechoic room. The window opening in the source room was assumed to be a second source. Measurements were taken at nine points in the receiver room each time one of the parameters was changed. See Fig. (1) These parameters were the following:
- a) Receiver angles (15°, 30°, 45°, 60°, 75°, 90°)
- b) Sound frequency (250, 500, 1000, 2000 Hz)
- c) Window opening ratio (1/4, 1/2, 1/1, 2/1, 4/1)

The area of the window opening was constant (1 m²). In every experiment reference measurements were taken on the normal of the source room window opening at a distance equal to the distance between the source room window opening center to the receiver room window opening center. The directivity effect of the source room window opening, which was determined by another experimental investigation, was taken into account when the attenuation of the sound was calculated. (N. Aksugür 1976, to be published)

RESULTS AND CONCLUSION

The results were processed with the aid of a computer, and graphs were drawn. It is seen that the attenuation of the sound level coming in an open window is determined not only by the receiving angle effect but also by the sound frequency. The effect of the window opening ratio is negligible if the window opening ratio is the same in both the receiver room and source room.

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ACOUSTICAL BEHAVIOR OF COUPLED ROOMS

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Glasserman de Dayan, H.R. Libertad 1235, 1012 Buenos Aires,
Argentina

ABSTRACT

The object of this paper is to study the behavior of the reverberant sound field in coupled rooms. As initially isolated rooms are coupled, there is a definite change in the excited reverberant field due to the energy interchange between them.

A mathematical model has been employed in order to analyze the system. That way, a comparison was performed between calculated sound energy density values (or the equivalent sound levels), for isolated and coupled states, both during the steady and the transitory states of the phenomena. Corresponding values were obtained experimentally both for coupled and isolated states, and the differences between the calculated and experimental values are presented and discussed.

This method has a direct application to the problem of sound insulation measurement, when low insulation partitions (or partitions with openings) are involved (1-3), since there is an interchange of energy between the source and the receiver, and vice versa. Another case where the same phenomenon is found, is the one related to the coupling between the stage and the hall of a theatre, where the reverberation time of the last is strongly influenced by the former. The shortcomings of the mathematical model are also presented, like, for instance the possible different interpretations of the equivalent area, when reverberation time and sound insulation problems are studied.

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(3) D. A. Bies and J. M. Pickles: "The measurement of the transmission loss of a low noise reduction test item" Proceedings Noise, Shock & Vibration Conference (Melbourne, Australia, 1974) 144/153
STANDARD ON AIR-BORNE AND IMPACT SOUND INSULATION FOR BUILDINGS (PART I)

Kimura, S. College of Science and Technology, Nihon University, Tokyo - Japan
Yasuoka, M. Faculty of Engineering, Toyo University, Tokyo - Japan

The proposed JIS standard provides for reference frequency characteristics denoting sound insulation grades (Fig.1, Fig.2) and the application of the grades (Table 1) in respect of average sound pressure level differences between rooms and floor impact sound levels, in order to serve as a basis for proper evaluation of the noise reduction efficiencies of buildings.

The Subcommittee collected as much relevant data as possible and compiled them according to the type of buildings and the frequency of noise complaints. The reference curves were prepared on the basis of the data analysis.

The standard curve of ISO for floor impact noises cannot make full evaluation of the floor structure whose peak is in lower frequency range. In Japan, the concrete slab thickness of common multifamily dwellings is smaller than that is in Europe with lower rigidity, of which the fact poses a problem of impact noise insulation to heavy impacts in low frequency range. Additionally, for house occupants, the problem may be the floor impact noise level itself that occurs practically so that we decided to employ inverse A characteristics that directly relates to the loudness evaluation of noise as standard frequency characteristics of impact sound insulation grade.

Three application grades were decided in accordance with the relations between many practical status of sound insulation performance for buildings and status of claim occurrence, the results of auditory tests conducted by the committee members.

<table>
<thead>
<tr>
<th>Wall and Floor between houses</th>
<th>Grade I (Standard)</th>
<th>Grade II (Permissible)</th>
<th>Grade III (Minimum)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Floor between houses</td>
<td>L-45</td>
<td>L-50, L-55</td>
<td>L-60</td>
</tr>
</tbody>
</table>

REFERENCE
STANDARD ON AIR-BORNE AND IMPACT SOUND INSULATION FOR BUILDINGS (PART 2)

Yasuoka, M.  Faculty of Engineering, Toyo University, Tokyo - Japan
Kimura, S.  College of Science and Technology, Nihon University, Tokyo - Japan

INTRODUCTION
The reference characteristics of JIS draft (see part 1) are discussed, especially, the necessity of inverse A characteristics is explained.

CONSIDERATION
For air-borne sound insulation, the standard curve of JIS is similar to that of ISO, because, it should be given as the result of assumptions that source room band level should be an envelope curve for various kinds of sounds, and receiving room band level should be equal annoyance or loudness level, then, the sound pressure level difference between source room and receiving room should be equivalent to the sum of A weight and the reference frequency characteristics (fig.1, fig.2).

For impact sound insulation, the standard curve of JIS, we call inverse A curve, is not so similar to that of ISO. In case of impact sound, reference frequency characteristics should be given not for level difference as airborne sound but for receiving room level itself, on various types of floors and impact sources (fig.3), and probably, A weight curve is the most suitable one for evaluation of loudness or annoyance, then it should be inverse A curve. In comparison with evaluation ability of JIS and that of ISO, we can say as follows. Fig.4, for impact source variation, ISO is not only in no proportion to loudness but in no order. Fig.5, for floor coverings, ISO is also in no proportion to loudness. Fig.6, for ceiling reduction etc., ISO gives too much value to them, because, ISC curve touches only at high frequency range in which T.L.is larger value.

CONCLUSION
Inverse A curve is the most useful evaluation standard for impact sound insulation of floors.

REFERENCE
RHEOLOGICAL CONSTANTS AND THE PROPAGATION OF ELASTIC WAVES IN ROCKS

Chudek, M.  Institute of Coal Mine Designing, Technical University of Silesia, Gliwice
Szyma, S.  Institute of Physics, Technical University of Silesia, Gliwice

1. In the practice of mining very often seismological methods as well as some subjective methods are applied. Hence on the one hand there arises the necessity of scribings different definitions of the intensity of stresses to different degrees of mechanical stimuli, whereas on the other hand we must evaluate the relations existing between the stresses in the orogen and these stimuli. Therefore for the purpose of describing the compressibility of rocks in the case of slight deformations we apply Hooke's law. In the case of rocks we must be aware of the fact that changes in volume are accompanied by some structural changes, it is the aim of this paper to show that such an effect leads to considerable complications and that Hooke's law cannot form a basis for the objectivisation of assumed subjective quantities.

2. Let us assume that the physical state of the orogen can be determined by means of the compression Z and the structure S. In the case of slow changes in the density of rocks the structure will change respectively. The change S is in such a case proportional to the same deviation S and to the time dt. If dz/dt \neq 0, the change Z leads to a further deviation from the parameter to the value S. Taking into consideration the value of the exchanged effect we obtain

\[ \frac{dS}{dt} = - \frac{1}{\alpha} \frac{dZ}{dt} - \frac{1}{\alpha} \frac{1}{\beta} \frac{dZ}{dt} \frac{dS}{dt} \]

Taking further into account the growth of stress \( \Delta \sigma_{l}^{n} \), corresponding to the equilibrium, and the partial stress \( \Delta \sigma_{l}^{p} \), resulting from the fact that this system has not reached its equilibrium, it is easy to find the relation between stress and the parameter S, the value of S being given:

\[ \Delta \sigma_{l}^{n} = \Delta \sigma_{l}^{n} + \Delta \sigma_{l}^{p} = - \kappa_{lm} Z - \alpha \beta \Delta S \]

where \( \Delta S = - \Delta \sigma_{l}^{n} \frac{1}{\alpha} \frac{1}{\beta} \frac{dZ}{dt} \). From /1/ and /2/ we deduce:

\[ \frac{dZ}{dt} = \frac{1}{\alpha} \frac{d\sigma}{dt} + \frac{1}{\beta} \frac{d\sigma}{dt} \]

From this equation results that a change of stress in the main directions leads to a compression of the orogen, comprising the compression Z, developing immediately, and the compression which changes with time, and in this case

\[ Z_{z} = - \Delta \sigma_{l}^{n} \frac{1}{(\kappa_{l} + \kappa_{p})}, \quad Z = - \Delta \sigma_{l}^{n} \frac{1}{(1 - e^{-\frac{\kappa_{p}}{\kappa_{l}}})} \frac{1}{(\kappa_{l} + \kappa_{p})} \]

where \( Z_{z} = Z_{z} (1 + \kappa_{p}) \) is the effective time of relaxation.

Making use of equations /5/ it is easy to prove that

\[ \frac{\kappa_{p}}{\kappa_{l}} = \frac{\kappa_{p}}{\kappa_{l}} \]

In order to describe the considered properties of rocks we must determine four independent parameters, viz. \( C_{i}, \tau / i = 1,2,3 \). If the measurements of stresses are limited to only two main directions— which is most often the case in the practice of mining—the number of parameters will be reduced to two independent parameters, viz. C and \( \tau \), as described by the formulae:

\[ C = \frac{\Delta \sigma_{l}}{\Delta \sigma_{z}}, \quad \tau = \frac{1}{\kappa_{l}} : (1 + C) \]

It results from these considerations that the method of measuring the investigated properties defined by the parameters \( C_{i}, \tau \) depends on the effective time of relaxation. If this time is very short or very long, direct measurements become rather difficult.

This assumption has been confirmed by the found values of C and \( \tau \).

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ON THE SOUND POWER RADIATED BY A VELOCITY MONOPOLE UNDER REVERBERANT AND UNDER FREE FIELD CONDITIONS

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INTRODUCTION

The acoustic power radiated by a source can be measured under free field conditions (i.e. in an anechoic chamber) or in a diffuse field (i.e. in a reverberation chamber). Several authors [1] have commented on the discrepancy of the results of the two methods, noting that the results for the reverberation chamber tend to be lower, especially at low frequencies. In this paper an explanation of this discrepancy will be given, which has been confirmed by experiments.

THEORETICAL BASES

From the MORSE eigenmode model of a sound field [2] it can be derived that the acoustic power $W$, radiated into a reverberation chamber by a velocity monopole driven by a band of noise, is given by

$$W = \frac{\langle p_{\text{rms}}^2 \rangle}{A} = \frac{W_o M}{M_0} = W_o (1 + T),$$

where $\langle p_{\text{rms}}^2 \rangle$ is the "true space average" of the mean square pressure, $A$ is the total absorption in the room, $M$ the total number of eigenmodes in the band, $M_0$ the high frequency asymptotic value of $M$ and $W_o$ the sound power radiated under free field conditions. $T$ depends on the room dimensions, is always positive and decreases with increasing frequency.

During sound power measurements in a reverberation chamber the source and the microphone are always kept away from the room boundaries. Therefore, in eq. (1) the true space average, which requires averaging of source and microphone positions over the whole volume (including boundary areas), is replaced by $\overline{p^2}$, the value averaged over the central area only. We can derive that for a rectangular room

$$\overline{p^2} = \frac{\langle p_{\text{rms}}^2 \rangle}{(1 + T)^2}$$

Combination of eq. (1) and (2) shows that the acoustic power $W_c$, measured in this way, is:

$$W_c = \frac{W_o}{1 + T},$$

which means that $W_c$ is indeed lower than $W_o$, especially for lower frequencies.

EXPERIMENTAL TECHNIQUES

Values of $p_{\text{rms}}^2$ have been measured for several third octave bands, at points in the whole volume of a reverberation chamber. Averaging over different volume areas confirms the above theory.

RESULTS AND CONCLUSIONS

This theory gives a good description of the discrepancy between sound power values measured in free field and reverberant conditions.

REFERENCES

WIND NOISE ENVIRONMENT IN HIGHRISE BUILDINGS

Berhault, J-P. A. Institute of Sound and Vibration Research
Davies, P.O.A.L. University of Southampton, England

INTRODUCTION

The aim of this work was to investigate the wind created noise environment in highrise blocks of flats. All the project measurements were carried out on a full scale level.

THEORETICAL BASES

Wind flow around buildings has received a lot of attention to this day. Scientists and designers are concerned with structural resistance to specific extreme wind conditions or with low pedestrian level speeds distribution. Thus, theoretical description of the lower atmosphere fluid dynamics is well documented and has been extensively used to help understanding the noise generation process which can be divided in two main groups:

- Atmospheric turbulence generated noise which lays mainly in the infrasonic range.
- Noise resulting from the building-wind interaction made up of higher velocities aerodynamic noise and of through flow noise within the building. This results in medium and high frequency noise strongly governed by local room acoustics.

EXPERIMENTAL TECHNIQUES

To get appropriate long range data, various buildings in the Southampton Area were instrumented. Wind characteristics, speed and direction, were continuously monitored. Noise and vibration data were acquired for a complete range of wind conditions, flat situation in the building etc. Then standard analysis techniques such as spectra, cospectra, cross-correlation were used on the wind, noise and vibration data to provide quantitative information on the noise/wind speed relationship. Then, the long range observations were used statistically to forecast the average wind noise environment in relation to cumulated wind data.

RESULTS AND CONCLUSIONS

Data collected over a period of three years have been used to evaluate the noise/wind-speed relationship in a certain number of room types located in normal permeable buildings. This relationship can be used to predict the wind noise environment in such type of buildings when one knows the averaged basic wind speed for this particular location. It was found as well that the noise climate depends very much upon the actual flat location in the building as regards the prevailing wind direction. Some recommendations are made to architects and planners for details of construction or proper siting of highrise buildings in areas where the basic averaged wind speed is in excess of 5 m.s⁻¹.

![Graph showing SPL vs Windspeed for various room types]
1. Environmental acoustics

D. ACOUSTIC MATERIALS
INTRODUCTION.
The reverberation theory used generally for evaluation acoustic properties of enclosures is useless for industrial halls and urban interiors, since in these cases the conditions of equal distribution of acoustic energy density are not fulfilled. Moreover, the means of arranging sound absorbing materials are not then defined. For these cases the development of the image sources method is very promising.

THEORETICAL.
Using the mirror method the absorbing surfaces are treated as an equivalent barrier against image sources, according to the following representation:

<table>
<thead>
<tr>
<th>Absorbing band</th>
<th>Equivalent barrier</th>
</tr>
</thead>
<tbody>
<tr>
<td>width d</td>
<td>height h ed</td>
</tr>
<tr>
<td>reflection coefficient $\rho_b$</td>
<td>transmission coefficient $\gamma$</td>
</tr>
<tr>
<td>length l</td>
<td>width b</td>
</tr>
</tbody>
</table>

where $d$ denotes the angle of an incident wave on the absorbing band. The Insertion Loss of absorbing surface can be expressed by the term

$$ IL_a = 10 \log \frac{P_r}{2P_{da} + P_{ra}} = 10 \log \frac{1}{\text{SNR} + \frac{\rho_b}{\gamma}} $$

The meaning of the squared sound pressures are: $P_r$ - for the wave reflected without absorbing material, $P_{ra}$ - for the wave reflected by the absorbing surface /penetrating the equivalent barrier/, $P_{ra}$ - for the wave diffracted by the equivalent barrier, N-Fresnel number.

EXPERIMENTAL.
To check the method given above, the experimental investigations were made in an anechoic chamber where the reflecting ceiling with and without absorbing surface was placed. The pulse method with the spark source was used.

RESULTS.
The results of measurements and calculations /Fig.1/ are in sufficient agreement. This proves the idea of treating sound absorbing surfaces as equivalent barriers against image sources. The method can be used for the calculation of barriers in enclosures as well as for evaluation of edge effect of absorbing materials.

REFERENCES.
GERICHTETE ABSORBER

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EINLEITUNG UND AUFGABESTELLUNG


THEORETISCHE GRUNDLAGEN


MESSUNGEN

Messungen fanden im Studio (VI) des Belgrader Rundfunks statt. \( V = 1600 \text{ m}^2, T = 0,45 \text{ s} \)

Messanordnung vom Bild (Fig.1) ermöglichte Empfang der ersten Reflexion und Aufzeichnung von Pegelwechsel bei Änderung des Absorptionskoeffizienten an der 1 m² oder 2,25 m²-großen Prüffläche. Die Impuls- und Dauersignale wurden vom Lautsprecher gesendet und von einem Richtmikrophon empfangen.

RESULTATE UND BESCHLÜSSE

Wegen Einfluss des stationären Schallfeldes und der Dimension der reflektierenden Fläche (1 bis 2,25 m²), die Messergebnisse zeigten Abweichungen gegenüber berechneten Pegelwerten von 1 bis 3 dB, je nach Frequenzbereich. Man bekam frequenzabhängige Pegelminderungen von etwa 3 bis 7 dB. Gemessene und subjektiv empfundene Resultate bestätigen die Erwartung. Das Problem wird weiter erforscht.

Fig.1 Messanordnung im Grundriss
THE ACOUSTICS OF GRADUAL TRANSITION ABSORBERS

Toivanen, J. Helsinki University of Technology, Espoo, Finland

INTRODUCTION

To calculate the sound inside an anechoic chamber, the reflection coefficient of the absorbers should be known, not only by perpendicular incidence.

THEORETICAL BASES

Defining a "perpendicular characteristic impedance" \( Z = Z / \cos \theta \) and "perpendicular propagation constant" \( \gamma = \gamma \cos \theta \), we can calculate the perpendicular impedance at the surface of a homogenous layer of thickness \( d \) by

\[
Z_\perp = \frac{Z_{\perp \perp}}{Z_{\perp \perp} + Z_{\perp \perp} \tanh (\gamma_\perp d)}
\]

where \( Z_{\perp \perp} \) is the perpendicular impedance loading the layer's back surface.

Common porous absorption materials are acoustically equivalent to lossy, isotropic fluids, provided their structural inhomogeneities are much smaller than the wavelength /1/. The equivalent density and compressibility of a composite medium consisting of two separate fluids are the arithmetic averages of those of the fluids, weighted by partial volumes /2/. At low frequencies, the reflection coefficient of a gradual transition absorber can be calculated by substituting an equivalent layered structure for the absorber and using eq. (1).

RESULTS

It can be shown that gradual transition absorbers, made of practically any common porous material (see /3/ for material constants), and mounted on a rigid wall, are approximately locally reacting, provided the absorber thickness is less than ca. one half of the wavelength of the airborne sound. This result is valid practically independent of the absorber shape. The reflection coefficient of practical absorbers used in anechoic chambers usually has a minimum by perpendicular incidence. These results enable us to calculate the sound field inside an anechoic chamber on the basis of absorption tube measurements. Even simple procedures, such as an image model with a plane wave approximation, give satisfactory results.

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(2) J. White, F. Angona, J.A.S.A. 27 (1955), 310...317
**Introduction**

On se propose d'étudier la propagation du son dans des conduits de section géométrique simple lorsque cette section varie de façon absolument quelconque dans une direction parallèle à un axe oz (fig. 1). La méthode utilisée, basée sur la théorie des modes, consiste à décomposer le guide étudié en une succession de petits changements de section élémentaires dont l'enveloppe extérieure épouse la forme du guide puis à étudier de proche en proche la modification de la matrice impédance généralisée $Z$ par les changements de section et par la propagation dans les petits éléments de guide rectiligne qui les raccordent.

**Théorie et méthodes de calcul**

Dans un élément de guide rectiligne et en régime harmonique, on sait que la pression et la vitesse vibratoire longitudinale sont décomposables sur la base des fonctions propres $\psi$ du guide, fixées par les conditions aux limites. Appelons $P$ et $V$ les vecteurs colonnes dont les éléments sont respectivement les composantes sur $(\#,)$ de la pression et de la vitesse au point $z$. Soit $Z(z)$ la matrice définie par

$$ P = Z(z) \cdot V $$

la matrice $Z(z)$ est une généralisation à l'ensemble des modes de la notion d'impédance de l'onde plane.

La théorie modale permet, par des relations matricielles, d'étudier l'évolution de $Z(z)$ le long d'un trajet $l$ dans un guide à section constante ou lors de changement de section.

Après discrétisation du conduit, la méthode de calcul est la suivante: on part de l'impédance de la face terminale supposée connue. On calcule de proche en proche les modifications de la matrice $Z$ lors des propagations élémentaires de longueur $dl$ et des changements de section de $h$ à $h + dh$. On aboutit ainsi à la matrice $Z_{n}$ à l'entrée du conduit. Si l'on se fixe un champ incident, on en déduit le champ réfléchi et le champ transmis.

Cette méthode de calcul implique l'usage d'un calculateur numérique mais a le grand avantage sur d'autres méthodes de discrétisation de ne pas imposer un encombrement exagéré en mémoire centrale et ceci quelle que soit la finesse du découpage. En effet les calculs se font successivement et il n'est pas nécessaire de conserver la trace des valeurs intermédiaires de $Z$. Seule la propagation des erreurs d'arrondis limite le nombre "d'escaliers". Les vérifications que nous avons faites, soit expérimentalement, soit par comparaison avec d'autres méthodes applicables dans le cas de sections lentement variables, ont permis de montrer que les résultats convergeaient très rapidement avec le nombre de modes utilisés et le nombre "d'escaliers".

La méthode proposée fournit avec une très bonne précision la modification du champ sonore résultant de variations quelconques de la section d'un guide. Elle s'avère très utile pour la prédétermination des effets de configuration compliquées des conduits et peut avantageusement remplacer des manquettes.
A rigid frame acoustic absorbent has been developed by ICI Organics Division which has a unique combination of structural strength and high sound absorption.

The precise description of the material has been given elsewhere (1). It is made up of mineral particles bonded to one another with an organic or inorganic binder. The developed absorbents have porosities in the region 0.3 - 0.5 and air flow resistances in the range 10 to 150x103 Mks Rayls/m. The flow resistance can be preset to within 10 Mks Rayls/m. The structure of the absorbent is far more rigid than that of any known absorbent. The absorbent mechanism would therefore be expected to conform to the rigid porous acoustic model (2), the impedance being given by $Z = Z_{ca} \coth \gamma l$ where $Z_{ca}$ is the characteristics impedance, and $\gamma$ is the propagation constant and $l$ the thickness. This has been found to be more precisely the case than with other absorbent materials where frame vibration complicates the simple model especially at low frequencies.

Using a modification of Delany and Bazley's empirical relations (3) for fibrous materials, paying special attention to avoid frequency/flow resistance ratios where frame vibration occurs, $\alpha$ and $Z_{ca}$ have been predicted for the absorbent. From these values the random sound absorption coefficient, $\alpha$ has been calculated assuming local reaction. This is compared in Figure 1 with the calculated $\alpha$ from tube impedance measurements and with the $\alpha$ measured to ISO requirements. Clearly theoretical predictions of absorption are extremely accurate for this material which has great potential where structural properties and high acoustic absorption are required.

![Graph](image)

**REFERENCES**

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3. DELANY M.E. AND BAZLEY E.N. APPL. ACOUSTICS 3, 2, 105-116, 1970
Up to now, a characteristic of reverberation sound absorption coefficient of a panel construction system has been formed by selecting proper mass of the outer plate, width of air gap, air chamber volume and the way of mounting.

However, it is possible to shape $\alpha = \frac{f}{f_0}$ by selecting respective thickness by means of another construction of a plate. To this purpose, thermoplastic properties of plastics have been taken into account and self-supporting shell elements have been constructed from polyamide resins reinforced with fibre glass /fig. 1/. The air gap has been filled with glass wool - thickness 5 cm. A change in the dimensions of an element and thus a change of its stiffness /fig. 2/ have an effect on a change of resonance frequency of the panel system.

In comparison with traditional elements, a wider absorbing range has been obtained, retaining maximal values $\alpha$ in the range of low frequencies.

The elements in the form of shells make the mounting easy. It is also possible to make these elements as perforated what in case of a different function of a room is of big importance.

By means of an element having the same shape and dimensions we can freely shape the acoustic conditions in a room.
INTRODUCTION

The method of calculating the reverberant sound absorption coefficient $\alpha_{rev}$ of a space absorber of sheet type porous materials with any geometrical parameter of suspension was obtained when the fractions of the random incident energy reflected $Y_m$ and absorbed $\lambda d$ by the materials itself were given.

THEORETICAL ANALYSIS

The chamber may be considered to be virtually divided into two spaces by the absorber (Fig.1); the space I and the space II. The sound fields in them are assumed to be diffused. From the differential equations of the energy exchange between two spaces, the reverberation time $T$ of the space I is obtained as follows,

$$
T = -\frac{\ln[(X_2+B/D)/(X_2-X_1)] + 6\ln10}{A + X_1C} (1)
$$

where

$$
A = -C = \frac{Co[Sm(1-Y_m)+Sw+(So-Sm)\omega]/(4V_1)}{Co[Sm(1-Y_m-\lambda d)+Sw+Sm\omega]/(4V_1)},
$$

$$
B = \frac{C}{2} = \frac{Co[Sm(1-Y_m-\lambda d)+Sw]/(4V_2)}{Co[Sm(1-Y_m)+Sw+Sm\omega]/(4V_2)},
$$

$$
C = \frac{Co}{2} = \frac{Sm[Sm(1-m-x^2)+Sw]/(4V_2)}{Co[Sm(1-m-x^2)+Sw+Sm\omega]/(4V_2)},
$$

$$
D = -C = \frac{Sm[Sm(1-km)+Sw+Sm\omega]/(4V_2)}{Sm[Sm(1-km)+Sw+Sm\omega]/(4V_2)},
$$

$X_1$ and $X_2$ ($X_1 > X_2$) are the solutions of the equation $[CX^2+(A-D)X-B=0]$. $Sm$: one side area of the absorber, $Sw$: side area of the space II, $So$: area of the inner surface of the chamber, $Co$: sound velocity, $V_1$: volume of the space I, $V_2$: volume of the space II, $x_0$: average reverberant absorption coefficient of So. The value of $\alpha_{rev}$ is given by

$$
\alpha_{rev} = \frac{55.26(V_1+V_2)}{Co Sm} \left[1 - \frac{1}{T_0} \left(1 - \frac{Sm}{So}\right)\right] (2)
$$

where $T_0$ is the reverberation time when the chamber is empty, i.e., when both of $Y_m$ and $\lambda d$ are taken equal to zero in Eq.(1).

RESULTS

The values of $\alpha_{rev}$ calculated by Eq.(2) and the measured values are shown in Fig.2. The calculated values when the absorber comes near to the reflecting boundary is corrected considering the interference pattern in the field near the boundary.$^9$

REFERENCES

1) R.V. Waterhouse, J.A.S.A. 27(1955)247

Fig.2 Reverberant absorption coefficient as a function of the height $h$.
In an earlier paper, one of us described [1] the use of certain pseudo-random sequences ("maximum-length" sequences) to construct hard-wall surfaces that reflect sound with high diffusion. In a subsequent Letter [2] we described the two-dimensional generalization of this concept. In this paper, we present another type of sequence, which we will call "Quadratic Residue" (QR) sequences, with even better sound diffusing properties — particularly in terms of the frequency range for which diffuse reflection is achieved.

In one dimension, a QR-sequence is given by

\[ r_n = e^{i\psi_n} = e^{i2\pi n^2/N} \]  

where \( N \) is an odd prime number and \( r_n \) is periodic with a period of \( N \). The periodic autocorrelation function of \( r_n \) is

\[ \psi_k = \sum_{n=0}^{N-1} r_n r_n^* = \begin{cases} 1 & \text{for } k = 0 \text{ modulo } N \\ 0 & \text{otherwise.} \end{cases} \]  

The corresponding power spectrum is flat i.e., it consists of discrete components of equal magnitude. The reflection pattern of a wall with reflection coefficients according to Equation (1) is uniform for all (discrete) angles of reflection.

For \( N = 7 \), the \( r_n \) have four different phase angles: \( \psi_0 = 0; \psi_1 = \psi_6 = 2\pi/7; \psi_2 = \psi_5 = 8\pi/7; \psi_3 = \psi_4 = 4\pi/7 \). Corresponding to 4 different wall depths:

\[ d_n = \frac{\psi_n}{4\pi} \]  

where \( \lambda_0 \) is the wavelength of the "design frequency".

The average wall depth according to Equation (3) is only one seventh (1) of the wavelength. The standard deviation of the wall step size equals only 1/14-th of the wavelength \( \lambda_0 \) (as opposed to 1/8-th for the maximum-length sequences). Thus, the QR-designed walls are relatively smooth — an important architectural advantage.

The most interesting property of the QR sequences is that walls, constructed according to Equations (1) and (3), have optimum diffusion also at all integer multiples of the design frequency up to and including \( N-1 \). (At \( N \) times the design frequency \( r_n = 1 \) for all \( n \); hence at that frequency, the wall becomes a specular reflector.)

SOUND REFLECTION FROM LOCALLY REACTING ELEMENTS IN PLANE TILES

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Physics Department, Acoustics Group

INTRODUCTION

Analysis of reflection of sound from rigid planes in which groups of AB/TR (absorption or transmission) elements of complex impedance are embedded (see Fig 1) has a major importance concerning acoustic tiles and barriers. A practical technique, based on Kirchhoff's integral of diffraction (1) gives a simple solution based on a complex efficiency coefficient.

ANALYSIS

The influence of an elemental area of a given tile on the sound pressure, caused by a spherical source, at a control point P is:

\[ T_{s} = \left( \frac{dU(P)}{ds} \right) = \sum_{m=1}^{N} \frac{1}{\omega_{m}} \exp(ikl_{em}) \left[ \cos(n, m) - \cos(n, l) \right] (a_{m} + ib_{m}), \]  

where \( a \) and \( b \) is the reflection coefficient of the \( q \)th kind of AB/TR elements. If the non-uniformity of the distribution of each kind of the AB/TR elements is mainly within grains of a side which is about the distance between two adjacent Z lines (2)

then the real and imaginary parts of the final sound pressure in \( P \) are:

\[ R_{m}(U) = R_{m}(U) + \sum_{q=1}^{N} \left( \frac{1}{\omega_{q}} \left[ \cos(n, m) - \cos(n, l) \right] R_{m}(U) \right), \]  

\[ I_{m}(U) = I_{m}(U) + \sum_{q=1}^{N} \left( \frac{1}{\omega_{q}} \left[ \cos(n, m) - \cos(n, l) \right] I_{m}(U) \right). \]  

\( S \) is the area of the tile, \( S_{q} \) is the overall area of the AB/TR elements of the kind \( q \). \( \Delta u \) is the contribution of a rigid tile of the same size. (\( L \)) indicates any quantity divided by \( \Delta u \). From equations (2, 3) we define the efficiency coefficient \( \eta \) :

\[ \eta = \sum_{q=1}^{N} \left[ \frac{1}{\omega_{q}} \left( \cos(n, m) - \cos(n, l) \right) R_{m}(U) \right] + \left[ \cos(n, m) - \cos(n, l) \right] I_{m}(U) \],

and finally the SPL at \( P \) is:

\[ \text{SPL} = \text{SPLO} + 20 \log_{10} \left[ (R_{m}(U) + R_{m}(U)) - (\Delta u) \right] \left( I_{m}(U) + I_{m}(U) \right) \left( \eta \right)^{2}. \]  

CONCLUSIONS

The presentation of \( \eta \) enables the acoustician to use results of rigid tiles for analysis of reflection from locally reacting nonhomogeneous tiles.

REFERENCES

(1) M. Born, E. Wolf, Principles of Optics, 4th ed. (1970), 380
(2) G. Rosenhouse, Acoustics (1977) 52.
INTRODUCTION

A fluid mechanical model is presented of the nonlinear acoustic behavior of Helmholtz resonators as a function of incident sound pressure level, frequency and resonator geometry.

THEORETICAL BASES

The model is based on the following assumptions: (1) sound approaches the resonator in a spherical manner, (2) the acoustic pressure and density are adiabatically related and (3) the amplitude of the incident acoustic pressure is small relative to the ambient pressure.

The model equations are derived as follows. First, the conservation of oscillating mass and momentum are specified. Second, the resulting equations are nondimensionalized by the quantities \( L, \omega^{-1}, q \) and \( P_i \) where \( L \) is a reference length (experimentally determined) proportional to the orifice diameter \( d \), \( \omega^{-1} \) is the radian sound period, \( q \) is the particle maximum speed at the orifice (vena contracta) outlet and \( P_i \) is the amplitude of the incident sound pressure. The relationship between \( P_i \) and \( q \) is provided from the experimental findings of Ingard and Ising (ref. 1). Their measurements showed that \( P_i = \rho q^2 \) whenever \( P_i \gg \rho (\omega d)^2 \). Third, the effects of fluid viscosity and heat conductivity are assumed to be negligible.

Retaining only first-order terms, the non-dimensionalized governing equation of motion, as shown in eq. (1), are

\[
E[F^* (\frac{2\pi L \omega^2}{V_0}) F] + FF = -\sin(t); \quad E = \sqrt{\rho/(\omega L)^2/P_i}
\]

The solution to eq. (1) is singular at \( t = (2n-1)\pi/2, n=0,1,... \). The method of singular perturbation theory is used to determine an approximate solution. The resulting Helmholtz resonator resistance \((R/\rho c)\) and reactance \((X/\rho c)\) are approximated as shown in eqs. (2) and (3)

\[
\frac{R}{\rho c} \approx 0.08 (d/L)^2 \sqrt{P_i/\rho c^4}
\]

\[
\frac{X}{\rho c} \approx 0.05 (d/L) (\omega d/\sqrt{c})[-0.431 nE-(2\pi L \omega^2/V_0^2) (1.54-0.7 E^{-3})]
\]

EXPERIMENTAL VERIFICATION

The two-microphone method of measuring the impedance of Helmholtz resonators was used to verify the validity of the model. The characteristic length \( L \) used to normalize the equations of motion, was determined experimentally to be approximately \( L=0.27d \). Substitution of this value of \( L \) into eqs. (2) and (3) resulted in good agreement between measured and predicted values of \((R/\rho c)\) and \((X/\rho c)\) for a wide range of resonator geometries.

REFERENCES

THE ACOUSTIC PROPERTIES OF PILE MATERIALS AND THEIR INVESTIGATION

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Vibro-Engineering Research Laboratory,  
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INTRODUCTION

Studies were made on the acoustic-aerodynamic behavior of pile materials in air flows and air flow constructions.

THEORETICAL BASES

Pile covered materials consisting of a base and pile covering being resistant to air flow eliminate the necessity of protective coverings, readily admit sound waves, therefore, as regards acoustics — pile materials have proved to be very effective deadeners. Due to their damping properties, pile materials exert influence on the aerodynamic parameters of the flow.

EXPERIMENTAL TECHNIQUES

Two cases of the acoustical effectiveness of pile materials were investigated: 1) in a duct in which the air flow and noise is caused by a fan, 2) in a duct absent of air flow the source of noise being a loudspeaker instead of a fan. In both cases the amplitude-frequency characteristics identically coincided. The turbulent pulsation of flow velocity was measured in pile lined and technically smooth ducts by thermoanemometry.

On the basis of the results of these measurements calculations were made on the energy spectral density of turbulence.

RESULTS AND CONCLUSIONS

In ducts, absent of air flow, pile coverings considerably attenuate noise due to sound absorption. The sound attenuating properties of pile materials are greatly enhanced in air flows. The intensity of sound waves is considerably reduced in air flow ducts lined with pile materials due to the sound absorption as well as the aerodynamic characteristics of those materials. Pile coverings lessen the turbulent pulsation of the flow in the center of the duct thereby reducing the intensity of sound waves in an air flow duct. The acoustic-aerodynamic properties of pile materials depend on elasticity and structural parameters.
INTRODUCTION

Where windows with high sound insulation are required, two panes separated by a large airspace are normally used. Two ways of getting less bulky constructions will be presented.

THEORETICAL BASES

The problem is to avoid the detrimental coincidence effect. One way is to hold the coincidence frequency above 3150 Hz by using thin multiple panes separated by very small airspaces. These are necessary to avoid the Newtonian rings. The other way is to use laminated panes with so high internal losses that no coincidence dip is achieved.

EXPERIMENTAL TECHNIQUES

Multiple panes with airspaces increasing from zero to several centimeters where measured according to ISO R140 to study the effect of resonances due to the spaces. Laminated panes with different transparent plastic binding foils where measured in the same way to study the required loss factor of the plastic material. The temperature dependence on the loss factor was measured down to -20°C with smaller samples.

RESULTS AND CONCLUSIONS

With airspaces below 0.05 centimeter the resonances do not limit the possibilities of getting a high "index of airborne sound insulation". Plastic intermembers with a loss factor larger than 0.50 down to -20°C are convinent for avoiding coincidence dips in laminated facade windows.

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REMARKS ON THE VOLUME OF REVERBERATION ROOMS TO BE USED FOR THE SOUND ABSORPTION MEASUREMENT

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UNDERLYING PROBLEMS ON THE ROOM VOLUME

For acoustic measurements in reverberation rooms, it is generally important to realize diffuse sound field in the room. It has been considered that several factors are closely related with the sound field in the room.

At low frequencies, the normal modes have been considered to be the predominant factor. From the well-known normal mode theory, the number of normal modes for certain frequency band increases in proportion to the volume of the room. The concept of Schroeder's limiting frequency would be considered to be the concluding guideline for the room volume. The necessary condition for room volume is given by the following formula, in relation with the lowest frequency of measurements $f$ and the reverberation time $T$ of the room (1).

$$V \geq 4 \times 10^6 \frac{T}{f^2}$$  \hspace{1cm} (1)

On the other hand, at high frequencies, the air absorption has been considered to be the important factor. By simple estimation, following condition should be fulfilled for the room volume.

$$V \leq \left( \frac{3 \bar{\alpha}}{20 \cdot \Delta m} \right)^2$$  \hspace{1cm} (2)

Here, $\bar{\alpha}$ is the average sound absorption coefficient of the room boundaries and $\Delta m$ is the variation of sound attenuation constant per unit length corresponding to the variation of temperature and humidity. In order that the measurements might be carried out at frequencies up to 4000 Hz under the condition of $\bar{\alpha} = 0.02$ and $\Delta m = 0.001$, the room volume should be smaller than 30 cubic meters.

RECOMMENDATION ON THE VOLUME OF REVERBERATION ROOM

It would be extremely difficult to use only one reverberation room over the wide frequency regions, for example from 125 Hz to 4000 Hz. As the reasonable solution for this problem, it would be recommended to be used two or three reverberation rooms with different volume to cover relatively narrow frequency band, respectively. Example of a set of reverberation rooms is shown in Table 1.

<table>
<thead>
<tr>
<th>Frequencies (Hz)</th>
<th>125~315</th>
<th>400~1600</th>
<th>2000~4000</th>
</tr>
</thead>
<tbody>
<tr>
<td>Volume (m$^3$)</td>
<td>1200</td>
<td>200</td>
<td>30</td>
</tr>
</tbody>
</table>

After adopting this set of reverberation rooms, some other problems will remain to be solved. These will be concerned with the so-called edge effect. It would be necessary to choose the optimum areas of test specimen for each reverberation room.

REFERENCE  \hspace{1cm} (1) Schroeder, M., Acustic;  (1954) 594
INTRODUCTION

There is, resulting from increasing environmental pressures, U K Government legislation and E E C regulations, a growing need for a more detailed understanding about the acoustic and vibration specifications of most materials, equipment and machinery. In order to satisfy this demand, Sound Research Laboratories have just completed the first building phase of an acoustic and aerodynamic research centre.

NEW ACOUSTIC AND AERODYNAMIC TEST CENTRE

This paper is intended as an appraisal of the building design concept, its commissioning and the data acquisition methods to be used. Several technical innovations have been incorporated. Some of these were necessary because of external constraints, while others were deemed appropriate by our own design team. The entire project should be viewed in the context of optimising cost effectiveness given that a budget limit of £100,000 was set for all buildings, plant, mechanical handling equipment and measurement systems.

DESIGN

Arguments will be submitted to support the design being developed around a main 300 cubic metre reverberant test chamber, with a novel split 110 cubic metre transmission room. Half of the room is "floating" with respect to the other half. This enables testing to be carried out in accordance with existing B S and T S O standards, but also anticipates impending changes such that transmission loss tests on a floor to ceiling, wall to wall barrier can also be carried out between two 50 cubic metre rooms. It is hoped to present comparative test results between transmission loss tests carried out on the same materials in each of the test arrangements.

DATA ACQUISITION AND ANALYSIS

In addition, an unusual real time 8 channel data acquisitioning system will be described. This is already in use for sampling spatial and temporal sound pressure variations in test chambers. It includes an interface between the R T A and a data logging computer terminal. This permits the use of pre-recorded digital cassette tapes to programme the R T A automatically or by requesting the operator to insert the required returns, and then to format the output data in a clear manner. The terminal can access a DEC KI 10 computer via a GPO modem unit. Simultaneous playback and record facilities have been provided.

COMMISSIONING

Finally the paper will cover the commissioning procedures and qualification tests of the test chambers to the A N S I S I, 21 - 1972. The results of further investigations using rotating and static diffusers will be presented subject to availability.

Ref: American National Standard Methods for the determination of sound power levels of small sources in reverberation rooms S I, 21 - 1972
ETUDE EXPERIMENTALE SUR MAQUETTE, DE LA PROPAGATION DU SON AU-DESSUS D'UNE SURFACE ABSORBANTE

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Introduction

On compare les résultats expérimentaux avec ceux provenant d'une théorie mise sous forme d'un programme machine.

Bases théoriques

L'étude de la diffraction d'une onde sphérique harmonique par une surface plane représentable par un milieu poreux équivalent d'impédance quelconque, est élaborée (1).

Pour cela, on fait appel à la transformée de Fourier bidimensionnelle qui permet d'aboutir à un problème caractérisé par une condition aux limites à une dimension. Cette transformée est mise sous la forme d'une somme de fonctions qui sont ensuite inversées aisément. On étudie en particulier le champ asymptotique quand la source et le point de réception sont au contact de la surface :

\[ G(S,X) = -2i \frac{k_1^2 - k_1^2}{k_1^2 - k_1^2} \cdot \frac{\rho_1^2}{\rho_2^2} \cdot \frac{e^{ik_1r}}{4\pi^2} \]

\( k_1 \) : constante de propagation du milieu I (air
\( k_2 \) : constante de propagation du milieu II (absorbant)
\( \rho_1 \) : masse volumique du milieu I
\( \rho_2 \) : masse volumique effective du milieu II
\( r \) : distance entre la source et le récepteur

Les milieux I et II sont supposés semi-infinis, homogènes, isotropes ; la surface est à réaction étendue.

Technique expérimentale

Dans la salle anéchoïque du L.M.A., un plan réfléchissant (5,5x4m) est installé et un dispositif mécanique permet de déplacer un microphone en tout point situé au-dessus de ce plan ; ce dernier étant non couvert ou couvert avec un matériau absorbant, de caractéristiques acoustiques connues \( (k_2, \rho_2) \).

La pression acoustique et la phase de cette dernière par rapport à une référence sont relevées de façon continue, juste au-dessus du matériau, à partir de la source. Ces quantités sont numérisées, introduites dans une mémoire et traitées par une calculatrice à l'aide d'un programme convenable.

Résultats et conclusions

Dans le champ proche, on observe une loi de décroissance en \( 1/r \) et dans le champ lointain une loi en \( 1/r^2 \) comme la théorie le prévoit. Ces résultats expérimentaux et théoriques montrent un accord satisfaisant.

Références

(1) Théorie établie et présentée à ce congrès par M. P. Filippi (L.M.A)
A TEST SYSTEM FOR THE MEASUREMENT OF ABSORPTION
COEFFICIENTS USING A SCALED REVERBERATION CHAMBER

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Most commercially available acoustical test equipment is for audio frequency use. Such equipment is not usually required to respond to transient amplitude changes more rapid than those to which the human ear can respond. However, with acoustic scaled models, ultrasonic frequencies up to 100kHz may be used, and sound level slew rates of more than 1000 dB/sec may occur. Special attention to the measurement system is therefore required.

Measurements are made using a Perspex chamber, which is a 4th scale model of the ISO (1) recommended chamber. To avoid the limitations imposed by anomalous ultrasonic sound absorption of air, it is intended to use pure oxygen-free nitrogen in the chamber. The velocities of sound in nitrogen and air differ by only 2%, and the substitution should not significantly affect the measurement of non-resonant absorbers.

Some of the sound generation and measurement instrumentation of the chamber is standard, (see diagram). However, a digital pulse synthesizer is being designed which will generate band-filtered impulses for exciting the chamber (2). A microphone preamplifier (3), having an equivalent noise level of 41 dB SPL, is being built. The standard measuring amplifier/rectifier cannot respond quickly enough to the fast decay rates which occur in the chamber at ultrasonic frequencies. An RMS rectifier (4), is being constructed using integrated multiplier circuits. It is proposed to control excitation and data acquisition from a Hewlett-Packard Model 9815A minicomputer. The equipment has been designed accordingly.

Using the system outlined it is hoped that the rate of testing can be increased, and that various methods of data analysis (5) can be easily implemented.

REFERENCES

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(4)C.G.Wahlmann, Brul and Kjaer Technical Bulletin (1958) 3
(5)T.E.Vigran, S.Sørsdal, Jn. of Sound and Vibr. (1976) 48, 1
AUTOMATIC MEASUREMENT OF ACOUSTICAL PARAMETERS USING A REAL-TIME ANALYZER AND A COMPUTER

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INTRODUCTION

A Real-Time Analyzer operating in conjunction with a Computer or a Desk-Top Calculator forms what in many respects is a universal system for the measurement of acoustical parameters. In the following, use of such a system to make automatic measurements of reverberation time and transmission loss is described.

INSTRUMENTATION REQUIREMENTS

It is the measurement of reverberation time which places the most stringent requirements of the instrumentation to be used. The Real-Time Analyzer must have a response time fast enough to enable it to follow a decay which might last a few hundred milliseconds. It must have sufficient dynamic range to give a reasonable measurement window. Finally, it must be able to transmit enough information to the calculator or computer to allow the reverberation time at each frequency of interest to be calculated with the required degree of accuracy.

The calculator or computer, on the other hand, must be capable of accepting the information at the rate at which it is transmitted, and must be able to store it until such a time that it can be transferred onto a secondary storage medium such as magnetic tape. (The amounts of data involved, especially in the measurement of long reverberation times, can become quite substantial).

MEASUREMENT TECHNIQUES

Given instrumentation which fulfils the above requirements, measurement of reverberation time, either using the impulse or the interrupted noise method, becomes possible. As the decay proceeds, the analyzer transmits spectra, at preset time intervals, to the computer or calculator. On completion of the decay, the computer or calculator holds in its memory a landscape of level against frequency against time, from which the reverberation time as a function of frequency can be calculated. Many decays can be so treated, and a statistical analysis performed on the results.

Measurement of transmission loss is somewhat simpler, since only two spectra are required, that in the transmitting room and that in the receiving room. However, even here, the Real-Time Analyzer can simplify matters, by allowing automatic spatial averaging of the sound field in each room, over several microphone positions. Measurement of transmission loss then becomes almost direct, the computer or calculator only apply corrections for the room response, etc., to the final result obtained.
Recent transmission loss measurements of plane sound waves in circular pipes with a sharp 90° bend carried out by G.F. KUHN and C.L. MORFEY [1] resulted in a much lower transmission loss of sound radiated by the pipe walls than the transmission loss predicted for pure breathing mode excitation of the pipe. This effect was most pronounced in the range $0.1 < k_0 x < 0.4$ ($k_0$: acoustic wavenumber, $c_0$: wave speed in air, $x$: pipe radius), where the radiated power exceeded the breathing mode prediction by up to 20 dB. The main source of this low-frequency sound (wavelength greater than pipe diameter) appears to be bending waves generated in the pipe by plane sound waves incident on the 90° pipe bend.

A technique has been developed for measuring the bending motion of the pipe under acoustic excitation, using a laser Doppler velocimeter operating on the Michelson interferometer principle. This makes it possible to study the bending-wave response of bent pipes to an internal plane-wave input, and thus to explain the difference between the actual transmission loss and the breathing mode predictions.

II. Noise and vibration

E. Noise Sources
INTRODUCTION

The power spectra of radiated sound from the interaction of a jet exhaust with wing and flap surfaces, as in the case of aircraft powered-lift systems, contain two or more broad peaks. These phenomena lead many to believe that each of the dominant peaks are associated with individual sources. In addition to the contributions from different sources, the reflection and refraction of any source with broad spectrum could cause reinforcements and cancellations at certain frequencies. Thus, experiments were conducted to study the effect of a surface on the spectra of radiated sound.

BACKGROUND AND EXPERIMENTAL MODEL

The broadband sound source near the finite rigid surface can be reflected and refracted depending on the location of the source relative to the surface, and shape and size of the surface. Ffowcs Williams and Hall(1) have shown that the presence of a sharp edge in the vicinity of a quadrupole source can amplify the radiated sound. In order to evaluate the effect of rigid surfaces on the sound field of flow/surface interaction noise, experiments were conducted using a subsonic cold jet exhausting on the upper surface of a wing. The flow is turned along the surface using the coanda phenomena. Radiated sound was measured with and without sound absorbing material on various reflecting surfaces of the experimental model.

RESULTS AND DISCUSSION

The power spectra of the far-field sound with and without sound absorbing material are shown in Figure 1. The humps and dip in the frequency range of 500 to 2000 Hz are eliminated when sound absorbing material is used. These results indicate that the presence of a rigid surface can change the spectra of the radiated sound generated by the jet flow over a wing and flap surface. Therefore, it is important to consider the effect of rigid surfaces in the vicinity of a noise source to describe the radiated sound field.

REFERENCES

INFLUENCE OF FLOW CONDITIONS AT THE NOZZLE EXIT ON JET NOISE IN FLIGHT

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At the present time there is still a controversy regarding the effects of forward aircraft velocity on the radiated noise from jet flows under flight conditions. Intuitively, one might expect the jet noise under flight conditions to be lower than that for a stationary power plant. The reason for this is that shear in the jet mixing layer is less under flight conditions. This results from a lower relative velocity of the jet with respect to its environment. However, jet noise measurements from aircraft in flight have shown little or no reduction close to $\theta_f = 90^\circ$, where $\theta_f$ is the angle between the observer and the jet axis referenced to the inlet of the engine. In fact some flight results show an increase in the noise level in the forward quadrant. Extensive experiments have been performed in the past in order to arrive at an understanding of the reasons for the disagreement between predictions and flight measurements. Experimental results presented by the authors (AIAA paper 76-558, 3rd Aeroacoustic Conference, Palo-Alto, CA, 1976) pointed out that the mixing of the outer boundary layer flow with the jet flow will result in a transformation region in the jet flow where much larger shearing stresses may occur than had been accounted for in previous prediction models. The present study shows that this transformation region where the separated boundary layer transforms to a fully developed shear layer profile plays a key role in the noise radiated to the far-field under flight conditions. The main purpose of this study was to determine the details of the jet structure in this transformation region and the influence of the initial flow conditions at the nozzle exit on this region under simulated flight conditions. Measurements of the fluctuating and of the mean velocities were made in the jet flow along with measurements of the radiated noise. Jet noise measurements were also made in which the turbulent mixing of the engine cowl boundary layer flow with the jet flow was enhanced.

Based on the velocity fluctuations in the jet mixing layer under flight simulation conditions, it was deduced that the outer boundary layer flow conditions at the nozzle exit have a pronounced effect on the jet noise sources in flight. The results show that for a fixed flight velocity, the changes in the outer boundary layer flow were noticed as far as three to four diameters downstream of the nozzle exit. The noise generated in this region is of relatively high frequency. Since this relatively high frequency noise dominates the spectrum close to $\theta_f = 90^\circ$ and in the forward quadrant, the initial flow conditions influence this segment of the radiated jet noise field the most. It is believed that the noise generated in this region is almost like that of a static jet. Therefore, little or no reduction of radiated jet noise in the forward quadrant would occur under flight conditions. In fact, the modification of the acoustic pressure field caused by the forward velocity may result in an increase of the noise in the forward quadrant with increasing flight Mach number as has been observed in flyover jet noise measurements.

Measurements obtained during the present study further show that the influence of the initial outer boundary layer flow on the shearing stresses was relatively insignificant downstream of the transformation region under simulated flight conditions. This part of the jet flow contributes largely to radiated jet noise in the rearward quadrant close to the jet axis. As one would expect, therefore, the jet noise and its spectrum in the rearward quadrant was relatively insensitive to the initial flow conditions at the nozzle exit.

Results also indicate that tripping the outer boundary layer flow caused a reduction in radiated jet noise. A tripped outer boundary layer significantly modified the primary jet flow in the vicinity of the nozzle exit. Tripping enhanced the jet mixing and this may in turn shorten the length of the major noise producing portion of the jet.
It has been suggested that the large-scale structures in the turbulence of free jet flow are the most efficient sources of the jet noise. These structures can be detected, e.g., by cross correlation of the turbulently fluctuating pressure in the jet [1]. On the other hand, it turns out that small disturbances of the jet exit velocity are strongly amplified on the way downstream by the instability of the jet flow [2,3]. Therefore, it is commonly assumed that the large-scale structures are instability waves which are excited statistically by the turbulence itself.

The present investigation consists of two parts:

1) The cross spectrum of the turbulent pressure fluctuations is compared to the pressure of artificial instability waves to find out whether the assumption mentioned above is justified.

2) The sound radiation from artificial instability waves was measured to decide whether a reasonable amount of the jet noise originates from the hypothetical instability waves in the jet turbulence.

The investigation was conducted in a circular jet and was restricted to the axisymmetric component of the turbulence. The main results can be summarized as follows: A comparison between the cross spectrum and the pressure distributions of the instability waves is possible only if one of the two microphones at the cross correlation measurements has a fixed position. Therefore it was decided to place one microphone on the jet axis at that point where the pressure of the artificial instability wave is maximum, at the frequency regarded. Thus, only the axisymmetric component of the turbulence is measured; furthermore, in the case of this special measuring condition, the considered component of the cross spectrum should be most similar to the pressure distribution in the artificial instability wave, provided the turbulence is mainly composed of instability waves. In fact, this similarity has been observed regarding the amplitudes and the phases of the cross spectrum and of the instability waves.

The instability waves are usually excited by a loudspeaker in the settling chamber. However, the loudspeaker itself radiates to the far field far more efficiently than the instability wave. Therefore, the instability waves are excited by axial oscillations of the trailing edge of the nozzle, in the case of the present investigation; 20 dB less sound is radiated from this mechanism than from the loudspeaker. Nevertheless, the radiation from the instability wave is even weaker than from the oscillating tip of the nozzle. A rough estimation as based on this result, yields that, at most, 10% of the jet noise originates from instability waves which, on the other hand, largely cause the turbulent pressure fluctuations in the jet flow.

ZUM EINFLUSS DER GROSSRAUMIGEN TURBULENZSTRUKTUR AUF 
DIE SCHALLENTSTEHUNG IN GASSTRAHLEN

Stiewitt, H. 
Grosche, F. R. 

1. EINLEITUNG

Den Theorien zur Schallentstehung in turbulenten Gasstrahlen lag bisher zumeist 
die bereits auf M. J. Lighthill zurückgehende Annahme zugrunde, daß die Schallab-
strahlung durch viele voneinander unabhängige Turbulenzballen erfolgt, die klein 
gegenüber dem Strahlendurchmesser sind. Neuere Untersuchungen haben jedoch ge-
zeigt, daß die turbulenten Schwankungen im Strahl über Entfernungen in der Größenord-
nung von mehreren Strahldurchmessern eine geordnete Struktur aufweisen, siehe z. B. 
[1, 2]. Die von A. Michalke [3] entwickelte Theorie zeigt, daß vermutlich gerade diese 
kohärenten Schwankungen des Strömungsfeldes die Schallerzeugung im Freistrahl be-
stimmen. Die vorliegende Untersuchung soll zur Klärung dieser Frage beitragen.

2. EXPERIMENTE

In der DFVLR-AVA wurde eine Methode entwickelt, um die Schallsignale zu untersuchen, die von einem 
egen umgrenzten Teilvolumen der Strömung abge-
strahlt werden. Dabei fokussiert ein großer Hohl-
spiegel die von diesem Teilvolumen ausgehenden 
Schallwellen auf ein außerhalb der Strömung ange-
brachtes Mikrofon [4]. Bei den hier beschriebenen 
Experimenten werden mit zwei Hohlspiegel-Mi-

crofonanordnungen gleicher Bauart die Schallsig-
nale aufgenommen, die von zwei verschiedenen 
Punkten im Freistrahl ausgehen, s. Bild 1. Die 
Signale werden miteinander korreliert. Wenn die 
Hypothese richtig ist, daß der Strahllärm im we-
sentlichen durch kleinräumige, statistisch voneinander unabhängige Turbulenzelemen-
ten erzeugt wird, sollte die Korrelation zwischen den Schallsignalen rasch mit zuneh-
mendem Abstand $\Delta x$ abklingen. Wenn dagegen die großräumige Turbulenzstruktur 
im Strahl die Schallerzeugung bestimmt, sollten die Schallsignale noch bei erheblich 
größeren Abständen zwischen den betrachteten Punkten eine starke Korrelation aufwei-

sen.

3. ERGEBNISSE

Nach Erprobung des Messverfahrens an einer annähernd punktförmigen Schallquelle 
wurden Messungen entsprechend Bild 1 an einem kalten Freistrahl bei der Ausström-
Machzahl 0, 8 durchgeführt. Der Korrelationskoeffizient fiel bei diesen Versuchen mit 
wachsendem axialen Abstand $\Delta x$ der beiden Spiegel etwa in dem Maße ab, wie bei An-
nahme kleinräumig korrelierter Schallsphäre zu erwarten wäre. Eine genauere Ana-
lyse der Meßdaten ist jedoch noch erforderlich und wird zur Zeit durchgeführt.

4. SCHRIFTTUM

INFLUENCE OF STATIC PRESSURE ON NOISE GENERATION OF AERODYNAMIC SOURCES

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INTRODUCTION

Noise generated by solids moving in gas were investigated experimentally in a wide range of static pressures values (760 Torr ... 1 Torr) using different aerodynamic sources. These sources are

- an aerodynamic reference sound source from the Electricité de France (fan system)
- a siren
- a rotating pair of bars (bar diameter 10 mm). Details of this source see (1).

EXPERIMENTAL TECHNIQUES

The sound power of the investigated sources were measured in a reverberant room according the methods given in IS 3741/42. The reverberant room is located in a boiler the atmosphere (gas) of which can be varied by changing static pressure and changing the gas (air, CO₂, ...) itself.

RESULTS

The influence of the static pressure $p_s$ of air on the sound power output $W$ of the investigated aerodynamic sound sources can be described by

$$ W = C \cdot p_s^n $$

respectively for the levels by

$$ L_W = n \cdot 10 \cdot \log \frac{p_s}{p_{00}} + \text{const.} $$

whereby $p_{00}$ is a reference static pressure, e.g. $p_{00} = 760$ Torr.

The exponent $n$ depends significantly from the center frequencies of the considered bandwidth and from the special mechanism typical for the specific source.

- The noise of the aerodynamic reference sound source has exponents $n$ between 2.2 and 3.0 regarding the range of 500 Hz ... 8 kHz. The lower frequencies are correlated with lower exponents, the higher frequencies with the greater $n$-values. This means the reduction of static pressure leads not only to lower noise intensities but also to a shifting of the relative distribution of noise energy over the frequencies (spectrum).

- The noise of the rotating bar system producing KARMAN vortices has exponents $n \approx 3.5$ for frequencies of 500 Hz ... 700 Hz. These exponents are significantly higher than those of the fan noise regarding the same range of frequencies. In the range of investigated Reynolds Numbers the Strouhal Number deviates only within accuracy of the measurement.

REFERENCES

Sound generated by low Mach-number fluid motion is usually determined by

\[ p(x, t) = \int G(x, y, t - \tau) \frac{\partial^2 T_{ij}}{\partial y^i \partial y^j} \, d^3 y \, d\tau \]

where \( G(x, y, t - \tau) \) is the Green's function of the wave operator, the \( u^i \) are the velocity components, \( \rho_0 \) is the ambient density, \( p \) the pressure. If the flow is approximated by an incompressible one, the \( T_{ij} \) fulfill Helmholtz's vortex equation

\[ \frac{\partial}{\partial \tau} \text{curl} \, \mathbf{u} + \text{curl} \, \mathbf{T} = 0 \]

One can use this equation to rewrite eq. (1) in terms of the flow vorticity

\[ p(x, t) = \int G(x, y, t - \tau) \frac{\partial}{\partial \tau} \text{curl} \, \mathbf{u} \, d^3 y \, d\tau \]

provided a function \( G \) with \( \nabla_y \times G = \nabla_y \mathbf{G} \) can be determined. The integrability condition of these equations requires \( \nabla_y^2 G = 0 \). As \( G \) fulfills the wave equation

\[ \nabla_y^2 G = \frac{1}{a^2} \frac{\partial^2}{\partial \tau^2} G - \delta(\mathbf{a} - \mathbf{y}) \]

\( G \) can be determined for small Mach-number in the far field. In equation (2), \( p \) is given by an expression, which is linear in the flow velocity, while in eq. (1) \( p \) is given by a quadratic expression. For compact quadrupoles in free space, \( G \) fulfills

\[ G = \frac{1}{2\pi} \frac{\partial}{\partial t} \left( t - \frac{x}{a^2} \right) \left( \frac{\mathbf{x} \times \mathbf{y}}{a^2} \right)^2 \]

and \( p \) can be rewritten from eq. (1) as

\[ p = \frac{\rho_0}{12\pi a^5 \mathbf{x}^3} \left( \int \frac{\partial^3 \text{curl} \, \mathbf{u}}{\partial t^3} \, \mathbf{x} \cdot \mathbf{y} \, d^3 y \right) + x \int \frac{\partial^2 \mathbf{u}^2}{\partial \tau \partial t} \, d^3 y \]

where \( t \) is to be taken at retarded time. The last term vanishes because of the incompressible energy equation. As a second example a random distribution of quadrupoles is considered. If the correlation length is small compared to the acoustic wave length one finds, that the 4-th spherical harmonic of the average acoustic intensity \( I_4 \) at frequency \( \omega \) is given by

\[ I_4 = -\frac{\rho_0 \omega^6}{16 \pi a^2 \mathbf{x}^2} \int \int (y - z)^4 \mathcal{Q}(\mathbf{y}, \mathbf{z}, \omega) \, d^3 y \, d^3 z \]

where \( \mathcal{Q} \) is the vorticity correlation \( \mathcal{Q} = \langle \text{curl} \, \mathbf{u}(y) \rangle \cdot \text{curl} \, \mathbf{u}(z) \rangle \) and \( P_4 \) the 4-th Legendre polynomial. Eq. (3) shows that \( I_4 \) is related to a quadratic expression of the random velocity field, while the intensity from eq. (1) would be related to biquadratic expressions.
INTRODUCTION

The C.E.G.B. spends a considerable amount of money on silencing steam discharges from safety valves, dump vents, trip vents etc. A comprehensive program of tests has recently been completed to establish: 1, the acoustic power output of high intensity steam discharges, 2, the fluid dynamic and acoustic performance of perforated cylindrical diffusers and shrouds, such as are used in steam discharge silencers, so that their design may be optimised.

THEORETICAL BASES

The sound pressure levels and spectral content of the unsilenced jet (at various angles to the jet axis, at flow rates up to 12.6 kg/s) were compared with those calculated from recent information on the generation of noise from aircraft jet engines, (1).

Various cylindrical perforated diffusers were then mounted on the outlet pipe to assess their effect. The parameters which were varied were hole size (4.2 and 12.7 mm diameter), area ratio (1.5, 2.0 and 2.5) and percentage open area (5, 10 and 15%).

EXPERIMENTAL TECHNIQUES

An array of ten microphones, mounted at angles from 22° to 90° to the jet, 5 m from the outlet, was used to measure the sound levels and hence the acoustic power output of each configuration. Signals were recorded using a 14 channel F.M. tape recorder for subsequent playback in the laboratory through a Real Time Analyzer. The test rig was fully instrumented for temperature, pressure and flow rate measurement, records of which were stored using a UV chart recorder. A computer was used to obtain spectra, directivity plots and sound power levels, and to process the fluid dynamic data.

RESULTS AND CONCLUSIONS

1. There was reasonable agreement between the measured open pipe sound levels, and those calculated from Reference 1, with no evidence of shock associated noise.

2. The acoustic efficiency for the regenerated noise from the cylindrical diffusers increased with increasing hole size and percentage open area, whereas the attenuation of upstream noise was reduced.

3. Changes in Area Ratio from 1.5 to 2.5 produced no significant effects.

4. The addition of a shroud surrounding the diffuser modified the acoustic efficiency for the regenerated noise and increased the attenuation of upstream noise.

5. For maximum attenuation to upstream noise, and minimum size, weight, cost and regenerated noise a diffuser silencer should have 4.2 mm perforations, an Area Ratio of 1.5 and less than 10% open area, and should be surrounded by a shroud.

REFERENCE

(1) Proposed SAE/ARP 876 Gas Turbine Exhaust Jet Noise Prediction.
THE GLIDER - A VALUABLE TOOL IN AIRFRAME NOISE RESEARCH

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INTRODUCTION: The successful quieting of aircraft propulsion systems has made airframe noise a focal point in design considerations for future aircraft. Airframe noise - the noise generated purely by the passage of air past the airframe - is considered the "Ultimate Barrier", that cannot, without major effort, be reduced much further. In essence there are two major aircraft components that are responsible for airframe noise: (a) the wings, including flaps and slats, and (b) the undercarriage; interaction between these components may constitute further combination sources. Theoretical prediction of airframe noise from basic principles is extremely tedious because of the often extremely complex geometries of aircraft components, such as landing gears. For these reasons much effort is devoted to the experimental compilation of airframe noise data, both through full-scale and model testing.

TECHNIQUES: Experimental tools that are commonly used are (a) wind tunnels, (b) spinning rigs, (c) tracked vehicles, (d) remotely piloted model planes, or (e) full-scale aeroplanes operated in special modes. All these experimental tools have their specific advantages and disadvantages. Recently, the glider has been introduced as another research tool for the study of airframe component noise (1;2).

RESULTS: Scaled models (1:12 and 1:4) of nose- and main landing gears were attached to the wings of the Braunschweig Akaflieg SB-10 glider (3), and their acoustic farfield characteristics determined (Fig.1). Results are used to develop a landing gear noise prediction scheme. Since landing gear noise results from the periodic or random shedding process of vortices off the various gear components, the measurement of fluctuating surface pressures on components reveals important information on the relevant source mechanism. Fig.2 shows surface spectra on the aft wheel of a main gear model flown on the SB-10 glider. Flush-mounted sensors on a flap-section of the SB-10 wing were used to study the surface pressure field characteristics. These unsteady pressures at and near the trailing edge constitute the source mechanism of trailing edge noise, which is presently considered the dominant airframe noise component.

CONCLUSIONS: Aerodynamically clean gliders with inherently low self noise can successfully be used for airframe noise related studies providing very realistic test conditions. The inherent drawbacks, that characterize all flyover tests, are more than compensated for by the cleanliness of data, say in comparison with tunnel tests, where tunnel self noise, incident turbulence etc. may adversely affect generating mechanisms and be responsible for insufficient signal-to-noise ratios.

Two-dimensional statistical vortex model for describing of one of the mechanisms of sound generation by turbulent boundary layer on nonuniform flexible surface (plate, shell) in slightly compressible medium is suggested on the basis of academician M.D. Millionshchikov's presentation [1] on the turbulent boundary layer as a totality of vortices, rolling on the surface of viscous sublayer. Turbulent boundary layer is presented in the form of totality of noninteracting vortex filaments. Vortices similar on force form straight chains, parallel to the boundary, in which the distance between vortices is a random value. Each chain is transferred with constant velocity, corresponding to the average velocity of turbulent boundary layer. Sound radiation by single vortex filament, interacting with plane nonuniform boundary, formed for example, by two different flexible plates, is determined initially. Acoustical nonstationary problem is solved by the method of matched asymptotic expansions while for the solution of nonlinear hydrodynamic problem the method of successive approximations is used [2].

In the framework of the considered vortex model and suggestions on the structure of turbulent boundary layer, sound radiation by the model is the stationary random impulse process.

Considering described vortex model over the plane boundary, consisting of two flexible plates [2], the dependences of average flow of acoustical radiation power on the following parameters: Mach number - $M^2$, $M^5$, plate thickness - $h^{-2}$, $h^{-3}$, medium density - $\rho^3$ are obtained. This is in a rather good agreement with the experimental investigations of sound radiation by turbulent boundary layer on flexible plates [3], where dependences on Mach number - $M^5$, $M^6$, thickness - $h^{-1}$, $h^{-1.6}$, and density - $\rho^{2.6}$ are derived.

Obtained coincidence shows that even such simple vortex model in general terms apparently describes properly the real processes of noise generation in turbulent boundary layer on nonuniform boundaries in slightly compressible medium.

REFERENCES:
INTRODUCTION

Le Comité ISO TC 70/SC 6 - Turbines à Gaz - a demandé au Comité ISO TC 43/SC 1 - Bruit - de préparer une norme décrivant les méthodes de mesure du bruit émis par les turbines à gaz. Le Groupe de Travail 14 a été formé en 1975 dans le but de préparer trois documents :

- une méthode simple de mesure des niveaux de bruit extérieurs, en des points déterminés, situés au niveau du sol, autour d'une installation complète, dans le but de vérifier si l'équipement répond aux prescriptions stipulées dans la commande.

- une deuxième méthode, plus élaborée, en vue de déterminer la puissance acoustique rayonnée par les sources de bruit individuelles (par exemple : entrée d'air, cheminée, bâtiment, auxiliaires), si l'origine des différents bruits doit être recherchée et analysée et pour vérifier l'efficacité des dispositifs prévus pour réduire le niveau de ces différentes sources de bruit.

- une troisième méthode pour la mesure des bruits intérieurs, aux endroits où le personnel d'exploitation est appelé à séjourner pendant de longues durées.

Un projet du premier de ces trois documents, intitulé "Méthode de spécification et de mesure des niveaux de bruit autour des installations à turbines à gaz" est prêt et doit circuler parmi les comités membres de l'ISO.

METHODE DE MESURE

La présente communication donne quelques détails sur le projet et explique les raisons pour lesquelles certaines options ont dû être prises. On considère successivement les différents points suivants :

- Environnement acoustique affectant les niveaux de bruit mesurés,
- Description de l'instrumentation nécessaire,
- Conditions de fonctionnement de la turbine à gaz,
- Spécification des niveaux de bruit : pondération A et C, ou niveaux dans 9 bandes d'octave, mesurés à une distance spécifiée qui est généralement de 100 m à partir du périmètre de l'installation complète.
- Détails concernant les positions du microphone et les caractéristiques de l'appareil de mesure,
- Calcul des niveaux de bruit moyens.

Les lecteurs et auditeurs de la présente communication sont invités à présenter leurs remarques, suggestions et critiques permettant d'améliorer le document.

REFERENCE

NEW RESULTS ON AERODYNAMIC SOUND GENERATION BY MEANS OF SINGULAR PERTURBATION METHODS

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The evaluation of the sound power generated by unsteady subsonic flows of finite extent requires the knowledge of the four-fold correlations of the velocity field ([1]). Currently, the explicit computation or the complete measurement of these correlations do not seem to be practicable. Therefore, in the present paper an alternative procedure is developed by means of singular perturbation methods.

An integral representation with a "linear" source distribution is employed to express the pressure field \( p \) in the hydrodynamic flow region (inner flow region) where the flow quantities can be approximated by their corresponding incompressible values ([2]). The ordinary Green's function \( G(x, y) \) is replaced by a so-called "Green's vector function" \( \tilde{G}(x, y) \), which is determined by the relation

\[
\nabla_y \tilde{G}(x, y) = \nabla_y \times \tilde{G}(x, y) \quad \text{with} \quad \Delta G = S(x) \quad (W. Möhrling, private communication).
\]

The explicit solution for the pressure field in the inner flow region runs

\[
p + \frac{\mu_2^2}{2} = \int \tilde{G}(x, y) \nabla(y \times \tilde{G}(x, y)) \, d^3 y, \quad \nabla_x = -\nabla_x \frac{1}{3} \left[ \int G(x, y) \, dy_1, \int G(x, y) \, dy_2, \int G(x, y) \, dy_3 \right],
\]

(\( \mu \) velocity, \( t \) time; all variables are in nondimensional form) where viscosity effects and entropy fluctuations are neglected.

The far field approximation of this solution which is needed to evaluate the corresponding sound fields (with the help of the method of matched asymptotic expansions) is obtained in the form

\[
\lim_{|x| \to \infty} p = -\frac{1}{3} \left[ \frac{\partial}{\partial x_1} G(x) \int \left[ \nabla_x \frac{\partial}{\partial t} \left( \nabla x_1 \right) \right] \, d^3 y + \frac{\partial^2}{\partial x_1^2} \tilde{G}(x) \int \left[ \nabla_x \frac{\partial}{\partial t} \left( \nabla x_1 \right) \right] \, d^3 y + \ldots \right].
\]

This representation provides new insights into sound generation mechanisms:

(i) A dipole-like sound distribution is only possible if at least one term of the form \( \int y_k \frac{\partial}{\partial t} \left( \nabla x_1 \right) \, d^3 y \) is not zero, i.e. if the centre of vorticity fluctuates with respect to time \( t \).

(ii) Quadrupole-like sound fields will be important if the first terms of Eq. (1) are zero but at least one term of the form \( \int y_k y_p \frac{\partial}{\partial t} \left( \nabla x_1 \right) \, d^3 y \) is not zero, e.g. if the second moments of the vorticity are time dependent.

These results are illustrated by means of simple two-dimensional flow models where solid boundaries are included.

REFERENCES

SCATTERING BY DENSITY GRADIENTS

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INTRODUCTION
At low speeds hot jets make more noise than cold jets, and empirical correlations of this excess noise have been obtained which presume the existence of monopole sources in the jet (1,2). It is known that dipole radiation results from the scattering of quadrupole near-fields by density inhomogeneities, but some analyses suggest that the radiation efficiency can be increased still further to that of monopoles (2, 3). We shall show that no monopole source results from scattering by density gradients.

ANALYSIS
A compact sphere vibrating without deformation at a density interface (see figure 1) constitutes an acoustic dipole. The far-field fluctuating pressure is obtained by the method of matched expansions:

\[ p = \frac{\omega^2 a^2 U_0 \sin \theta \cos \omega(t-r/c)}{2\pi r} (y>0) \]

\[ p = \frac{\omega^2 a^2 U_1 \sin \theta \cos \omega(t-r/c)}{2\pi r} (y<0) \]

There is no monopole splash.

Two compact spheres situated close to and on opposite sides of a density interface constitute a quadrupole source only if the components of the forces on the spheres perpendicular to the interface are equal in magnitude but opposite in direction and the components of the forces per unit fluid density parallel to the interface are also equal but opposite. The method of matched expansions shows that in other cases dipole rather than quadrupole efficiency is achieved.

The reciprocal theorem demonstrates that these results hold for compact bodies of arbitrary shape in fluids with varying density and speed of sound. The pressure at \( r_0 \) due to small vibrations with frequency \( \omega \) of a rigid body in an inhomogeneous medium at rest is given by

\[ \int_S \frac{\mathbf{p}(\mathbf{x}) \cdot \mathbf{U}}{q} \, dS(\mathbf{x}), \] where \( \mathbf{U} \) is the velocity of the body and \( S \) its surface. \( \mathbf{p}(\mathbf{x}) \) is the pressure in the reciprocal problem when a point source of strength \( q \) and frequency \( \omega \) is placed at \( \mathbf{r}=\mathbf{r}_0 \) in the same medium but with the rigid body stationary. Variations in \( \mathbf{p}(\mathbf{x}) \) across \( S \) are solely responsible for the non-vanishing of this integral, and these variations occur only because of differences in retarded times. Consequently there is no monopole contribution to the sound radiated by the vibrating rigid body.

REFERENCES
Introduction

A series of measurements developed by the author in some ships is shown in this article. The scope is to improve and bring up to date the knowledge of this question in order to get a better level of comfort in some rooms of the ship. The convenience of simultaneously taking measurements of vibrations is considered, but lack of sufficient data forces the author to leave this question for a future job.

Theoretical Bases

The acceptability of a noise level may be expressed in dB(A), but everyone knows that on some occasions this is not enough, above all in those cases on which low frequencies are predominant, as is normal in ships. That is why the room noise levels have been measured in octaves and expressed according to ISO noise rating curves. The procedure followed in the measurement was that stated in DIN-80061 and the instrument used was a precision sound meter which complied with IEC-179 (DIN-45633, pages 1 and 2).

Experimental Techniques

The ships measured are all cargo vessels of the same type, with engine and accommodation aft and built at the same shipyard. All measurements were taken in the same condition of ballast and speed.

Results and Conclusions

The graphs show the results for 15 ships, with maximum and minimum figures for six representative rooms. Each graph shows the recommended NR value for the room considered. The conclusions are as follows:

- The engine room workshop is not acceptable; this is true of all ships.
- The galley is acceptable, only without a ventilation fan.
- The lowest and aftermost cabin is the worst and, therefore, unacceptable. Other cabins have no problems generally. Solutions and other comments are made in this article. Limits for different rooms in the ship are shown.
GERÄUSCHENTSTEHUNG UND LÄRMMINDERUNG AN GASBRENNERN FÜR INDUSTRIEOFEN

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LÄRMBELÄSTIGUNG DURCH INDUSTRIE-GASBRENNERN

Aufgrund des häufigen Einsatzes von Industriegasbrennern in der Eisen- und Stahlin-
dustrie bilden sie eine wesentliche Quelle der Lärmbelästigung. Da die Brenner oft
in nicht abgeschlossenen Öfen betrieben werden, können in unmittelbarer Nähe der
Brenner am Arbeitsplatz Schalldruckpegel von über 120 dB(A) auftreten.

GRUNDLAGEN DER GERÄUSCHENTSTEHUNG AN GASBRENNERN

Die wesentlichen Lärmursachen beim Betrieb von Gasbrennern sind folgende: Strömungs-
geräusche, Resonanzen und Pulsationen im Brenner und in den Zu- und Ableitungen, Ge-
räusche von Hilfsaggregaten sowie als Hauptlärmquelle die Verbrennung selbst. Wäh-
rend die Strömungsgeräusche und die anderen Nebengeräusche sich z.T. berechnen las-
sen, sind die Ursachen für das Verbrennungsgeräuschenoch weitgehend ungeklärt. Die
Verbrennung selbst sowie der zeitliche und geometrische Ablauf in der Flamme beein-
flussen die Geräuschtentstehung.

EXPERIMENTELLE UNTERSUCHUNGEN

Um dem Phänomen der Verbrennungsgeräusche näher zu kommen, wurden verschiedene Typen
von Hochgeschwindigkeitsbrennern sowohl im Freistrahl als auch in einem Ofenraum mit
varialher Geometrie untersucht. Im wesentlichen wurde der Einfluß folgender Vers-
suchsparameter auf die Geräuschabstrahlung ermittelt: Luftzahl λ, Gasart, Wärmelei-
stung, Ausströmgeschwindigkeit, Gesamtmpuls und Impulsstron-
dichte. Im Bild 1 sind die Schalldruckpegel für ver-
schiedene Brenner über der
durchgesetzten Brennerlei-
ustung aufgetragen. Am ge-
enschlossenen Ofenraum werden
etwa 30 dB geringere Schalldruckpegel gemessen als am
Freistrahl.

LÄRMMINDERUNGSMASSNAHMEN

Aus den experimentellen Un-
tersuchungen ergaben sich
zwei primäre Lärmminderungs-
maßnahmen. Einmal kann der
Lärm über die Betriebspara-
meter gemindert werden, wo-
bei jedoch auch die wärme-
technischen Ergebnisse geän-
dert werden. So kann z.B. durch die Halbierung der durchgesetzten Brennerleistung
eine Verminderung des Schalldruckpegels um 15 dB erzielt werden. Zum anderen ist es
möglich, über konstruktive Maßnahmen am Brenner die Geräuschabstrahlung zu beein-
flussen. Die dadurch erzielte verbesserte Vermischung und gleichmäßigere Verbren-
nung bewirken eine geringere Geräuschabstrahlung sowie eine günstigere spektrale Zu-
sammensetzung der Brennergeräusche. Durch Beispiele konstruktiver Veränderungen wer-
den Lärmminderungen von ca. 10 dB belegt.
DIE DIE HÄUFIGKEITSVERTEILUNG VON VERKEHRSGERÄUSCHEN BESTIMMenden FAkTOREN

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Einführung

Untersuchungen haben gezeigt, daß die von vielen Autoren angenommenen vereinfachenden Verfahren zur Berechnung der Häufigkeitsverteilung von Verkehrsgläuschen nur unbefriedigende Ergebnisse und im allgemeinen keine Information über die Einflußfaktoren und deren Wirkung liefern. Der vorgesehene Beitrag wird u. a. zeigen, mit welcher Genauigkeit die einzelnen Faktoren bestimmt werden müssen, um eine vorgegebene Zuverlässigkeit bei der Prognose einer Häufigkeitsverteilung zu erzielen, bzw. wie bestimmt werden kann, ob eine gemessene Häufigkeitsverteilung repräsentativ ist.

Theoretische Grundlagen

Die Häufigkeitsverteilung wird durch folgende drei Einflußfaktoren bestimmt:

1. Vorbeifahrtpegel der einzelnen Fahrzeuge (Maximalpegel)
2. zeitliche Abstände zwischen den Momenten der Vorbeifahrt der einzelnen Fahrzeuge (Folgezeiten)
3. zeitliche Änderung des Schallpegels des einzelnen Fahrzeuges (Pegelverlauf) am Beobachtungspunkt.


Das in dieser Veröffentlichung beschriebene Rechenverfahren erlaubt eine Bestimmung der Häufigkeitsverteilung mit einer Genauigkeit von ± 1,5 dB.

Ergebnisse und Schlußfolgerungen

Messungen und Berechnungen von Häufigkeitsverteilungen an Straßen (mit und ohne Bebauung), befahren von PKW bzw. PKW, LKW und Kräder, haben bisher zu folgenden Ergebnissen geführt:

1. Die Prozentpegel \( L_{0.1} \), \( L_0 \) und \( L_{10} \) (bei hohen Verkehrsdichten auch \( L_{100} \)) werden vorwiegend durch die Maximalwerte bestimmt. Für die Prognose ist somit die Berücksichtigung einer Maximalwertverteilung zwingend, wenn eine Genauigkeit von ± 1,5 dB erreicht werden will. Eine Beschreibung durch eine Normalverteilung erscheint ausreichend. Der Vertrauensbereich wird durch die Anzahl berücksichtigter Fahrzeuge gegeben.

2. Die Folgezeiten können nicht durch eine Gleichverteilung beschrieben werden (Fehler der Prozentpegel \( L_{99} - L_{10} \) von 20 - 5 dB). In nur wenigen Fällen reicht eine Näherung durch eine negative Exponentialverteilung (Poisson-Strom) aus. Die Abweichungen der Prozentpegel \( L_{99} - L_0 \) zu gemessenen Situationen liegen zwischen 10 dB und 3 dB. Pulkbildungen müssen durch entsprechende Verteilungsfunktionen erfaßt werden. Die Folgezeiten beeinflussen in besonders starkem Maße die Pegel \( L_{99} - L_{50} \).

3. In gleichem Maße werden die Prozentpegel \( L_{99} \) bis \( L_0 \) durch den Pegelzeitverlauf bestimmt, der nicht durch eine einfache geometrische Beziehung bestimmt ist (auch nicht bei idealen Verhältnissen: keine Bebauung, gerade Straße), wenn eine Genauigkeit ± 2,5 dB gewünscht ist. Es ist mit Fehlern, insbesondere bei \( L_{99} - L_{90} \), von 10 dB und mehr zu rechnen.

Bisherige Untersuchungen lassen Übereinstimmungen innerhalb ± 1,5 dB zwischen Messungen und Rechnungen mit idealen Annahmen bei \( L_{99} - L_0 > 20 \) dB als ein "glückliches" Zusammentreffen der einzelnen bestimmenden Faktoren erscheinen.
The accuracy of traffic noise predictions obtained using the TRRL computer model of traffic noise (1) depends to a considerable extent upon the degree of simplification adopted in categorising vehicles according to their sound output in the traffic stream. The aim is to provide a noise level and speed characteristics for each acoustically different vehicle category that embraces as wide a range of vehicle speeds as possible. This Paper examines and summarises the available data on acoustic classification of vehicles in traffic streams and investigates the improvement in prediction accuracy obtained by increasing the number of vehicle categories considered.

Measurements of speed, noise level and vehicle type have been made in road conditions ranging from fairly congested urban situations with speeds around 20 km/h to free flow on motorways with speeds over 100 km/h. At speeds greater than 50 km/h vehicle noise increases at approximately 9 dB(A) per doubling of speed and the total vehicle population divides most readily into light vehicles with an unladen weight not exceeding 1525 kg and lorries. An analysis of vehicle noise data obtained from vehicles operating at low speeds and in non-free flow conditions revealed that six acoustically separate vehicle categories could be identified and that these classes tend to form three rather than two distinct groups (2). It was found that under non-free flow conditions the noise levels from the noisiest vehicles exceeded the noise levels from the quietest vehicles by approximately 17 dB(A) whereas under free flow conditions this difference was less and amounted to 9 dB(A).

In order to examine the potential of the TRRL computer model to predict urban traffic noise and to investigate the change in accuracy of the model when using different levels of vehicle categorisation, approximate speed and noise level characteristics suitable for input to the computer model for six vehicle categories were constructed. The characteristics have been used to form sets of vehicle characteristics representing 5, 4, 3, 2 and 1 vehicle category divisions of the total vehicle population. A comparison of predictions of L10 for a range of traffic conditions using each of the vehicle category divisions showed that the least number of vehicle categories that could be used without causing a significant change in the accuracy of predictions was three, although it was confirmed that for free flow conditions two levels of vehicle categorisation were adequate. The three vehicle categories were:

- LIGHT vehicles, which include cars, car-based vans and 2 axle commercial vehicles with an unladen weight less than or equal to 3000 kg.
- MEDIUM HEAVY vehicles, which include commercial vehicles with 2 axles and an unladen weight exceeding 3000 kg and buses and coaches,
- and HEAVY vehicles, which include all commercial vehicles with three or more axles.

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INTRODUCTION

Changes in the national railway system have released land for other uses in city centres and suburbs. Previously used as shunting yards and sidings these plots of land can now be used for housing and other developments; however, railway tracks are still in use adjacent to most of these plots of land and the noise and vibration of the railway traffic is relevant to most possible uses of the land.

TYPICAL SIGNAL STRENGTHS

At a site in London, proposed for housing, a 24 hour survey of noise and vibration showed the following results at a position where dwellings would be built:

- number of trains passing: 343
- range of noise levels of passing trains: 67 dBA to 96 dBA (slow response)
- L(eq): 73 dBA (12.00 - 20.00 hrs.)
- L(10): 72 dBA (12.00 - 20.00 hrs.)
- typical maximum ground vibration acceleration values: 3x10^-2 m/s^2

TYPICAL SOLUTION TO NOISE AND VIBRATION PROBLEMS

(a) Noise. Most noise problems can be solved by the use of a single-aspect "barrier" block of apartments placed as close to the railway tracks as possible. This gives maximum barrier-attenuation and can provide acceptable noise levels over the rest of the site and on the shadow face of the barrier block. The track-side wall, and roof, of the barrier block must be designed to a high standard of noise insulation.

(b) Vibration. Levels of vibration tend to increase with height and allowance for this must be made. To achieve low, hardly perceptible, vibration levels within dwellings, and some other buildings such as hospitals, resilient mounting of the buildings may be necessary. In London, vibration is likely to be at a maximum in the 20 Hz/40 Hz region and good isolation at these frequencies is essential.

TYPICAL INSTALLATIONS

In the recent past, installations of housing and commercial development have been completed in accordance with these principles. Publicly owned housing, a privately owned hospital and commercial buildings, have been mounted resiliently with success.
INTRODUCTION

This work is part of a large-scale traffic noise survey in two Belgian cities Antwerp and Brussels and is sponsored by the Belgian Ministry of Health.

The physical measurements and the discussion about the results will be treated during this lecture and in Part II the inquiry concerning annoyance and the correlation with physical quantities will be discussed.

MEASUREMENTS

The physical measurements have been executed during a total of about 1100 hours in the period 1975 and 1976. In Antwerp 42 points all over the city have been measured over 24 hour periods. From the experiences it was concluded that only a few measuring hours are necessary to get sufficient information about traffic noise levels at 25 different places in Brussels.

From each measuring point hourly histograms are determined and from this 11 quantities are computed, namely $L_1$, $L_5$, $L_{10}$, $L_{50}$, $L_{90}$, $L_{95}$, $L_{eq}$, $TNI$, $NPL$ and $\sigma$. All these quantities are determined each hour. The overall $L_{dn}$ values are also calculated.

These results give the possibility to elaborate prognoses about traffic noise. Different parameters play a less or more important role. The vehicle intensity is of course the main factor to determine the traffic noise in town streets. Theoretically one can expect a 3 dB(A) increase for the $L_{eq}$ value by a doubling of vehicles per hour. Taking into account the other parameters which can influence the sound levels, a total variation of $\pm 5$ dB(A) can be superposed on the sound level variation as a function of intensity of the vehicles. Nevertheless if one takes correctly into account those other parameters as there are: vehicle speed, percentage heavy vehicles, street width, street height, ground cover, traffic lights etc..., the agreement between predicted and measured sound levels can seriously be improved. Two of these parameters, ground cover and street width are extensively measured and will be discussed.

An example of the influence of two types of ground cover on the sound level as a function of vehicle intensity is given in Fig. 1.
By the acoustics laboratory of the Catholic University of Leuven an enquiry concerning annoyance by traffic noise was carried out at the same time as the acoustical measurements were made. This research project was performed by order of the Belgian Ministry of Health. In Antwerp 1,319 inhabitants of 42 different streets were interviewed by two collaborators. In Brussels we asked 494 inhabitants of 25 streets for their opinion. The field research work was performed from May until October 1975 in Antwerp and from May until October 1976 in Brussels.

The questionnaire we used principally measures five variables:
Factor 1: Disturbance of diurnal activities by traffic noise
Factor 2: Nocturnal disturbance
Factor 3: General statements on traffic noise
Factor 4: Presumed physical effects of traffic noise
Factor 5: Satisfaction with the environment

Correlation coefficients were computed between the results of the subjects on these five variables on one side, and the results of the acoustical measurements on the other side.
We used following acoustical values: \( L_1 \), \( L_{10} \), \( L_{50} \), \( L_{90} \), \( L_{eq} \), TNI, NPL and \( L_{dn} \).
Finally we computed also the correlation between annoyance and the number of passing cars per hour.

The first factor (Disturbance of diurnal activities by traffic noise) furnishes the highest correlation values, whilst the third factor (General statements on traffic noise) correlates rather poorly.

We tried also to identify some of the variables that lower the correlation value between annoyance and acoustic noise variables.
Recent legislation in the United Kingdom (1) has given local authorities the powers to limit noise emission from construction sites. Controls on the plant or machinery employed, the hours of working and the noise levels emitted into the community are included. A Code of Practice (2), recommends use of the energy-equivalent continuous sound level, Leq, measured over the 12-hour working day as the scale of assessment.

The major sources of noise in road construction have been identified as the diesel-engined plant used in earthworks operation (3). Noise levels from individual plant at 10m measuring distance range from 98 dB(A) for unsilenced motor scrapers to 80 dB(A) for silenced off-highway dump trucks. Measurements at the site boundary have shown that 80% of the one-hour Leq levels for earthworks operations were between 68.5 and 81.5 dB(A), (mean level 75 dB(A)).

A procedure for calculating Leq has been derived so that noise assessment can be carried out at the planning stage (4). The procedure is based on an equation to calculate an Leq level for a particular operation, together with corrections to allow for distance attenuation over hard or soft ground and the screening effects of barriers. The prediction includes the reference process noise energy (Lax), which is calculated by measuring the Leq level, Leq (R), at a reference position (distance R from the process centre) over a short duration t(R), (usually one cycle of the process).

Several methods of noise control have been studied (5). A reduction of site boundary noise of up to 8 dB(A) could be achieved by careful scheduling of operations along the site so that noise events were redistributed in time and location. Further reductions of between 5 and 10 dB(A) could be obtained using barriers to screen the noise. Depending on the site, these measures could add up to 10 per cent to the cost of earthmoving. A further noise reduction could be achieved by reducing the intensity of earthmoving, but very high cost penalties would be incurred by this method.

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EINLEITUNG


Auch die Prognose zukünftiger Straßenverkehrslärmimmissionen wird erleichtert.

GRUNDLAGEN


DURCHGEFÜHRTE UNTERSUCHUNG

Anhand von durchgeführten Messungen praktischer Lärmimmissionen, bei denen gleichzeitig verkehrstechnische, topografische und klimatologische Parameter miterfaßt wurden, werden für die Parameter Fahrzeugmenge, Lkw-Anteil, Entfernung der Straße zum Meßort sowie für weitere Parameter die Kurven der Summenhäufigkeitsverteilung gemessener Immissionen mit den theoretisch abgeleiteten Werten verglichen und diskutiert.

Die dargestellten Kurvenscharren der Summenhäufigkeitsverteilungen eignen sich auch zur Prognose und gestatten dann bei den prognostizierten Immissionen auch die Beurteilung der Lästigkeit der Pegelschwankung, was bei der üblichen Prognose allein des Mittelungspegels nicht möglich ist. Bei Prognosen für Zwecke der Bauleitplanung ist dies vorteilhaft. Es werden Beispiele zur Prognose einzelner und überlagerter Lärmimmissionen angeführt.

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INTRODUCTION
The noise and vibration of 35 locomotives were measured and recorded on the purpose of evaluating the level and quality of noise and vibration and also the respective exposure of the engineers. The study was a subproject of an investigation on the work conditions and health status of locomotive engineers.

THEORETICAL BASES
Noise exposure was measured according to ISO 1999 as the mean levels of the A-weighted sound levels. Vibration of the engineers' seats was measured according to ISO 2631. The measuring instruments fulfilled the requirements of IEC 179 or 123; and the analyzes those of standard 225.

EXPERIMENTAL TECHNIQUES
Equivalent noise levels $L_{eq}$ were measured by several noise exposure meters or were calculated from a tape-recorded sample by means of statistical analyzes. Noise levels during different driving conditions were analyzed in 1/3 octave bands for the evaluation of their effect on noise exposure. Whole body vibration was recorded in three directions at the same time. The accelerometer was fixed with a magnet to a steel plate (35x35 cm$^2$) on the engineer's seat. The recording time for one sample was about 5 min. During the analyzes the taped sample was played at a speed 10 times higher than the original. The frequency of the sample was also multiplied by 10 and the lower limiting frequency was 1,1 Hz.

RESULTS AND CONCLUSIONS
Fig. 1 shows the results of noise exposure in different locomotives and the noise levels of the whistles and pressure air-flows. The influence of these sound sources on equivalent noise levels were even more than 3 dB. At the maximum speed of the locomotive the whole body vibration measured from the engineers' seats was usually higher than the 8-h risk limit specified by ISO 2631. The most harmful components occurred in 2...4 and 8...12 Hz 1/3-octave bands where it was influenced by the unevenness of the tracks, curves in the tracks and the cars behind. The vibration of the engine and the power transformations had no effect on the engineer.

REFERENCES
(1) ISO 1999
(2) ISO 2631

Fig. 1. Results of noise exposures in passenger trains (○), freight trains (●), and yard locomotives (★). Momentary noise levels from whistles (●) and pressure air-flows (●).
INTRODUCTION AND METHOD

In 57 offices noise measurements were performed and simultaneously 228 office employees were questioned about their judgement of the noise disturbances. These offices varied considerably in site, exposure to outdoor noise, size and number of occupants (1 - 28).

The indoor-noise was measured in fully occupied rooms, with windows closed, the outdoor-noise in empty offices with open windows. The noise was determined in dB(A) weightings, the following noise parameters were evaluated: L1, L50, Leg, NPL, and the standard deviation of noise (STD).

ANALYSIS AND RESULTS

In a further analysis interactions between the objective and the subjective criteria and the character of the noise were studied.

The objective criteria are given by the noise measurements, the subjective criteria by the answers of the office employees. The indoor- and outdoor-noises were originally considered as the two fundamental noise characters, but it proved necessary to introduce a third noise concept. This because on a number of occasions people in the same room, exposed to the same noise, had given different answers to our question about the worst noise. It seems that one can develop a strongly negative feeling towards a specific type of noise and is apt to spot out this noise from a mixed noise environment. According to this hypothesis we introduced a so-called "subjectively worst noise" character.

Altogether 90 interrelationships were tested leading to the following results:

- The objective criteria which correlate best with the assessments are those parameters which take into account the variations of the noise: STD, L1, and NPL. Not suited for this purpose is L50.
- The subjective criteria which correlate best are: questions concerning the loudness and frequency of annoyance, and questions concerning the frequency of interference with communication. Those questions seem to be noise-related.
- The subjective criteria which did not show any correlation to the noise levels were: questions concerning interference with concentration, and impairment of one's own work efficiency. These disturbances do not seem to be noise-related. They appear to be influenced by individual factors, such as professional demands, functions in the office, personal susceptibility, and others.
- The noise characters where the best correlations between noise levels and assessments are found, are: "the subjectively worst noise", and the outdoor-noise. These noise characters seem to be seizable in dB-values, and judgeable in relation to annoyance.
- No relationship between assessments and noise values was found for: the indoor-noise. The fact that conversations were given as the "worst noise" source, emphasizes the legal independent judgement of indoor-noise.
THE EFFECT OF SCATTERING ON SOUND PROPAGATION IN CITY STREETS

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INTRODUCTION

R.H. Lyon (1) in a review paper on the role of multiple reflection and reverberation of the sound propagating from sources in semi-confined areas such as city streets discussed the importance of scattering at the walls. The simple model described below that includes diffuse scattering as well as multiple reflection of sound at the walls of the street leads to estimates that seem to agree quite closely with experimental data.

A MULTIPLE-REFLECTION DIFFUSE-SCATTERING MODEL

Schlatter (2) estimated the sound level in a street by summing the mean-square pressures at the observer due to the source, of power output $W$, and an array of image sources $RW$, $RW$, and so on, where $R$ is the reflection coefficient of the walls. This will be called the multiple-reflection field; it accounts for reflection but not scattering.

It is assumed now that the total noise field is the sum of the multiple-reflection field and a diffuse scattered field. Power balance equations are written for the diffuse sound powers $P^s$ and $P^t$ that propagate away from and towards the source, respectively. The equations account for absorption at the walls and (complete absorption) at the open top of the street, and for power input to the diffuse field from scattering of both the multiple-reflection field and the diffuse field. The resulting pair of simultaneous differential equations for $P^s$ and $P^t$ is solved and the mean-square diffuse sound level estimated.

RESULTS

Typical estimates of the total noise levels at various distances (measured in street widths) along the street are shown in the figure. $ß$ is the scattering coefficient and $H$ the height of the walls (in street widths). The dashed lines represent Schlatter’s results ($ß=0$, that is, multiple-reflection only). The effect of scattering close to the source is quite marked, but at increasing distances from the source, the diffuse field falls off more rapidly than the multiple-reflection field.

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Ground-borne noise due to rapid transit operations in subways

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Rail rapid transit trains operating in subway tunnels generate ground-borne noise that is perceived in nearby buildings as a low frequency rumbling noise. As the rumbling noise is often audible and sometimes intrusive in buildings up to 60 meters from the subway structure, it is important to be able to predict the level of ground-borne noise that will occur adjacent to new rail rapid transit systems.

Ground-borne noise from transit trains generally exhibits a narrow frequency band characteristic indicating that the transmission path from the subway to ground surface acts like a narrow band filter. Typically the ground-borne noise is predominant in the 31.5 or 63 Hz octave bands tapering off steeply at lower and higher frequencies. Due to the filtering effect, only the components of ground-borne noise in the relatively narrow frequency range of 20 to 200 Hz need to be considered in the control of ground-borne noise intrusion.

At the present time there is only a limited amount of data available on soil conditions and ground-borne noise adjacent to transit system subway tunnels, making it difficult, if not impossible, to derive correlations between soil parameters and the characteristics of ground-borne noise propagation. A major source of ground-borne noise data are measurements performed by Wilson, Ihrig & Associates, Inc. at the Toronto Transit Commission (TTC), the Washington, D.C. Metro (WMATA), the Chicago Transit Authority (CTA), and the San Francisco Bay Area Rapid Transit systems. These data allow making some limited comparisons of the transmission characteristics of ground-borne noise.

Figure 1 is a comparison of typical 1/3 octave spectra of vertical acceleration levels at the ground surface in Chicago, Toronto and Washington, D.C. All three measurements were made directly above the subway structure with the top-of-rail 12 to 18 meters below the ground surface. The train speed was approximately 48 km/hr at CTA and 72 km/hr at TTC and WMATA.

The spectra shown for TTC and WMATA have similar maximum levels, however, the frequency of maximum level in Washington, D.C. is about one octave lower than in Toronto. The difference in frequency of maximum response is primarily due to differences in soil parameters and coupling between the soil and the subway structure. Measurements on the subway structure show essentially the same source levels and spectra.

The data from Chicago have a similar low frequency character to the TTC and WMATA data but with the maximum level at 40 Hz. However, above 40 Hz the spectrum does not drop off as rapidly as found in Toronto and Washington, D.C. The more efficient transmission of the high frequency components in Chicago may be partially due to the rigid fastening of the rails to the subway structure via cast-in-place wood ties. In Toronto and Washington, D.C. resilient rail fasteners are used.

In conclusion, ground-borne noise adjacent to transit system subways generally has a very narrow frequency band characteristic. Data on ground-borne noise adjacent to subway structures of transit systems in different cities show that soil parameters have a strong influence on the propagation of ground-borne noise from transit trains to nearby buildings, even when the source level and buildings locations are similar.

FIG. 1 COMPARISON OF TYPICAL GROUND-BORNE NOISE SPECTRA
La recherche du niveau du bruit produit par le trafique urbain s'effectuent, couramment, par des mesurages statistiques, déterminant les paramètres physiques $L_{10}$, $L_{50}$, $L_{90}$ qui représent les niveaux du bruit qui dépassent respectivement 10%, 50% et 90% du temps de mesure, l'écart moyen quadratique $\sigma$, etc.

Parmis les indices de gênes les plus employées sont les indices TNI (Traffic Noise Index); INP (Noise Pollution Index), $L_{10}$ (Niveau maximum admis) $L_{DI}$ (Livello di disturbo del rumore), $C = L_{10} - L_{90}$ (Climat du bruit), ou encore d'autres indices.

Dans le travail, on présente quelques résultats concernant les coefficients de corrélation qui existent entre les paramètres physiques et les indices de gêne.

En se basant sur les mesurages statistiques concernant le bruit, effectuées dans les villes de Bucarest (1976) et de Rome [1], on présente quelques conclusions qu'on peut obtenir de l'analyse des dispersions qui peuvent apparaître, par rapport à la loi théorique, qui lient par exemple le climat du bruit $C$ et l'écart moyen quadratique $\sigma$, ainsi de la comparaison des droites de régression respectives, etc.

Ensuite, on présente quelques conclusions concernant la liaison de corrélation qui existent entre les indices de gênes TNI et INP avec $\sigma$, pour les mêmes mesurages mentionnés plus haut, les ellipses de contour et une discussion concernant la probabilité que les vecteurs aléatoires $(INP, \sigma)$ et $(TNI, \sigma)$ se trouvent à l'intérieur des ellipses de contour, qu'on vient de calculer.


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INTRODUCTION

Since few weeks moved in a house at a distance of 45 m from a main rail route. After more than a year investigating train noise I can now experience it personally. 100 trains in daytime (06.00 to 22.00) and 36 trains in nighttime are travelling on this route.

NOISE SITUATION

Fig. 1 shows $L_1$, $L_{99}$ and $L_{eq}$ as measured in the period 17.15 to 01.30 at the end of each previous 15 min. The microphone was placed in front of a window of a bedroom in the first floor. The small difference between $L_1$ and $L_{eq}$ at 19.30 indicates that a train passed by in a shorter time than 1% of 15 min i.e. 9 s with a peak sound level much higher than $L_1 = 60 \text{dB}(A)$. $L_{eq}$ would be a better indicator for the peak sound level of passing trains as the shortest pass-by time of a train on this route is 2 s. The relation between the plateau level $L$ of the noise profile of a train during its pass-by time $t$ and $L_{eq}$ in a time $T$ is:

$$L_{eq} = L_p + 10 \log \frac{t}{T} \text{ dB}$$

In the daytime 6.25 trains/h pass which leads to the following relation assuming that the average pass-by time of trains is 8 s:

$$L_{eq} = L_p - 18.6 \text{ dB}$$

RESPONSE TO NOISE

In Winter with closed windows is the sound level in the rooms 25 dB(A) lower than the outdoor level. No interference is noticed during the following indoor activities: conversation, listening to broadcasts or music and telephone calls. Since removal train pass-by wake me (40 years) up each night after 4 to 5 hours of sleeping time, but no difficulty is encountered with falling asleep.
INTRODUCTION

Noise analysis consisting in level, spectrum and impulse determination is quite common. But beyond this primary parameters human ear and hearing are able to extract further information, thus recognizing salient features of actual noise source. For instance, engineering or medical experts commonly establish diagnosis by auscultation alone.

EXPERIMENTAL TECHNIQUE

In order to obtain similar data without specially trained staff, an electronic device was worked out, modelling partly the process of auscultation. A vibration transducer or simple microphone placed in the immediate vicinity of the tested equipment picks up the noise generated from it; after convenient filtering and mathematical processing valuable information about running characteristics is separated from redundant components and appears either on analogic (screen) or digital display.

RESULTS AND CONCLUSIONS

The desk model of this device, tested on prototype weaving looms, offered data to detect some ten machine parts requiring improvement. After reshaping them, noise, excessive wear and running irregularities were cut down.

Analysing the sound radiated by working textile machines in a weaving mill the desk model detected inadequate alignment and incipient wear of subassemblies or moving elements.
The efficiency of noise isolating pads is in vehicles lower than in the laboratory by DIN 52 210. By a theory of by-way-transmission of engine-noise in vehicles Betzhold (1) could explain this difference between the transmission loss measured in laboratory and the practical efficiency in the vehicle. He found the equation

\[ \Delta R = 10 \log \frac{p_{21}^2 + p_{22}^2}{p_{22}^2 + p_{22}^2} \]

\( \Delta R \) means the difference between the transmission loss of a car-body-panel and the transmission loss of a combination of car-body-panel with an isolating pad, \( p_N \) the component of noise pressure of by-way-transmission, \( p_{21} \) the noise pressure in vehicle cabins before and \( p_{22} \) the noise pressure in vehicle cabins after sticking them with a barrier \( p_B \).

Equation 1 is the result of discussions of several trials to reduce the noise in vehicle cabins.

It is possible to prove the equation directly during trials on vehicles, if the engine is separated from the car-body (\( p_N = 0 \)). In this state of the trials the barrier pad will be stuck on the partition the reduction of the noise level is equal to the values of \( \Delta R \) measured by DIN 52 210. In fig. 1 curve 1 shows the values of \( \Delta L \) measured in the passenger room of a bus and curve 2 the values of \( \Delta R \) measured in the laboratory by DIN 52 210.

![Graph showing \( \Delta R \) and \( \Delta L \) vs. Frequency](image)

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INTRODUCTION

During the past decade cross correlation and cross spectrum techniques became increasingly attractive to acoustical analysis problems. Since the invention of the FAST FOURIER TRANSFORM (FFT) in 1965 digital methods are effectively used for the determination of auto- and cross spectra and further the coherence function.

THEORETICAL BASIS

Coherence is like correlation an analysis tool for determining dependence through comparison. The coherence function is capable to determine whether a spectral component of one signal is correlated with a component at the same frequency of another signal. It is derived from the auto spectra $G_{11}$, $G_{22}$ and the cross spectrum $G_{12}$ of the two signals 1 and 2:

$$\gamma_{12}^2 = \frac{|G_{12}|^2}{G_{11} G_{22}}$$

Coherence is a function of frequency and takes the values from zero to unity for a linear process. It is independent of the signal levels at the points of observation and the gain of the transducers and the transmission path.

EXPERIMENTAL TECHNIQUES

The approach used for the experimental work is purely software oriented in order to have greatest flexibility concerning spectral resolution and averaging. The original stored digital data are available for different evaluation procedures. The programs are written in FORTRAN and can easily be altered through direct terminal access. The results are being transferred to a high speed plotter for hard copy output.

Acoustical experiments were performed to analyze blower noise and printer noise. Air-borne sound and vibration signals were correlated with a total signal received at a monitoring microphone location.

RESULTS AND CONCLUSIONS

The ordinary coherence function was determined for simulated and practical sound sources. Different frequency resolution and smoothing was applied. For most experiments a resolution bandwidth of 20 Hz and averaging over at least 20 sample records were found to yield satisfactory results.

The coherence function is considered to be a useful analysis tool to determine the fraction of the total power that is due to an individual source e.g. to identify sound sources.

However caution must be observed in the use of ordinary coherence to multiple input source problems if the inputs are somehow correlated. This may occur when using microphones to pick-up the individual source signals; they may receive fractions of signals originating from other sources. To avoid erroneous high coherence the partial coherence function may be used to reveal the existence of a linear relationship between input signals.
INTRODUCTION: When analyzing the noise emitted by a given sound source immersed into a pre-existing acoustic field, we have to resolve the problem of separating the first from the second without the rather costly means of arresting all the noise sources except that to be analyzed, that is, find the frequency spectrum and sound pressure of a given sound source in the presence of others.

THEORETICAL BASE: This determination is feasible by using the "coherence function" which gives the degree of coherence between two time signals, i.e., if the two signals are coherent, the function will be unity while, when incoherent, the function will be zero. Defining now the coherence function we obtain: \( \gamma^2 = \frac{|H|^2}{G_{xx}} \) that is, the numerator indicates the power measured at Y generated by the source in X, while the denominator indicates the total power measured in Y. Evidently, when multiplying the total power \( G_{yy} \) by the coherence function \( \gamma^2 \) we obtain the power at Y generated by the source in X.

EXPERIMENTAL TECHNIQUES: In this specific case the two signals, whose degree of coherence is to be analyzed, are given one by the noise near the source and taken up by a microphone, and the other one directly on the source, as vibratory acceleration. The use of this technique requires however the taking of certain precautions for the purpose of avoiding erroneous results, such as, essentially, phenomena of multiple coherence, time delays between the microphone signal and that of the transducer applied to the source and possible numerical errors caused by the limited dynamics of the analytical instrumentation.

The system we are here considering is a hydraulic unit in a workshop. We have measured the generated noise with a microphone positioned near the hydraulic unit, thus obtaining the PSD \( G_{yy} \) (auto spectrum). When placing now an accelerometer onto the pump of the hydraulic unit we obtain the acceleration PSD, \( G_{xx} \).

At this point we cannot yet express any reasonable hypothesis regarding our system treating the relationship between the frequencies, because we do not yet know their phase relations. Principally we cannot exclude the possibility that the accelerometer detects stationary vibrations originating outside of our system. In our investigation we have used the coherence function instead of the cross-spectrum power function \( G_{yx} \), because the first, resulting from a combination of mean power spectra \( (G_{xx}, G_{yy}, G_{yx}) \), results to be independent from the level of the signals at X and Y, the path of the transmission and the transducer gains.

The coherent noise spectrum in Diagram 1 is very well defined allowing us to individualize the characteristic frequency of our system. This type of diagram is the starting point for a better understanding of the sources of noise by their plotting for different operating conditions allows to determine also quantitatively the more numerous elements allowing us to intervene already at the design stage.
INTRODUCTION

The localization of acoustic sources in a subsonic jet has been studied according to Siddon's (1) causality method. A hot-wire probe has been specially designed in order to avoid the contamination of the measurements. Although some of the results obtained are in agreement with previous investigations along that line, some others support the idea that the notion of acoustic source strength, as usually defined, is questionable.

EXPERIMENTAL

The contribution of fluctuations across the exit plane being negligible (3), the acoustic source strength at frequency f has been measured according to the following formula:

\[ S(f, \theta) = \frac{4}{\pi} \frac{d P}{d y} = \frac{n_f^2}{r a_o^2} \left[ C_p v_o^2 (f, \theta, \frac{r}{a_o} - \frac{1}{2f}) + 2v_0 C_p v_o (f, \theta, \frac{r}{a_o} - \frac{1}{2f}) \right] \]

where \( C_{p,v} \) is the covariance of the components at f of the acoustic pressure and of the velocity fluctuation in the direction of observation \( \theta \). \( v_o \) stands for the component along this same direction of the mean velocity.

All measurements have been made at \( \theta = 30^\circ \) or \( 90^\circ \) within the first ten diameters of a jet whose velocity and diameter are respectively 125 m/s and 20 mm. The frequencies investigated ranged from 1 kHz to 8 kHz.

CONCLUSION

The data obtained allows, in principle, the determination of the regions of the jet which are the most acoustically active. In particular, in the neighbourhood of the peak frequency (Strouhal number \( \approx 35 \)) and at \( \theta = 30^\circ \), the source strength exhibits a maximum between the seventh and eighth diameter and only decreases slowly further downstream, which confirms Seiner's findings (4). However it is worth noting that, depending on the frequency, the source density can well be negative in some regions of the jet: near the exit plane at low frequencies (Strouhal number \( \approx 15 \)) and far downstream at high frequencies (Strouhal number \( \approx 8 \)). In view of such a result, it is not clear, as was foreseen by Ffowcs-Williams (5), what precise physical meaning should be attached to the quantity \( i \), at least for the above mentioned frequencies.

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INTRODUCTION

Recently rather different theories, e.g. those of Mani and of Ribner, have demonstrated considerable success in predicting the directional pattern of jet noise in the far field. This points up a certain insensitivity of the single-microphone response (m.s. pressure) to details of the model of the source region. Two-microphone correlations, on the other hand, would be expected to be substantially less ambiguous. The present paper deals with such two-point correlations and draws inferences on the source structure and coherence. (See (1) for full references to the relevant papers of individuals cited herein.)

THEORETICAL BASIS (1)

The model of jet-noise generation is an approximate form of that of Ribner (1969), which is a development of the Lighthill formalism. The earlier development (mean-square pressure) is extended to provide the two-point correlations of pressure. A series of simplifying assumptions and restrictions, based on physical considerations, is applied to render the sixfold integration tractable.

The predicted shape of polar plots of correlation is found to derive from two main factors: (1) the noncompactness of the source region, which allows differences in travel times to the two microphones - the dominant effect - and (2) the nature of the sources which reflects the composite directivity of the constituent quadrupoles - a weak effect.

RESULTS AND CONCLUSIONS

The present theory is compared with experiment (Maestrello 1976) in Figure 1. The agreement outside the "refractive zone" - not allowed for in the theory - is very satisfactory. The predicted cusps in the pattern are particularly striking.

The experimental confirmation strongly suggests the essential correctness of the source structure postulated in the theory. The sources were modelled as an array of "eddies" (correlation volumes) uncorrelated with one another and individually compact, but in aggregate noncompact.

Characterization of the source region as noncompact and essentially incoherent neither proves nor disproves a role for the large-scale coherent structures in the noise generation process. However, the experimental absence of axisymmetry in the correlation pattern (cf Fig. 1) rules out any significant role for axisymmetric source structures.

REFERENCES

ACOUSTICAL IDENTIFICATION OF THE NOISE SOURCES

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The work coincides into the region of acoustical diagnostics. It gets out of the analysis of the time course of the acoustic pressure annexed to the working cycle of the motor. Under given conditions the consequence of this connection is the possibility to determine the contribution of the individual partial sources in the motor in order to change the noise spectrum.

This work makes full use of the Fourier transformation of the acoustical signal. The instantaneous spectrum, the power spectrum and the power cepstrum are defined and evaluated by using FFT methods.

The evaluation of the instantaneous spectrum of 20° and 2° of crank shaft turning of the motor was made in the frequency range guaranteeing the 10 % accuracy of evaluation. /1/ Further the power spectrum of the one turning and working cycle of the motor and similarly the power cepstrum was evaluated. In order to complete the information on the noise parameters the acute analysis with the eligible band was made (3,14 Hz; 314 Hz), from the time course of the acoustical pressure for a larger set of working cycles. The example of the time course of the acoustical pressure for one working cycle is shown on Fig. 1.

![Fig. 1](image)

Fig. 1

Fig. 2 shows an example of the evaluated power spectrum of one working cycle by the FFT method.

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SOME THEORECTICAL CONSIDERATIONS OF FAR FIELD SOURCE LOCATION TECHNIQUES AND THEIR APPLICATION

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During the past five years a technique called "Polar Correlation" has been developed at I.S.V.R., for evaluating the distribution of sources along the axis of aero-engines. This technique determines the apparent source image, limited by resolution: it defines the source strength per unit length as observed at a specified angle in the acoustic far field. Measurements are made with an array of microphones on a polar arc in the acoustic far field, and it may be shown (1) that the source image is defined by the spatial Fourier transform (with respect to axial separation) of the cross power spectral density between signals from the reference microphone (at the specified observation angle) and others in the array. Essentially this technique is an application of the wavenumber spectrum principle to randomly fluctuating sources, and has been used successfully at both full and model scale.

However in order to use this technique correctly the microphone array must be designed to obtain an unambiguous description of the far field information. Since the technique uses a spatial Fourier transform from discrete measurement points, there exists a spatial Nyquist sampling interval, which is determined by the overall length of the source distribution in wavelengths, and defines the maximum permitted spacing between microphones in the array. Conversely the spatial resolution is defined by the spatial extent of the data; by analogy with the power spectrum, which may be calculated by Fourier transformation of the autocorrelation function, the resolution is determined by the number of lags, or in this case microphones considered. These criteria determine the design of the microphone array, and show that for a given resolution the required number of microphones depends upon the length of the source distribution. The practical implications of these results and the effects of working outside these limits will be illustrated.

Also the relationship between the errors in estimating the cross power spectral level between microphone signals and their effects on the source distribution can be defined. This demonstrates the features of the microphone array design which transmit errors to the source distribution, and gives the required signal averaging time for acceptable error limits. These results are again analogous to power spectra, where high resolution gives a larger statistical error for a given signal averaging time.

While these results are discussed in terms of "Polar Correlation," the same principles apply to any far field source location technique which uses a microphone array. e.g. (2).

REFERENCES


STOCHASTIC PROPERTIES OF ROAD TRAFFIC NOISE

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The sound levels of traffic noise in the vicinity of the roads are randomly fluctuated by the traffic conditions. The stochastic properties of the time sequence of sound levels were studied.

The probabilities that the sound level \( n \) dB is observed at a certain sampling, and \( n-k \) dB or \( n+k \) dB will be observed in the next sampling, are given in terms of the joint events, that is,

\[
W(n,n-k) = \frac{e^{-\mu} \mu^n}{n!} \left( \frac{n}{k} \right)^{n-k} \left( 1-P \right)^{n-k} \tag{1}
\]

\[
W(n,n+k) = \frac{e^{-\mu} \mu^n}{n!} \left( \frac{n}{k} \right)^{n+k} \left( 1-P \right)^{n+k} \tag{2}
\]

Where, \( N \) is the maximum value and \( \mu \) is the average of sound levels during the observation time.

The \( r \) is correlation coefficient between successive two sound levels. By using the eqs. (1) and (2), for example, we obtain the frequencies which equal levels continue, as shown in Fig.1-4.

Fig.1 The frequencies that each sound level is observed separately. The solid line is measured, and the dashed line is calculated.

Fig.2 The frequencies that equal levels are observed two times successively. The solid line is measured, and the dashed line is calculated.

Fig.3 The frequencies that equal levels are observed three times successively. The solid line is measured, and the dashed line is calculated.

Fig.4 The frequencies that equal levels are observed four times successively. The solid line is measured, and the dashed line is calculated.
INTRODUCTION

Recently many researchers subject to assessment of noises are interested in quantity of energy of noise exposure. This quantity is suitable to unified treatment for various kinds of noises and recommended to use as an assessment of noise with respect to community response. In road traffic noise this quantity is supposed to be easy to predict and statistically stable comparing with other quantities. Therefore some fundamental statistics for the equivalent sound level $L_{eq}$ and the exposed sound energy due to road traffic are considered in the present paper.

VARIATION COEFFICIENTS OF EXPOSED SOUND ENERGY AND INTENSITY

For simplicity, let us consider a noise field radiated in an open half space by freely flowing vehicles on a straight lane. The exposed sound energy and its time average are

$$E(p; T) = \int_0^T I(p, t) \, dt,$$

$$\bar{I}(p; T) = \frac{E(p; T)}{T},$$

where $I(p, t)$ is the instantaneous sound intensity at observer distance $p$ from the road and $T$ is observation time length. Some statistical treatments lead to variation coefficients of $E(p; T)$ and $\bar{I}(p; T)$

$$\delta_E(p; T) = \frac{\sigma_E}{\bar{E}} = \sqrt{\frac{1}{N} \frac{2}{\pi c} \left\{ \tan^{-1} \beta - \frac{1}{2} \ln \left(1 + \beta^2\right) \right\}},$$

where $\beta$:velocity of a vehicle, $\nu$:acoustic power of a vehicle, $v$:strength of traffic stream, $dw$:variation coefficient of $w$, $\langle N \rangle = \nu T$, $T = \langle w \rangle / \langle w \rangle$, and $\langle \rangle$:symbol of ensemble average.

DISCUSSION ON STABILITY OF $L_{eq}$

From eq. (3) fluctuations of the exposed sound energy and its time average depend largely on observation time length, traffic volume and radiated acoustic power fluctuation by each vehicle. As a measure of fluctuations of $L_{eq}$ and the exposed sound energy level, let consider 3σ range around their mean values

$$R_\pm(p; T) = 10 \log_\pi \left\{ 1 \pm 3 \sigma (p; T) \right\}.$$  

It is easily verified the range is almost confined in ±3 dB, provided the following condition is satisfied.

$$\langle N \rangle = \nu T \geq 36 \left(1 + \delta_{w^2}\right) \frac{2}{\pi c} \left\{ \tan^{-1} \beta - \frac{1}{2} \ln \left(1 + \beta^2\right) \right\}.$$  

In particular for large $T$ the above condition can be simplified as

$$\langle N \rangle = \nu T \geq 36 \left(1 + \delta_{w^2}\right).$$

Observation time length (or traffic volume) should be appropriately selected in order to confirm stability of $L_{eq}$. In ordinary case the time length is mainly determined by strength of traffic stream and variation coefficient of acoustic power radiated by vehicles.
One of the most common methods to determine the sound power of smaller sources is the reverberation room method. The procedures of the measurement have been standardized in ISO 3741 and 3742 standards. The results obtained when determining the sound power according to these standards are subject to two kinds of errors: one, the error due to the small sampling size and inaccuracies in the processing of the signal, and two, the error due to the radiation of sound power.

The proposed paper will discuss both kinds of errors and the means of their reduction, as a result of the research conducted over several years.

The precision of the results is lower when the source radiates single frequencies than for sources with broader bands of spectra. Also, the precision decreases with the decreasing frequency particularly with inadequate room volume. The use of a rotating diffuser can reduce the errors. Another important factor of the measurement is the placement of the measured noise source.

In order to determine the extent of the errors and the effect of various parameters on the precision of the measurements, a systematic investigation of the effect of the source position, modal damping and rotating diffuser has been conducted. Also, in order to follow, at least, quantitatively, the effect of a rotating diffuser on the sound field in the room, a theoretical model of the time-varying boundary effects (modeling the rotating diffuser) has been worked out and the calculations of the variables of the sound field with the time varying boundaries have been performed. Because the major inaccuracies exist at low frequencies, most of the calculations were performed at 125 Hz. The results obtained from the theoretical model are in good agreement with the results obtained from the measurements with a rotating diffuser.

One of the important quantities affecting the precision of the results is the variance of the sound pressure squared. The effect of modal damping on $p^2$ variance with time varying boundary and with fixed boundary has been determined. Another important factor discussed over recent years is the actual power radiated by the source into the reverberation room as compared with the power radiated into the free space. The moving boundary has very little effect on the sound power radiation. Higher modal damping increases the total radiated power. Finally, the effect of the mentioned parameters on the qualification of the room has been analyzed and practical data are presented.
HOW TO DEFINE THE DIMENSION IN SPACE OF "THE" NEAR FIELD OF A SOUND SOURCE?

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INTRODUCTION

The dimension of the near field of a source is of interest among others if the sound radiation of a technical sound source shall be determined or if sound pressure values at the operators place or in the neighbourhood are to be predicted from data received from noise measurements near the source. In the acoustics different definition of the "Near Field" of a sound source are used. These definitions were collected and compared in respect to their dimension in space for the same sound source.

DEFINITIONS

At first we distinguish between two groups of definitions for the near field

(1) Definitions which are based on quantities of the sound field, as there are the sound pressure \( p \), the sound velocity \( v \), the phase angle \( \gamma \) between both. Thereby the behaviour of these quantities in function of the distance from the source is used as the relevant criteria. Reference are the asymptotic laws for very large distances.

(2) Energy quantities, meaning the intensity, both in a single point or space averaged over an enveloping surface. These quantities are regarded in function of the distances from the source too.

The definitions using field quantities are more familiar but in general of less interest in the acoustical planning and for the measurement technique on technical sound sources.

RESULTS

In general the near field definitions basing on energy quantities lead to significantly smaller near field spaces than those using sound field quantities.

For example the spherical sound source of \( n \)-th order unlimited in its size has sound field quantities (sound velocity, phase angle \( \gamma \)) with a typical near field behaviour in the vicinity of the source. This means this source has a "near field" of a certain dimension based on field quantities. On the other hand the energy quantities \( 1/2 \rho c \cdot p^2 \), \( (p \cdot v) \) for the same source has a dependence of distance which is exactly the same beginning close to the outer-surface of the source and in each greater distance. According to the usual understanding this source for the energy quantity has a near field of zero dimension, meaning no near field.

In addition to this extreme example the magnitude of the near field dimensions was calculated for several spherical sound sources of higher orders and for multipole sources using the different near field definitions.

The smaller dimension of the near field for energy quantities is an important advantage because this understanding of the near field has greater relevance for a lot of problems in acoustical planning and for sound power output measurements of technical sound sources.

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INTRODUCTION

The impact of spheres has been considered as a model for ball bearing noise and punch press noise. Measurements and theory have shown the form of the sound pulse to be similar to that of a damped sine wave. (1). The radiated sound is highly directional, and the peak sound pressure is proportional to the velocity to the power 1.2. (2). This paper presents the results of measurements with a sound level meter of the impact of dissimilar spheres.

EXPERIMENT

An experiment was carried out in which the sound pressure in the near field was measured by a sound level meter. See fig.(1). Various collision velocities were used for impacting balls of both equal and unequal sizes.

RESULTS

It was empirically established that the mean square sound pressure $p^2$ as measured is given by

$$p^2 = 3(m_1^{0.5} u^{0.5} + m_2^{0.5} u^{0.5}) \, (N/m^2)$$

where $m_1$ and $m_2$ are the masses of the colliding balls and $u$ is the velocity just before impact. See fig.(2).

The constant of proportionality will depend upon the distance of the microphone from the point of impact.

REFERENCES

ESTIMATES OF MONOPOLE POWER RADIATED IN A REVERBERATION CHAMBER

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A source, idealized as a constant-velocity monopole, radiates a steady, pure tone signal in a Reverberation Chamber (RC). We assume that the signal excites many room modes. We wish to estimate, from sound pressure measurements, the free-field value \( W_0 \) of the sound power output of the source. If the source is placed at one point only, an accurate value of its power output in that position can be found by averaging the mean square pressure \( P^2 \) at many points. However, to find \( W_0 \), the source position must also be varied. We have

\[
W_0 = \frac{\langle P^2 \rangle A}{4 \pi c}
\]

where \( \langle \rangle \) means the space-averaged value when the positions of both source and receiver are varied. \( A \) is the total absorption of the RC and \( c \) is the wave impedance of the air. \( P^2 \) is a random variable, and is a function of two other random variables, \( r \) and \( s \), representing the source and receiver positions. These are continuous variables, but we sample them discretely. We now consider the quantity

\[
W(r, s) = P^2(r, s) \frac{A}{4 \pi c}
\]

We can write the dependence of \( W \) on \( r \) and \( s \) as a product of two (unknown) functions, \( A(r) \) and \( B(s) \), i.e. as

\[
W(r, s) = A(r)B(s)
\]

Now by reciprocity, the functions \( A(r) \) and \( B(s) \) must be the same, so

\[
W(r, s) = A(r)A(s)
\]

From earlier work the probability density function (pdf) of \( A(s) \), i.e. \( P_A(x) \), is known to equal \( e^{-x} \). Therefore the pdf of \( W \) is

\[
P(x) = \int_{\infty}^{\infty} \frac{1}{u} P_A(u) P_A(x/u) du = \int_{0}^{\infty} \frac{du}{u} e^{u+x/u}
\]

This integral does not appear to be evaluable in closed form, but it can be shown that \( P(x) \) equals the pdf obtained by sampling the algebraic function

\[
W(r, s) = (\ln r)(\ln s)
\]

over the square \( r, s = 0 \) to 1. The mean of the distribution is unity, and the variance is 3.

Fig. 1 shows the pdf and cumulative functions of \( W(r, s) \), obtained from 10,000 values randomly sampled on the computer, from the function \((\ln r)(\ln s)\). \( P(x) \) is highly asymmetrical about the mean, the most probable value being zero.

It is clear that many values must be averaged to give a good estimate of the mean. For the 10,000 values the mean was 2.9% off the true value. Calculations are in progress to give the probability of the sample mean being within \( \pm 1 \) dB of the true mean, versus the number of values.

Fig. 1. Probability density \( P(x) \) and cumulative function \( F(x) \) for sampling the power output of the pure tone source in the RC. \( W_0 = 1 \) = mean value.
INTRODUCTION

A significant part of the noise of a modern rotary press is associated with the vibration of the web of paper that moves under variable tension between high speed rollers. We have been attempting to model this process and to determine what parameters affect the noise and the sensitivity of the sound field to changes in these parameters. The practical problem is extremely complex, and our model is necessarily highly idealised.

ANALYSIS

We consider a semi-infinite elastic sheet initially at rest with a prescribed displacement. The sound produced by suddenly tugging one end is investigated. We first determine the displacement of the sheet by solving the elastic equations of motion within the membrane, in the limit where the fluid loading has a negligible effect on web response. It is found that a tension wave propagates supersonically through the sheet and triggers off a transverse vibration. The sheet is stationary ahead of the tension front and behind it the sheet is uniformly tensioned and supports the waves associated with a tensioned elastic membrane. This transverse wave is composed of elements propagating in both directions with a constant subsonic speed and may be expressed in terms of the prescribed initial displacement.

Once the normal displacement of the sheet at all times has been determined we can write down an expression for the sound produced as an integral of the normal velocity over the sheet. It is found that the motion of the sheet is silent except at the tugged end and at the tension front. Transverse waves are reflected noisily from the tugged end and the sudden switch-on of normal velocity at the tension front is like a moving sound source. Sound is only heard within a two-dimensional Mach wedge, with the Mach number based on the speed of the tension waves. The far field density perturbation has all the characteristic features of a two-dimensional sound field; it decays as the inverse square-root of the distance from the source, and involves half-derivatives and Doppler factors. When the tension is applied instantaneously there is a sonic boom on the Mach wedge similar to that of a supersonic aircraft, but allowing for the tension to be applied over a small but non-zero time restricts the strength of the density perturbation on the Mach wedge in the far field.

The effect of fluid loading is described and the similarity of the problem with the practical situation is commented on with recommendation for noise control.
For estimating aerodynamic sound generated by the devices in duct system, it is desirable to standardize the method expressing the flow conditions and the sound data. The relation between aerodynamically generated sound and flow condition is derived from the Lighthill's theoretical study [1] as equation (1) and verified experimentally [2].

\[
L_W(f) = L_G(S) + 10 \log Q^* P^{* \alpha} \frac{d}{U}^2
\]

where \(L_W(f)\) in dB re 10^{-12} watt is the generated power level of the octave band with center frequency \(f\) Hz. \(S = f d / U\) refers to Strouhal number. \(Q^*, P^*, d^*\) and \(U^*\) denote the ratios of the air flow rate \(Q\), the pressure drop \(P\) of the source, length \(d\) and velocity \(U\) to the reference values of 0.1 m³/s, 60 Pascals, 0.01 m and 10 m/s, respectively. The exponent \(\alpha\) of the pressure drop and the function \(L_G(S)\) by means of Strouhal number depend on the shape of the source in the flow.

Concerning the sound generated by rows of plates as well as cylinders, the primary component of the sound is represented by the same function of Strouhal number in which the velocity is expressed as \(U^* = \sqrt{P^*}\), and regard to the length, when the ratio of the opening size to the bluff body size \(B_o / B_P\) is between 1 and 5, \(B_o\) is used but when the ratio is not in this range, \(B_H\) may be used, see figure 1. Particularly, in case \(B_H / B_P\) is between 1 and 2.5, the strong periodic component of the generated sound takes place at Strouhal number of approximately 0.2. In regard to miscellaneous devices including diffusers, grilles, dampers and sound traps, for some of them \(L_W(f)\) is expressed by the equation (1) and for others by the same equation including \(L_G(f / U)\) instead of \(L_G(S)\).

**References**

Acoustic analogies can give a plausible description of the sources of aerodynamic sound but the descriptions are essentially non-unique. When vorticity concentrations are admitted, the basic flow with which the sound interacts is sometimes unstable. Acoustic analogies often suppress these instabilities with the inevitable violation of 'causality'. This paper examines the importance of these issues with regard to the theoretical modelling of the jet noise generation process.

The paper mainly concerns a particular generalisation of Lighthill's analogy in which the Navier Stokes equations, written symbolically as $NS = 0$ are exactly rearranged within one region of space into an inhomogeneous wave equation $\Box^2 \rho = \Box^2 \rho - NS = \frac{\partial^2 T_{ij}}{\partial x_i \partial x_j}$, $T_{ij}$ being Lighthill's stress tensor. In another adjacent region the equations are rearranged as an inhomogeneous convective wave equation $\Box_m^2 \rho = \Box_m^2 \rho - NS = \frac{\partial^2 T'_{ij}}{\partial x_i \partial x_j}$ where $T'_{ij}$ is Lighthill's stress tensor in a different reference frame.

The two regions are matched with boundary conditions corresponding to an exact non-causal vortex sheet modelling of the flow. At first sight this procedure seems to violate obvious physical constraints, but it is shown that there is in fact no contradiction and that the non-causal approach is more easily made relevant to physical processes than a causal view with its unconstrained exponentially growing free modes. The paper shows that the real field is given exactly as the convolution of $T_{ij}$ with a convected quadrupole Green's function that is essentially non-causal.
EXPERIMENTS ON THE GENERATION OF LOCAL ACOUSTIC RESONANCES IN FLOW DUCTS

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INTRODUCTION

Local acoustic resonances of the Parker β-type (1,2) have been studied in the region of single rectangular cross-section elements spanning the width of a flow duct. In addition to measurements of the modal sound pressure levels, hot-wire anemometer measurements were made of the velocity fluctuations in the shear layers resulting from flow separation at the elements' leading edges.

EXPERIMENTAL APPARATUS

The experiments were performed in a flow duct of cross-section 1x.1m. The range of element heights, L_y (dimension normal to the direction of flow), was 0.04-0.075m and element lengths, L_x (dimension in the direction of flow), was 0.03-0.06m. The maximum duct velocity was about 30 m/sec.

RESULTS

In general, sound pressure maxima were measured in two velocity ranges for each geometry, with conditions for maximum excitation characterized by Strouhal numbers

\[
N_{st} = \frac{f L_y}{U_{A.D.}} = 1.39 \text{ and } 2.41
\]

(U_{A.D.} is the average velocity on the side of the elements). Acoustic pressures of the order of the dynamic head of the duct flow were measured for the lower Strouhal number. The modal frequencies were found to be close to the same value for both maxima and to be related to the velocity fluctuations measured in the shear layers.

REFERENCES:


Fig. 1 Modal sound pressure level VS. \( U_{A.D.} \), \( L_x = L_y = 0.05 \)
REGENERATED NOISE IN SPLITTER SILENCERS

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INTRODUCTION

Of the problems occurring with site installations of rectangular silencers in ventilation systems, over half are due to regeneration of additional noise by the flow of air through the silencer system. Broadly speaking, these problems fall into two categories; those of low frequency (< 1 kHz) and those of high frequency (> 1 kHz) noise generation. The low frequency noise is due to the turbulent wake noise at the silencer exit whilst the high frequency noise is generated by the flow over the splitter facing material.

THEORY

The "bluff body noise" generated at low frequencies can be treated using a point dipole source model as in the method used by Gordon [1]. This approach relies on the linear relationship between the amplitude of the fluctuating drag force acting on the body to the steady state drag force as established by Heller and Widnall [2]. Application of the model to silencer splitters results in the following relation of radiated sound power $W$ with silencer passage velocity $U_a$, silencer height $h$, pressure loss 'K' factor and number of splitters $n$:

$$W = n h K^2 U_a^6,$$

The high frequency surface noise can be treated as a continuous generation and absorption of noise along the length of the silencer. Following an analysis by Ingard [3], it can be shown that the sound power radiated at the end of the silencer is given by

$$W = \frac{n h U_a^6 (1 - e^{-\beta L})}{\beta},$$

where $\beta$ is the energy absorption coefficient per unit length and $L$ is the length of the silencer.

EXPERIMENTAL INVESTIGATION

A series of experiments on various silencer configurations was carried out in order to evaluate the constants of proportionality in the above expressions. Air from well silenced axial fans was passed through test silencers and the sound power produced was measured in a reverberation chamber into which the test duct passed. The results were collapsed to give generalised spectrum shapes for both the high and low frequency noise, thus enabling the prediction of noise radiated by various silencers under different flow conditions.

REFERENCES

IDENTIFICATION OF THE SOURCE OF ACOUSTIC ENERGY IN A FLOW EXCITED HELMHOLTZ RESONATOR

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INTRODUCTION

The production of aerodynamically generated sound has undergone exhaustive theoretical study over the previous two decades. Despite this, the understanding of the physics of the interaction of flow and sound has been held back by considerable experimental difficulties. The emergence of laser anemometry has now enabled the simultaneous point measurement of two orthogonal velocity components in low Mach number airflows. This technique has been used to examine the interaction of sound and flow in the neck of an aerodynamically excited Helmholtz resonator. The feedback mechanism governing the fluid behaviour ensures correlated vortex shedding in the resonator neck giving an effectively two dimensional flow. In addition, fluid fluctuations occur predominantly at a single frequency. The measurement of amplitude and phase of the two fluctuating velocity components at this frequency enables the complete description of the unsteady flow field, together with the ability to evaluate local properties of the flow such as vorticity, Reynolds stress and Coriolis acceleration.

EXPERIMENTATION

The Helmholtz resonator investigated consisted of a rectangular hard walled box with a long sharp edged slot milled in the upper surface. An air jet was directed across the slot at a sufficient speed (= 10 m/s) to maintain a Helmholtz resonance at 605 Hz. The laser Doppler velocimeter used for the velocity measurements is described in reference 1. Seeding of the airflow with oil mist particles ensured a continuous signal from the LDV and time resolved measurements of the fluctuating streamwise and vertical components could be taken. Small glass windows in the sides of the box permitted optical access for the laser beams. Pressure measurements were also taken in the neck of the resonator using a 1 mm diameter probe tube. The single frequency nature of the pressure and velocity fluctuations enabled their relative phases, with reference to the cavity pressure, to be measured directly on an analogue correlator.

RESULTS

Results have shown that there is a large gradient of the fluctuating streamwise velocity component in the vertical direction, giving a large value of mean square vorticity in a region near the downstream edge of the slot (Fig. 1). Terms associated with acoustic energy production have also been shown to be large and concentrated in this region and the pressure measurements have revealed substantial gradients of pressure in this same area (Fig. 2).

REFERENCES

EXCITATION OF AN ACOUSTIC RESONATOR BY TURBULENT FLOW

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INTRODUCTION

It has long been recognized that an acoustic resonator exposed to grazing turbulent flow may be strongly self-excited by the shear layer that forms over its orifice. This phenomenon limits the use of resonant silencers in flow systems.

Resonators excited by turbulent flow may profitably be treated as jet-edge-resonator systems by drawing on well established edgetone theories, which have been shown to apply to turbulent as well as laminar flows [1]. Interaction of shear layer instabilities with the downstream orifice edge creates an oscillating pressure gradient in the orifice which disturbs the shear flow. If the feedback thus produced is of the appropriate phase the disturbance will be amplified by the flow, and further if a resonator natural frequency lies in the range of shear layer instability, the feedback will be considerably enhanced, effectively broadening the regime of instability.

EXPERIMENTAL INVESTIGATION

An annular resonator was formed between two concentric sections of pipe of 75 mm and 150 mm diameters, the inner carrying turbulent air flow. An orifice of variable streamwise length, L (6 mm to 25 mm), was formed by a complete circumferential cut of the inner pipe (see Fig. 1). The system was tested with the orifice centrally located, at centre-line pipe velocities (U) from 10 ms⁻¹ to 160 ms⁻¹. The resonator SPL was monitored using a probe microphone. Fig. 1 Flow duct-resonator cross-section.

RESULTS

It was found that the system operated in two stages (cf. edgetones) and that peak self-excitation data were well reduced by

\[ f_n L = \kappa U (m - 0.5) \quad m = 1,2 \]  

(after Rossiter [2]), where \( f_n \) are the frequencies of the annulus transverse acoustic modes, and \( \kappa \) may be interpreted as the ratio of the shear layer disturbance convection velocity to the centre-line velocity, U. Values for \( \kappa \) inferred from the results are typical for free shear layers (0.4 to 0.6). The success of equation (1) indicates: (i) that peak self-excitation in stage 1 (m = 1) occurred when the fluid disturbance wavelength was 2L and (ii) that at resonance, feedback from the downstream edge to the upstream separation point was effectively instantaneous for all flows tested (i.e. compressibility may be neglected). It is suggested that these results apply to flow-excited resonators which satisfy the criterion SL < V where S and V are orifice area and resonator volume respectively. Equation (1) may be used to interpret the results of other investigators, e.g. [3].

REFERENCES

(1) A. Powell, JASA (1961) 33, 395.  
(3) R.L. Panton and J.M. Miller, JASA (1975) 58, 800.
LE PROBLÈME DE LA RÉDUCTION ACTIVE DU BRUIT ET DES SOURCES ANTI-BRUIT DANS LES CONDUITS

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L'absorption active est réalisée au moyen de sources secondaires utilisées suivant le principe de Huygens.

Le principe utilisé est l'insertion sur le trajet de l'onde dans le conduit d'une source sonore dipolaire et d'une source monopolaire fonctionnant en même temps. Le dipôle est disposé dans une enceinte symétrique (géométriquement et acoustiquement) par rapport à un plan médian perpendiculaire au conduit (1). Le monopôle est disposé face au dipôle sur l'autre paroi du conduit. Cet ensemble constitue une source tripolaire dont le diagramme de directivité doit être tel que la source ne rayonne aucune énergie vers l'amont du tube (zone de bruit) et prodigue un champ sonore vers l'aval de niveau égal au bruit à réduire et de phase opposée. On obtient ainsi 50 dB d'affaiblissement aux basses fréquences. La zone anti-bruit est située au niveau des sources antagonistes.

RESULTATS

La courbe d'affaiblissement résultant de l'absorption active est décrite à l'aide de la théorie des découpages (2). Une approche théorique de la réalité expérimentale est obtenue en utilisant un découpage flou et une fonction de découpage continue aux dérivées premières et secondes continues, par exemple :

\[ S(M) = \frac{1}{2} - x - \frac{1}{\pi} \sin 2\pi x \]

fondction qui est représentée par une courbe symétrique par rapport au centre de l'absorbeur actif. L'expérience montre que la courbe n'est pas symétrique ; elle atteint son minimum à une distance constante de 10 cm en moyenne par rapport à l'axe des sources anti-bruit, pour des ouvertures différentes du monopôle, du dipôle, quelle que soit la fréquence. Les dimensions des enceintes, la surface des bouches d'émission et la géométrie influent sur l'absorption du son. Les travaux actuels que nous poursuivons ont pour objet l'optimisation des caractéristiques de ces sources liées à leur efficacité.

REFERENCES

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ON THE NOISE OF CENTRIFUGAL FANS

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INTRODUCTION

Outlet noise spectrum measurements of centrifugal fans show maxima corresponding to the resonances of the outlet duct. One observes no such resonances in the inlet noise, but there are present resonances of the fan volume with the inlet duct. Pressure fluctuations at the cut-off show no duct resonances at all.

THE PROPOSED MODEL OF NOISE GENERATION

One assumes that a noise source of a given pressure and low mass impedance is located near the cut-off.

CALCULATION

The equivalent circuit of the above model is shown on fig. 1. The quotient of outlet to inlet volume velocities is

$$\frac{v_o}{v_i} = j\omega m_v \left[ \frac{3}{2} \cos kl \right]^{-1} = \omega m_v S_d \left[ \frac{3}{2} \cos kl \right]^{-1}$$

If $\omega > (m_v / \rho)^{1/2}$, the fan must be considered as a cavity resonator. For this more complicated case, the formula is:

$$\frac{v_o}{v_i} = \left( \frac{3}{2} \cos kl \right) \left[ \frac{3}{2} \cos kl \right]^{-1}$$

where $J, N$ - Bessel functions, $S_d$ - cross-section of the outlet duct, $S_o$ - orifice, $a$ - width, $c$ - radius of the fan casing, $\rho$ - inlet radius, $\rho_o$ - outlet radius, $\rho_c$ - acoustic mass at the inlet, $k = \omega c / \rho$. The proposed model and calculations can explain the phenomena mentioned in the introduction.

DISCUSSION

Both expressions for $v_o / v_i$ show, that the outlet noise compared with the inlet noise has maxima when half-wave resonances of the outlet duct ($kl = \pi n$) occur. When the sum in the brackets vanishes (inlet cavity resonances) $v_o$ becomes greater than $v_i$. Detailed examination of the spectral density of $v_o$ show that the interaction between the resonances of the outlet and inlet vanishes with $m_v \rightarrow 0$.
The fan sound pressure is described by the following expression (1):

\[ K = \rho_0 U^2 \cdot F(St, \varphi, x_i/D, Re) \cdot G(He, \varphi, x_i/D, Re) \]  

\[ K = K_d = \Delta \bar{p} \] for the discrete frequency sound (2)

\[ K = K_r = \Delta \bar{p} / (\Delta f / \lambda_0)^{1/2} \] for the random sound (3)

\[ \text{Re} = \frac{U \cdot D}{\nu}, \quad \text{St} = \frac{f \cdot D}{U}, \quad \text{He} = \frac{D}{\lambda}, \quad \varphi = \frac{4 \nu}{\pi \rho_0 U} \] (4)

\( \Delta \bar{p} \) is the sound pressure within the filter bandwidth \( \Delta f \); \( x_i \) are the coordinates of the measuring position. Diameter, width, blade number and tip speed of the impeller are denoted by \( D, b, Z \) and \( U \), respectively. \( V \) is the volume flow. The working fluid is characterized by the kinematic viscosity \( \nu \) and the density \( \rho_0 \). \( \lambda_0 \) and \( \lambda \) are sound speed and wavelength, respectively.

The important feature of equation (1) is that the spectrum of the sound is governed by two different functions: The term \( F \) describes the spectral distribution of the sound generated inside the fan; it depends mainly on the Strouhal number \( St \) or, in other words, it is determined by the motion of the impeller. The term \( G \) is mainly a function of the Helmholtz number \( He \) which is a purely geometric parameter. Therefore \( G \) can be considered as an acoustic frequency response function, characterizing the acoustic resonance and radiation properties of the fan.

Experiments were made with two geometrically similar fans having impellers of 140 and 280 mm diameter. Anechoically terminated ducts of 70 and 140 mm diameter, respectively, were mounted on the fan outlets. Figure 1 shows the sound pressure levels of various spectral components in the ducts as functions of the Helmholtz number, when the tip speed is varied. The value \( St = 1 \) corresponds to the blade passage frequency. For all spectral components, there is only a weak influence of the fan size which means that the influence of the Reynolds number is negligible for practical purposes. More details are given in references (1-4).

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(3,4) W. NEISE, J. Sound Vibr. 43 (1973); 61-75; Proceedings DAGA 76, 307-310.

![Fig. 1: Variation of various a) harmonic and b) random components as the tip speed is changed.](image)
INTRODUCTION

A theory has been described in ref (1) which gives the far field radiation from most multipole source distributions and disturbances in motion.

GENERAL SOLUTION

The sound pressure is given in terms of a series of simple separate functions as represented by equation (1) below:

\[ (SP)_n = K_n X_s X_m B (\frac{ap}{q}) [H] \]  

\( K_n \) is the familiar multipole order directivity function, \([H]\) defines the source strength and \( X_s, X_m \) contain information regarding the source disturbances (time history) and source distributions.

These latter functions also contain the essential information regarding the flight path and velocity of the source. Therefore, \( X_s \) and \( X_m \) rather than \( K_n \) contain the major radiation properties of a source in motion.

EXAMPLE

An application of equation (1) is shown in figure (1). It shows three similar source situations, but with quite different polar directivities. (a) is a moving time modulated source (flying loudspeaker). (b) is a moving source-stationary spatial disturbance (flying aerofoil) and (c) is a stationary source-moving disturbance (aerofoil in an open wind-tunnel). The radiation properties are for a monochromatic disturbance, \( ka = 0 \) (point source), \( ka = 0.1 \) (stationary compact source) and \( ka = 10 \) (stationary non compact source), where \( ka \) is the usual wave number source dimension product.

The sources are defined for a rectangular force distribution and constant rectilinear motion \((M = 1)\). The 0 dB circle represents a stationary point source, and + and - dB indicate amplification and attenuation respectively.

RESULTS AND CONCLUSIONS

Of particular interest is (i) the absence of the acoustic beaming effect for a non compact source. \((ka > 1)\) in situation (a) and (b). (ii) there is a motional effect at right angles to the direction of motion in (b) and (c). (iii) there is an acoustic beam only in the case of a non-compact source in (c).

It is important therefore to take account of finite source distributions and define precisely the source situation.

REFERENCE

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APPLICATION DE METHODES DE CALCUL DU BRUIT DE SOUFFLANTE

INTRODUCTION - Les méthodes de calcul du bruit de soufflante sont de première utilité pour prévoir l'émission sonore d'une nouvelle machine et pour connaître l'importance des imperfections de montages au banc, telles que la présence de bras supports supplémentaires. Le but de l'étude est de comparer l'application de méthodes trouvées dans la littérature à des résultats expérimentaux.

PREVISION A PARTIR DE MESURES A LA SOURCE SONORE - A partir des équations de Lowson(1), le niveau des modes acoustiques propagatifs peut être déduit du spectre des fluctuations de force mesurées sur aubes mobiles(2), fournissant une indication directe sur la source sonore. Les mesures sur un ventilateur caréné ont bien confirmé le calcul dans diverses configurations, puisque les écarts restent dans tous les cas inférieurs à 3 dB.

PREVISION A PARTIR DE MESURES DANS L'ECOULEMENT - L'accès à la connaissance de la source sonore nécessite l'implantation de capteurs sur rotor(3), ce qui conduit à des montages assez compliqués, réalisables seulement dans des cas particuliers et non de manière systématique. C'est pourquoi il est intéressant de disposer de calculs basés sur les caractéristiques aérodynamiques de l'écoulement. Ceux-ci comprennent un intermédiaire supplémentaire par rapport à la méthode précédente, celui de la réponse de l'aubage mobile, qui se traduit par une fonction de Sears.

Les équations de Goldstein(4) et Mani(5) fournissent les niveaux sonores pour chaque mode propagatif. En ce qui concerne le bruit dipolaire, les deux approches peuvent être ramenées à des formes semblables mais il existe une divergence importante pour le bruit quadrupolaire. Les écarts par rapport aux mesures sur une soufflante restent assez grands.

CONCLUSION - Les études en cours consistent à comprendre d'où viennent les différences entre ces derniers calculs et les expériences et à tenter d'améliorer les prévisions en partant d'hypothèses plus réalistes.

Les essais effectués concernent des profils de type et de corde différents (NACA 0012 c = 8 cm ; NACA 6512 A10 c = 8 cm et c = 5 cm) et une pré turbulence dont l'intensité des fluctuations transversales peut atteindre 13 % et dont l'échelle intégrale relative aux fluctuations transversales et à une séparation longitudinale varie de 0.8 cm à 3.5 cm. La vitesse moyenne est ajustable de 20 à 40 m/s. La pré turbulence est créée par diverses grilles (maille : 6 cm, 3 cm et 1,5 cm).

L'effet de la pré turbulence est d'accroître le niveau acoustique dans une gamme privilégiée de fréquences. Par exemple, cette gamme s'étend de 100 Hz à 1500 Hz pour le profil NACA 0012 de 8 cm de corde (U = 20 m/s).

L'augmentation du bruit émis peut en principe être calculée si l'on connaît le champ de pression instantanée en tout point du profil. Cela a été vérifié dans le cas de perturbations déterministes (1), (2). Dans le cas d'une pré turbulence, on a en général recours à des expressions simplifiées. On introduit alors une fonction de transfert aérodynamique pour caractériser le champ de pression instationnaire sur le profil et une pondération pour tenir compte de la compacité acoustique du profil. Les expressions les plus simples sont obtenues lorsque l'échelle des perturbations incidentes est supérieure ou égale à la corde (3) (4).

Dans les différentes expériences effectuées, nous avons d'une part utilisé les diverses formulations proposées pour calculer le spectre du bruit émis, et d'autre part mesuré le champ de pression instationnaire sur le profil. Cette double approche nous a permis de préciser les limites de validité des simplifications envisagées.

REFERENCES

Ce travail a bénéficié de l'aide du Service Technique de l'Aéronautique.
BASE PRESSURE OSCILLATIONS IN A RECTANGULAR DUCT WITH A SUDDEN ENLARGEMENT

Anderson, J. S. (1)  
Grabitz, G.  
Jungowski, W. M. (11)  
Meier, G. E. A.

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(1) The City University, London, (11) Warsaw  
Technical University

INTRODUCTION

The flow in a duct with a sudden enlargement has the following characteristics: (i) sonic flow at the smallest cross-section, (ii) expansion fans from the smallest cross-section with supersonic velocities and dead air in the base region at the corners of the duct, (iii) a strong normal shock at the end of the expansion region, followed by a mainly subsonic flow.

EXPERIMENTAL RESULTS

Experiments show (1) that the flow field described above can be both steady (Fig. 1a) or unsteady (Fig. 1b), and in the latter case self-excited oscillations occur. These oscillations exist if the base pressure $p_W$ is greater than a certain value, and if the cross-sectional area ratio $(h/H)$ is in the correct range for a given length of duct $L$. The frequency of the oscillations is mainly determined by the duct length. Typical amplitudes are 10% of the rest pressure. The oscillations produce considerable external noise.

THEORETICAL OUTLINE

The theoretical model of the flow (2) incorporates a supersonic expansion area (bounded by straight jet boundaries and a straight normal shock) which is treated with an integral method applied to the gas dynamic equations. The feedback between the base region and the flow area behind the shock is described by various model equations. The results for both experiment and theory are of the same order of magnitude, and an example of the agreement is given in Figure 2.

REFERENCES

(2) G. Grabitz, Max-Planck-Institut für Strömungsforschung, Göttingen, Ber. 14/76
PROCESSING PRECISION INFLUENCE UPON THE GEAR NOISE AND VIBRATIONS

Merticaru, V. Polytechnic Institute, Jassy, Romania

INTRODUCTION

The gear vibrations in direction of line-of-action were considered. An original mathematical model permitted to establish the calculating relations for the gear noise. These relations permitted to establish the processing precision influence upon the tooth separations as well as gear noise.

THEORETICAL BASES

The tooth separations occur when inequality (1) is accomplished,

\[
\frac{F}{K} \cos\omega t + B\sin\omega t + e^{-\frac{h_1^2}{2M}}(C\cos\frac{\sqrt{4MK-h_1^2}}{2M}t + D\sin\frac{\sqrt{4MK-h_1^2}}{2M}t) < E\sin(\omega t + \alpha),
\]

where \( F \) is normal force on the gear teeth, \( K \)-gear rigidity, \( h \)-damping constant, \( M \)-reduced mass of gear, \( E \)-reduced error of teeth profile, \( \omega \)-reduced pulsation of the forced vibration of gear, \( \alpha \)-reduced initial phase.

The reduced error of teeth profile enters in the calculating relations of the constants \( A, B, C, D \). For instance the constant \( C \) is given in relation (2),

\[
C = \omega E(h \cos \alpha - (MK - M^2 \omega^2 - h^2) \sin \alpha) / ((K - M^2 \omega^2 + h^2 \omega^2)^2 - F/K).
\]

EXPERIMENTAL TECHNIQUES

The testing machine was installed in an anechoic room.

Table 1 shows the increasing of the acoustic pressure level \( AL (\text{dB}) \) due to increasing of the profile error from \( E = 2.3 \times 10^{-5} \) m to \( E = 3.9 \times 10^{-5} \) m, where

<table>
<thead>
<tr>
<th>E (m)</th>
<th>1 - 2</th>
<th>2 - 3</th>
<th>1 - 2</th>
<th>2 - 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.3 \times 10^{-5}</td>
<td>5.2</td>
<td>3.6</td>
<td>1.8</td>
<td>2.4</td>
</tr>
<tr>
<td>3.9 \times 10^{-5}</td>
<td>5.3</td>
<td>2.4</td>
<td>5.3</td>
<td>2.4</td>
</tr>
</tbody>
</table>

Table 1

The measurements were made with Bruel - Kjaer apparatus.

The main conclusive results are summarized as follows.

1. The experimental values confirm the calculated results.
2. The reduced error of teeth profile shows a great influence upon the gear noise as well as tooth separations.
3. Tooth separations do not occur at the nominal exploitation charge of gear transmission.

REFERENCES

(2) V. Merticaru, Papers of Second Conference on Noise Abatement, Budapest, 1974
INTRODUCTION

Small electrical motors in common use are moderate noise emitters. However because of their small size and flexibility they are being considered for units where stricter noise standards are required. Therefore, research has been pursued to reduce their noise emission at the design stage.

THEORETICAL TREATMENT

The main sources of noise in electrical machines are due to electromagnetic excitation, structural resonances, brush noise, windage noise and out-of-balance noise. In small machines, the electromagnetic noise was found to predominate, this being due to the alternating flux exerting varying forces and moments on the machine structure. Theoretical prediction of this source based on Alger's approach (1) yields

\[ S_I = 121 \times 10^{10} C_n(kr) + 20 \log_{10} C_n'(kr) + 20 \log_{10} (2df) \] (1)

where \( S_I \) = sound intensity in dB; \( C_n \) terms are Hankel functions; \( k \) = wave no., \( r \) = radius of motor yoke; \( r \) = external radius where noise is measured; \( d \) = peak amplitude of yoke surface vibration at frequency \( f \). For the structural resonance calculations theory was developed based on Morse (2) and Beranek (3).

EXPERIMENTAL TECHNIQUES

Pole skewing, increasing or profiling the air gap, and permanent magnets (Bovey 1975, unpublished) were shown to give small noise reductions. On the structural side, predicted methods were tested showing greater promise. One such method was the application of shear damping into the motor yoke. It can be seen from Fig.1 that this method gives noise reductions of 10-15dBA both at full and no-load conditions and throughout the speed range.

CONCLUSIONS

Theoretically-predicted methods to reduce noise of electrical machines were applied and shown to give good noise reductions - in one case up 10-15dBA in overall noise at 18" from yoke surface.

REFERENCES

(2) P.M. Morse, Vibration and Sound, 1948, McGraw-Hill.
MAGNETIC NOISE SOURCES IN VARIABLE-SPEED ELECTRICAL MACHINES

Yang, S.J. Heriot-Watt University, Edinburgh, Scotland, United Kingdom

INTRODUCTION

Thyristor inverters are widely used to provide an adjustable power supply for variable-speed electrical machines. The supply usually contains significant voltage and current harmonics. These harmonics cause 'unwanted' air-gap flux density waves, which in turn produce 'additional' magnetic noise.

THEORETICAL ANALYSIS

For a 3-phase electrical machine, part of the air-gap flux density waves produced by the fifth and seventh current harmonics can be expressed as:

\[ B_{5-7}(\phi, t) = B_{1-5} \cos(5\phi + 5\omega_1 t) + B_{7-5} \cos(7\phi + 5\omega_1 t) + \ldots \]

where \( B_{ij} \) represents the amplitude of a flux density wave due to the \( i \)th space harmonic and the \( j \)th current harmonic, \( p \) is the pole-pair number, \( \phi \) the angular position and \( \omega_1 \) the fundamental angular frequency. The interaction between any two of these flux density waves or the interaction of one of these waves with one of the flux waves produced by either the fundamental or other current harmonics introduces a series of 'additional' radial magnetic force waves. For example, the source components for the radial force waves at a frequency of \( 12f_1 \) (\( f_1 = \) fundamental frequency) are as follows: (1) for zero mode of vibration: \( B_{1-5}, B_{1-7}, B_{5-1}, B_{5-7}, B_{11-5}, B_{11-7}, B_{1-11}, B_{11-11}, B_{5-11}, B_{11-11} \) \( \psi \) \( \phi \) and \( B_{5-11} \); (2) for a mode of vibration equal to \( 6\phi \): \( B_{1-5}, B_{1-7}, B_{5-1}, B_{5-7}, B_{5-11}, B_{11-5}, B_{11-7}, B_{11-11} \); (3) for a mode of vibration equal to \( 12\phi \): \( B_{7-5}, B_{7-7}, B_{5-11}, B_{5-11}, B_{11-11}, B_{11-11} \). It can be shown that there are other force waves at frequencies of \( 6f_1, 18f_1, 24f_1 \), and \( 30f_1 \) with various modes of vibration.

EXPERIMENTAL RESULTS

Noise and vibration tests on variable-speed electrical machines supplied from inverters confirmed the existence of these additional force wave components (Fig.1).

REFERENCES

INTRODUCTION

Of great importance have become the use of acoustic noise as some information about the dynamics of interaction of machine elements.

The method of acoustic signal estimation has been introduced /1/ making use of differential representative and resonant spectra.

MODEL

\[
P_m(i) = S_m + R(i) \quad ; \quad m = m_1, m_2, \ldots \quad i = i_1, i_2, \ldots
\]

where: \(S\) - representative spectrum, \(R\) - resonant spectrum, \(P\) - measured spectrum, \(i\) - measuring point, \(m\) - frequency of excitation /e.g. the frequency of meshing in a gear train/.

BASIC SET OF EQUATIONS

Differential spectra, as invariants of eq. /1/, may be determined basing on the results of measurements.

RESULTS

The differential representative spectrum, determined for a substitutional source that is independent of the choice of the measuring point does not depend on the localization of the sensing device, either. Such a spectrum is being determined in a relative scale of frequency and if bands of fixed percentage are applied, in a certain range of changes of the characteristic frequency of excitation it will not be dependent on this frequency. This provides a basis for the assumption of some evaluation of the elements co-operating dynamics for a given object, which is independent of the sensing device localization and of the frequency of excitation at the given moment.

The differential resonant spectrum is independent of the operation of the investigated object. It is, however, a function of the choice of the measuring point and makes it possible to identify the local resonances within the investigated object.

The method has been verified on the example of gearbox.

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SIMULATEUR ANALOGIQUE DE MOTEUR À COMBUSTION. PREDICTION ET REDUCTION DES BRUITS D'ÉCHAPPEMENT

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INTRODUCTION :

Il est décrit le principe d'un simulateur analogue de moteur à combustion basé sur les analogies acoustiques-électroniques. L'appareil est introduit à l'extrémité de lignes électriques analogues aux tuyauteries d'échappement ; le montage permet la prédiction du spectre des bruits B.F. et l'optimisation de silencieux dans les conditions de régimes réels.

BASES THÉORIQUES :

Le moteur analogue synthétise, en fonction du cycle, les relations thermodynamiques qui régissent l'écoulement du gaz en phase d'échappement, en tenant compte de la cinématique et de la réaction acoustique de la tuyauterie, l'état stable est obtenu implicitement.

Des circuits électroniques linéaires résolvent instantanément, en fonction de l'angle θ de rotation, les équations différentielles correspondantes aux régimes thermodynamiques des écoulements soniques et subsoniques ; pour une vitesse donnée, les principales variables sont les pressions et températures instantaneous dans le cylindre et en aval des soupapes, le débit masse d'échappement, les sections d'ouverture des soupapes et le volume instantané du cylindre. Toutes ces variables correspondent à un analogue électrique et en particulier, le débit masse et les pressions instantanées sont respectivement analogues au courant électrique et à la tension.

Chaque cylindre injecte donc des "bouffées" de courant à l'entrée du simulateur de tuyauteries, et la variation de pression autour de la pression moyenne derrière chaque soupape engendre ainsi le bruit.

RESULTATS ET CONCLUSIONS :

La fig. ci-contre montre un exemple de comparaison entre la mesure expérimentale réelle de pression instantanée (bruit) dans le collecteur d'un moteur 4 cylindres muni en aval d'un silencieux et la tension analogue relevée sur simulateur ; on remarque la bonne concordance entre les spectres de Fourier correspondants.

Le simulateur permet donc, avec suffisamment de fiabilité, la prédiction du spectre des bruits B.F. des échappements connaissant uniquement les caractéristiques cinématiques et dimensionnelles des moteurs.

Les silencieux seront ainsi plus facilement adaptés aux spectres sonores des différents régimes de fonctionnement.

Cette étude a fait l'objet d'un contrat de recherche entre la S.A. BOET et l'Institut de Recherche des Transports (1975)
INTRODUCTION

Although the control of noise in furnaces is often considered mundane, there are many situations in which the major source is difficult to locate and modify. Large natural gas fired boilers in steam power plants are especially prone to elusive combustion instabilities which can cause derating of the boiler to as little as half design rating. The present paper describes a much smaller plant, a one tonne capacity oil-fired rotating drum foundry furnace, as an illustration of some of the problems of finding the source of a combustion instability. The foundry was threatened with closure by the local authority.

THEORETICAL BACKGROUND

Combustion instabilities are of three main types: (i) System instabilities, involving the interaction of various components in a system including air and fuel lines and the exhaust stack; (ii) Chamber instabilities, which depend on the combustion chamber geometry and burner characteristics; (iii) Intrinsic instabilities, associated with the fuel-oxidant reaction kinetics.

Typically system instabilities are low-frequency, chamber instabilities mid-frequency and intrinsic instabilities high-frequency. The first two are under the control of the designer while intrinsic instabilities can only be overcome by a change in the fuel or oxidant.

IDENTIFICATION OF INSTABILITIES IN FURNACE

Octave and third octave spectral measurements at various locations around the furnace for a range of air and fuel flow rates revealed strong peaks at 37.5 Hz, 200 Hz and 400 Hz as well as at the blade passing frequency of the secondary air fan (600-700 Hz). From the dimensions of the system components and the acoustic velocity of the gas in the various components it could be calculated that the most probable oscillation frequency of the combustion chamber was 200 Hz and that of the secondary air duct was 35-50 Hz depending on the temperature, which in turn depended on the flow rate. The primary air line could not be forced into oscillation by downstream effects because sonic conditions existed at the burner nozzle, however oscillations originating upstream of the nozzle could drive system instabilities by affecting the fuel atomisation.

The mechanism of the secondary air duct oscillation is interesting as the secondary air acted as a spring on which the mass of gas in the combustion chamber was 'suspended'. The introduction of a punched plate flow resistance between the secondary air duct and the chamber, to give a measure of frictional damping to the spring-mass system, reduced the 37.5 Hz peak by 10 db to an acceptable level.

The combustion chamber oscillation remained a mystery until a seemingly minor vibration of the primary air pressure gauge was noticed. Inspection revealed that the gauge mechanism had been severely overloaded. For a true pressure of 55psig the gauge reading was 20 psig. Changing the gauge and setting the primary air pressure to the manufacturers specification produced an immediate and dramatic elimination of the instability.

Although the detailed investigation did produce an improvement in the design of the secondary air system, the primary problem was one which could have been corrected or avoided by an alert tradesman.
ON THE CHARACTERISTICS OF INFRASOUND GENERATED BY DAM DISCHARGE

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Introduction

Infrasound is becoming one of the source of environmental pollution. We have done the investigation about infrasound generated from arch style dam discharge by changing the discharge conditions, and obtained following results. 1) Long distance distribution of Sound Pressure Level around the dam site was measured. SPL is about 70 dB at 500 m apart from dam gate. 2) Two distinguished spectrum are found and change these patterns according to the discharge conditions. 3) Main source origin is assumed the fitting surface of discharge fall.

Investigation procedure

Frequency range of measurement system; 0.5 -- 2000 Hz(+1 dB), Discharge conditions; Individual gate discharge (3 gates) and combination gates discharge. Discharge amount; 30 -- 350 ton/Second (per gate). Generated infrasound is random process with lower frequency component, then the analytical equipments have long averaging time constant (about 50 seconds).

Results and conclusions

1. SPL distribution; Fig 1 indicates equal SPL contours from 1st gate(---) and 3rd (---) gate discharge respectively. SPL distributions are affected by geographical features. We can assume that Sound source area may be large from these contours.
2. SPL vs discharge amount; SPL increase according to dam discharge amount, but decrease at full discharge. (Fig 2) It may be caused by the fall figure. The spread figure is obtained at intermediate gate open and straight figures is at full open.
3. Spectrum; Distinguished features are that two dominant spectra are shown in Fig 3. Lower spectra do not change by the discharge condition, but second spectrum (about 5--10Hz) shift by changing discharge amount.
4. Infrasound source; In order to obtain the sound source point, 3 microphones set at rectangular point with some distance and measured time delay of sound pressure patterns between two points. From this time lag, direction was determined. Source direction fluctuates but the estimated main source should be water surface movement area of discharge fall point.
INTRODUCTION

Recent legislation in the UK (1) and an accompanying code of practice (2) on the control of noise from construction and demolition sites has made the study of noise from construction machines relevant. In particular it was found that there were few data available concerning the characteristics of noise from impact pile drivers. This paper will describe the results of a measurement and analysis programme designed to fill this need.

A series of recordings of noise from conventional pile drivers (diesel, pneumatic, and drop hammers on sheet and bearing piles) were made during the course of 1975/6 on working civil engineering sites. In late 1976 a set of recordings were made of noise from six 'quiet' pile driving devices at a demonstration organised by the British Steel Corporation and Construction Industry Training Board. In all cases data were gathered with precision grade instruments at distances ranging from 15 to 30 metres from the machines.

ANALYSIS AND RESULTS

Analysis procedures were developed based upon the ISVR PDP 11/45 data analysis computer which allowed close examination of pressure waveforms and sound pressure level time histories. The effects of different integration times were also examined.

It was concluded that in most cases noise from impact pile drivers can be characterised as an exponential pulse and that any departure from this shape has little effect on the equivalent sound level. The implication of this is that it is the sound resulting from hammer impact rather than exhaust or other secondary noises that dominates the acoustic output from these devices.

In all cases peak and equivalent sound levels were determined. In the table below these are presented for the range of quiet pile drivers and the conventional hammer measured at the demonstration mentioned above. Data are presented with relation to three separate days of demonstration.

These results will be amplified in the full paper together with the results of the survey of noise from conventional pile drivers (3).

<table>
<thead>
<tr>
<th>DEMONSTRATION DAY</th>
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<td>RANGE OF MAXIMUM DB(A)*</td>
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<td>105-107</td>
<td>105-107</td>
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<td>107-107</td>
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</tbody>
</table>

** = Measured at a height of 1.5m and at a distance of 15m from base of sheet pile. ** = Extraction and driving operations analysed together.

REFERENCES

(2) Noise Control on Construction and Demolition Sites, British Standard 5228, 1975.
(3) J.E. Ludlow, Noise from Impact Pile Drivers, ISVR Memorandum 541, 1976.
NOISE SURVEY IN PETROLEUM REFINERIES IN HAIFA

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PURPOSES

The purposes of the survey were:

(a) to determine the noise levels in the different departments in dB(A) as well as noise criteria NR.
(b) to specify the characters and possibilities of desirable noise control measures and their priorities.
(c) to test several types of ear muffs for sound attenuation in the different octave bands and to select the suitable models according to the noise levels in the different departments.
(d) to provide the data for the noise control measures required in the different work stations or departments.

FINDINGS

1 Noise levels were recorded at 243 different spots, generally selected for their high noise levels. It was found that, in 44.9% of the cases noise levels of less than NR85 were found, in 28% - NR86 to NR90, in 19.3% - NR91 to NR100, in 7% - NR101 to NR110, in 0.8% - NR111 to NR117.
2 The two most noisy spots were found in the workshop, near the pneumatic mail cleaning device (NR113; 109dB at 4000Hz) and near the pneumatic hand tool testing device (NR117; 119dB at 500Hz).
3 Most noisy spots, in different departments are near the furnaces (NR101 to NR110 maximum level at 250Hz).
4 Certain operations in the periodical overhaul expose the personnel to high noise levels, as for instance cleaning welded areas (NR109; 108dB at 2000Hz) and cleaning furnaces (NR105; 103dB at 2000Hz).
5 There is a wide range of differences between the ear muffs utilized in the plant as well as between those which can be purchased locally. The results of the test show differences between 3dB to 23dB noise attenuation in the octave band range of 125Hz for instance.

CONCLUSIONS

In addition to the fulfilment of the purposes specified above, the survey describes the noise control measures recommended for the different places. These are divided into three groups: (1) places where noise control is indispensable and must therefore be carried out urgently, (2) places where standardized noise control devices can be utilized, (3) some places where the noise level does not reach NR85, but where quieter working conditions are desirable. (NR85 is the maximum permissible noise level according to the Ministry of labour for an 8 hour working day without individual protection).

Regarding ear muff selection, the survey provides recommendations as to the types to be used in different departments also taking into account the safety aspect of communication (mainly at frequencies of 1000 to 4000Hz) and cost considerations.
INTRODUCTION

We measure the A-weighted noise levels and their octave analysis of selected parts of twenty factories in Singapore. This is the objective response. We also interview selected workers in the above factories using questionnaires. This is the subjective response.

RESULTS OF NOISE MEASUREMENTS

The categories of the factories are: metal industries, shipyards, oil refineries, electronic industries, textile industries, mechanical services, gas plant and soft drink industries. The noise measurement results show that the noisiest factories are textile industries and metal industries. Fifty percent of the factories measured have noise levels over 90 dBA. This shows that noise is quite a pollution problem in Singapore factories. The plots of the octave frequency analysis of various types of machineries selected agree with the usual shapes of curves and these are useful for future noise abatement programs in those factories.

IMPACT OF PLANT NOISE ON THE WORK ENVIRONMENT

This is investigated by interviewing selected workers in those parts of the factories that we have done the noise measurements, using questionnaires. The contents of the questionnaires include items like identifying the sources of the loudest noise and rating of noise conditions in the factories using rating scales etc. The results agree with that obtained from noise measurements.

NOISE REDUCTION PROGRAMS FOR INDUSTRIAL PLANTS IN SINGAPORE

From the above surveys we conclude that noise pollution is quite a serious problem in Singapore industries. So far the noise legislation (1) in this country is not enough. We only have a Factories Act which includes a small clause stating that workers can claim for compensation if their hearings were damaged. Therefore a Noise Control Act should be passed which must state the maximum noise levels allowed and the length of exposure time permitted for such a level.

METHODS OF APPROACH

The potential for reducing interior noise of industrial plants is in general excellent. The engineering and architectural techniques for reducing this noise along its transmission paths are known at present. However, reducing the noise at its source may be difficult and expensive, often resulting in the degradation in performance of the equipment, machine, or process.

REFERENCES

(1) N. Boratynski, Noise Control (1956)2, 37-46
NOISE SOURCES IN DEVELOPING COUNTRIES

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INTRODUCTION:

In the developing countries noise emanates not only from traffic and industry as in developed countries but the social customs traditionally handed over also acts as a contributory noise source. For instance, marriage and funeral ceremonies are marked by lots of noise in marked contrast to what happens on similar occasions in the developed countries.

THEORETICAL BASES:

From our study on noise pollution we observed that our noise sources were peculiar and different from that of Western Countries. Hence it was decided to measure the noise sources individually.

EXPERIMENTAL TECHNIQUES:

The noise levels were measured using sound pressure level meter at selected industrial complex, road traffic sides and at community activities. The measurements were made at the level of the ear over a period of time. The condenser microphone used in the SPL meter for this study was one inch in diameter having a frequency response of 10 Hz to 20,000 Hz with 1 dB variation. Every day before commencing the study, the output of the SPL meter was checked using a pistonphone.

RESULT AND CONCLUSIONS:

Apart from industrial complex, the traffic noises have a great effect on the population as in tropical countries the windows are all open and traffic noises easily reach the house-hold, whereas in industrial complex we have a noise level of about 95 dB, along major trunk roads the traffic produces as much as 85 dB. Among community gatherings in fairs and Shandy the noise level is 95 dB. During marriage ceremony 101 dB and during Diwali festive crackers (Chinese) 103 dB, and in a Community dance (Karagam) the noise reaches a level of 95 dB. A child born in such a society is literally exposed to loud noise source from birth till death. This situation is likely to continue till a major social reformation occur in this country.
Le signal acoustique transportant une quantité d'information très importante -grande dynamique et large bande passante- le problème de la mesure acoustique est de réaliser efficacement une compression de l'information.

Compression efficace signifie qu'il y a réduction effectuée du débit d'information, et ce, dans des proportions considérables si l'on veut pouvoir utiliser commodément les résultats, mais avec une perte réduite de l'information relative aux paramètres étudiés.

Dans le cadre de l'étude des bruits fluctuants sur de longues durées, ces caractéristiques sont :
- les niveaux moyens, (méthodes intégrales, analyse statistique)
- les écarts de niveau (analyse statistique)
- l'énergie acoustique totale (problèmes de dosimétrie).

Les moyens mis en œuvre (méthodes intégrales, analyse statistique) seront plus ou moins bien adaptés à cette compression de l'information, et réaliseront plutôt qu'une mesure, une estimation des paramètres utiles du phénomène.

Il s'agit bien d'une estimation car la mesure est incomplète (durée limitée, prise en compte d'une partie seulement des valeurs réelles).

Notre étude va donc porter sur la convergence de ces estimateurs dans le cadre d'hypothèses simplificatrices mais réalistes.

La méthode d'analyse statistique met en œuvre des estimations de probabilité ainsi que des moyennes de position (indices statistiques Lx). L'étude de la convergence de ces estimations va nous conduire à mettre en évidence l'influence de la largeur de classe, de la durée de la mesure et enfin la difficulté d'estimation de certains paramètres (extrémités des distributions).

Les méthodes intégrales mettent en œuvre des estimations de moyenne. On verra de nouveau apparaître l'influence du temps de mesure ainsi que la largeur de classe.

Les résultats généraux obtenus dans cette étude permettent de juger de l'adaptation des systèmes de mesure existant au problème particulier et de donner les conditions d'utilisation de ces systèmes en vue d'une plus grande efficacité.

Ils nous permettent de préciser les limites de validité des différentes méthodes ainsi que de définir l'instrument le mieux adapté au problème considéré.
La majorité des études du bruit lié à la circulation automobile est fondée sur des hypothèses de fluidité. Ces hypothèses qui conduisent à des expressions analytiques simples et qui sont réalistes dans le cas des voies rapides, sont rarement vérifiées en milieu urbain.

Le trafic urbain réel pose un problème complexe par son manque de fluidité, par son irrégularité et par la disparité des véhicules qui le composent.

Il est, en outre, nécessaire de faire intervenir les conditions de propagation : la géométrie de la rue, son temps de réverbération ainsi que les pulsations liées aux feux ou simplement à la proximité d'un carrefour.

Le but de cette étude est la prédiction des niveaux acoustiques à partir des paramètres de circulation fournis par le réseau de capteurs de la ville de TOULOUSE. Il ne peut s'agir que d'une étude macroscopique car les paramètres utilisés, nombre de véhicules et temps d'occupation des capteurs, ne permettent pas d'appréhender la nature des véhicules et ne donnent qu'une idée approximative de leur distribution le long de la voie.

Le choix des paramètres de bruit utilisés a été guidé par la nécessité d'obtenir un paramètre de valeur centrale et un paramètre de dispersion. Le $L_1$, donné par une intégrale énergétique, a été choisi parce que bien représentatif de l'énergie reçue, quant au $L_{10}$, sa sensibilité aux crêtes de bruit l'a fait préférer au $L_{10}$, plus sensible, mais souvent moins significatif.

L'acquisition d'un grand nombre de mesures acoustiques a été rendue possible par l'utilisation d'une station automatique fournissant sur cassette digitale, des données directement utilisables sur calculateur.

L'analyse des résultats relatifs à plusieurs rues, de configurations géométriques différentes, et soumises à des régimes de circulation différents a permis de proposer un modèle de prévision, à partir des données circulation, et tendant à prendre en compte la géométrie de la rue.

Une étude plus fine doit permettre d'introduire dans le modèle des constantes liées à la nature de la source et aux conditions géométriques complexes. Ceci permettrait de réactualiser le modèle en fonction de l'évolution des règlements sur le bruit des véhicules.
II. Noise and vibration

F. NOISE CONTROL
INTRODUCTION

Because of their very favourable power-to-weight ratio and their high energy conversion and transmission efficiencies, hydraulic power transmissions systems are more and more widely used in various applications fields. They tend to be noisy systems. The recent introduction of still higher speeds and pressures in these systems, to further increase the power-to-weight ratio and the efficiency, is just making the situation worse in terms of noise production. To reduce and control the noise climate active and passive measures are currently been developed.

ACTIVE MEASURES

Noise sources in hydraulics are mainly related to fluctuating pressures in fluids and forces in structures. Noise reduction can be achieved by cinematic modifications and hydrodynamic operation optimisation, particularly so for pumps and motors.

To reduce fluid-borne noise, various types of hydroacoustic filters are available.

PASSIVE MEASURES

Noise reduction can also be achieved in various other ways by use of more passive measures such as specially designed hoods for characteristic hydraulic noise frequencies range or close-fitting housings, internal structural damping in the various system elements.

EXPERIMENTAL TECHNIQUES

A large anechoic chamber has been designed and built on purpose specially for noise reduction techniques in the hydraulic field. The power available in the room is now 300 kW and the back ground noise level, with full power, has been kept low enough to allow meaningful acoustical measurements to be made in all conditions in the usual engineering frequency range.

RESULTS

As this installation has been made in a typical hydraulic power transmission environment, to make use of the hydraulic power centrale (Power: 1 MW), its design and construction has made it possible to test and evaluate various noise and vibration techniques in relation to their possible application on a large scale in the hydraulic industry.
INTRODUCTION

Le bruit d'un turbo-alternateur résulte de plusieurs sources (1) dont celles qui constituent les ensembles inter-corps couramment désignés par paliers. La puissance acoustique de ces ensembles se situe actuellement, sans aménagement particulier, autour de 100 dB(A) pour les machines dont la vitesse nominale est égale à 3 000 tr/mn et autour de 85 dB(A) pour les machines dont la vitesse nominale est égale à 1 500 tr/mn.

Des solutions simples sont possibles qui réduisent ces puissances acoustiques.

LES ELEMENTS

Les ensembles inter-corps comprennent, en général :
- deux coussinets à huile (diamètre = 600),
- un accouplement (diamètre = 1 400),
- des barrières d'étanchéité alimentées en vapeur.

L'AMENAGEMENT DES SOURCES

Le bruit émis par les coussinets est, en général, faible par rapport aux bruits émis par l'accouplement et les barrières.

Pour éviter l'effet de sirène, la tête des broches et les écrous associés doivent être protégés ou noyés dans le corps de l'accouplement ; la fréquence de l'amplitude dominante est égale au produit du nombre de broches par la vitesse de rotation, par exemple : (24 x 25 = 600 Hz).

L'écoulement de la vapeur à travers les barrières d'étanchéité peut produire un sifflement dont la fréquence de l'amplitude dominante se situe autour de 2 000 Hz, par exemple 102 dB dans la bande 2 250 - 2 800 Hz à 0,3 m.

Une organisation convenable des barrières permet une réduction de 10 dB environ.

A ces bruits se superpose un bruit de fond dû au frottement du rotor dans l'air ; les vitesses varient autour de 75 m/s environ.

CONCLUSION

Des aménagements simples permettent des réductions importantes du bruit émis par les sources élémentaires ; de plus, le raidissement des carter par nervurage et augmentation des épaisseurs et leur insonorisation par des matériaux absorbants permettent d'obtenir une réduction de 15 dB(A).

REFERENCE :

(1) "Steam turbine-generator design and noise" — R. BIGRET et J. DELCAMBRE
Les corps basse pression des groupes turbo-alternateurs sont une source sonore importante dont le rayonnement s'étend sur une large gamme de fréquences. Ce rayonnement est dû à des sources internes puissantes liées aux écoulements fluides et aux excitations mécaniques de structures.

L'énergie acoustique rayonnée à l'extérieur est fonction de paramètres qui caractérisent la réponse des structures et notamment de l'atténuation des parois et des modes propres acoustiques des volumes internes.

**BASES THÉORIQUES**

Dans une approche simplifiée, on peut assimiler un tel ensemble à une salle, de symétrie cylindrique, comportant des diffuseurs et de multiples sources sonores internes. Pour l'une de ces sources \( j \), on peut écrire :

\[
L_{p1,j} = L_{w,j} + 10 \log \left( \frac{1}{E_{ij}} + \frac{4}{R} \right)
\]

\( L_{p1,j} \) étant le niveau sonore dû à la source \( j \) en un point \( M_i \) voisin de la paroi.

\( L_{w,j} \) étant le niveau de puissance acoustique de la source \( j \) à la fréquence \( f \).

\( E_{ij} \) étant une surface caractéristique de la géométrie du système, du point \( M_i \) et de la source \( j \).

\( R \) étant la caractéristique acoustique du volume intérieur, fonction de \( f \).

On en déduit : \( L_{p1} = L_{p1,j} - TL \)

\( L_{p1} \) étant le niveau sonore en un point \( M_p \) voisin de la paroi et proche de \( M_i \).

\( TL \) étant l'affaiblissement dû à la paroi du corps basse pression.

Si l'on connaît l'émission sonore des sources internes, il convient de déterminer les valeurs \( R(f) \) et \( TL(f) \) pour préévaluer le rayonnement extérieur de l'ensemble et pour envisager une action sur l'un ou l'autre de ces paramètres en vue de réduire le bruit rayonné à l'extérieur, \( f \) étant la fréquence.

Parmi plusieurs méthodes possibles pour déterminer \( R(f) \) et \( TL(f) \), le choix s'est porté sur :

- une méthode dynamique pour la mesure de \( TL(f) \),
- une méthode classique pour la mesure de \( R(f) \),

a partir de la mesure du temps de réverbération des volumes internes.

Les manipulations ont été effectuées à l'intérieur de l'un des corps basse pression d'un groupe turbo-alternateur de 900 MW, avant sa mise en service.

**RESULTATS ET CONCLUSIONS**

L'isolement du corps est nettement plus réduit que la valeur prévisible à partir d'un calcul classique d'isolement de parois, et la valeur du temps de réverbération décroît de façon régulière en fonction de la fréquence. Il en résulte qu'une réduction du bruit des corps basse pression peut être obtenue, d'une part, en recherchant une atténuation plus élevée des parois et, d'autre part, en diminuant la réverbération des volumes internes.

Suivant les impératifs qui limitent la puissance acoustique de ce matériel, soit pour la protection du personnel d'exploitation, soit pour la protection du voisinage des usines, les moyens de réduction de bruit peuvent donc être envisagés à partir de la mesure de ces paramètres et de la connaissance des matériaux technologiquement compatibles avec le fonctionnement de cette partie d'un groupe turbo-alternateur.

**REFERENCES**

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THE REDUCTION OF NOISE FROM KNITTING MACHINES

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INTRODUCTION

An extensive survey of working noise levels found in those parts of this United Kingdom textile industry which produce knitted goods (1) indicated that one of the major problem areas concerned with machinery used for the production of ladies seamless hose where the mean Leq was found to be around 92dB(A). An exercise to study the noise-generating mechanisms in such machinery was undertaken with the aim of reducing mean noise levels to below 85dB(A).

STUDIES OF NOISE GENERATION MECHANISMS

Machines of this type operate through a work cycle which shows significant time variations in noise output with two major peaks at which the level rises by 10 to 15dB(A). These peaks correspond to the operation of a compressed air jet used to induce mechanical operation of the knitting needle latch combination with an oil-mist lubrication system. This jet is only approximately 1.6mm diameter and at its operating velocity of around 450 m.s⁻¹ would, in isolation, generate a level of around 82dB(A) at 1m; however, the turbulence created by the impact of this jet on the needles results in a level of around 89dB(A) being generated, to which must then be added the noise resulting from all other elements of the machine.

RESULTS AND CONCLUSIONS

From the noise generation studies it will be seen that the prime requirement is to reduce the noise of the air-jet. However it is not possible to achieve the desirable 7 to 10dB(A) by reducing velocity, which is normally the simplest way of reducing airjet noise (2), and the physical complexity of the machine has so far defied attempts to utilise add-on jet silencers. Thus it has been found necessary to adopt a solution which is technically less elegant, but in practice just as efficient, utilising a partial acoustic screen to surround the top section of the machine. Noise reductions of the order of 10dB(A) were established to result from the use of such a screen in laboratory tests. The dependence of noise reduction on the screen material used is illustrated in Figure 1, where the theoretical attenuation provided by the screen, which of course depends on normal mass-law considerations, is shown to be heavily modified by its use for only partial enclosure. This results in an increase in material mass/area above around 1000 g.m⁻² giving little improvement, since the overall attenuation is limited by the level of the 'noise floor' arising from the remainder of the machine, and this interpretation is seen to be in good accord with the test data.

Having established the validity of the technique under laboratory conditions, factory tests have been carried out using screens of 2500 g.m⁻² material, and incorporating some further element of absorptive treatment in their construction. These trials gave substantial confirmation of the laboratory tests and resulted in acceptable working noise levels being achieved, together with the equally-important achievement of acceptable machine accessibility and process viewability.

(1) G. M. Coles, Hatra Research Report 43
(2) G. M. Coles, Aero Quarterly, 14, 1-16
In system identification techniques the structure of the model is generally assumed a priori and the unknown parameters within it then "identified" by such methods as least squares curve fitting or autoregressive moving averages. This is reasonable when the physical system is well understood, as in linear multi-degree of freedom vibrations. In contrast, a strategy is outlined here for the initial selection of a model structure in more complex cases of strongly nonlinear and continuous systems. The emphasis is on the extraction of a simple model which captures the qualitative dynamical behaviour in some local domain and recent mathematical developments in differentiable dynamics are used [1].

It is assumed that the (unknown) equations of motion can be expressed as a dynamical system \((M,X_u)\) [1], where \(M\) is the state space and \(X_u\) a dynamic acting on it whose form is determined by \(\mu\in\mathbb{R}^m\), a multi-dimensional control parameter. One needs little or no more information on this notional "complete" model but with this scheme in mind one concentrates on qualitative changes in behaviour in the state space, such as the onset of divergence (buckling) or flutter, which occur as external parameters \((\mu)\) are varied. These changes are modelled by bifurcations, which almost always occur on a low dimensional subset \(\mathcal{W}\) of the state space. In seeking local models for phenomena such as stability loss one is thus justified in restricting attention to a subsystem or essential model \((\mathcal{M},X_u)\subset(M,X_u)\) where \((\mathcal{M},X_u)\) is rigorously defined by use of centre manifold theory [2]. \((\mathcal{M},X_u)\) can be considered as a set of ODEs \(\dot{\mathbf{z}} = X_u(\mathbf{z})|_{\mathbf{z}\in\mathcal{W}}\), where \(n\) is the dimension of \(\mathcal{M}\), and one can thus use the classification of generic (= typical [1]) bifurcations for 1,2 ...parameter ODEs in the selection of topological or qualitative structures for essential models. The strategy is to experimentally observe parameter values at which qualitative changes occur and then to attempt to (locally) match this measured bifurcation set with one of those structurally stable sets classified in the literature (eg [3]). Structural stability implies that the model will retain its qualitative features under small perturbations and that the normal form of the model thus chosen provides a reasonable candidate for conventional parameter fitting. This approach is developed in detail and with many examples in the report [4].

In cases in which the governing equation is known, centre manifold and bifurcation theory can also be used to simplify the analysis of complex nonlinear problems [5]. The concepts outlined here should thus be seen as additional tools for the engineer interested in problems of vibrations and dynamics rather than as a specific approach to a single class of problems.

REFERENCES

HIGH-DUTY ACOUSTICAL SIREN

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INTRODUCTION

At the Institute of Physics, Pedagogical College of Rzeszów, there was performed, fifth in Poland, a model of acoustic siren designed by the lead author. The siren is provided with elongated ports, ensuring times when ports are fully opened and fully closed being longer than closing and opening times. This is then possible to approximate the rectangular pulse which, accordingly with (3), gives twice as great efficiency as the sine pulse under the same conditions:

\[ \eta_{ac} = \frac{I_m \cdot R}{2E} \quad \text{sine pulse} \]  
\[ \eta_{ac} = \frac{I_m \cdot R}{E} \quad \text{rectangular pulse} \]  

where \( I_m \) - volume flow through ports,  
\( R_m \) - real part of throat impedance,  
\( E \) - positive gauge pressure in siren's chamber.

EXPERIMENTAL

On the basis of theoretical works (4), (5) the siren was tested at the experimental stand. The following quantities were measured: power, efficiency, wave-form, the effect of the pulse shape on the wave-form, the effect of tube characteristics and the influence of non-linear phenomena. The examination of the siren was carried on in an anechoic chamber using fully automatized measuring system (B+K microphones connected with a testing equipment for taking sirens directional response pattern). The results we have obtained proves that under proper conditions the acoustical efficiency of the siren can achieve the value of order of 30%. There was also observed that the siren shows much greater drop of power and efficiency with increasing frequency than that prescribed by the classical theory (1), (2).

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ETUDE ET REALISATION D'UNE CHAMBRE SOURDE AVEC SOUFFLERIE AERODYNAMIQUE, POUR RECHERCHES AERO-ACOUSTIQUES

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BUT : Les caractéristiques d'émission sonore des jets et compresseurs de moteur d'avions sont différentes pour les moteurs au point fixe, d'une part, et en vol d'autre part. Le développement de systèmes suffisamment peu bruyants pour satisfaire les règlements de certification acoustique des avions, nécessite des moyens d'essai simulant la vitesse d'avancement. C'est pourquoi le CEPr et l'ONERA réalisent en commun, à Saclay, une soufflerie à veine libre placée dans une enceinte anéchoïque, pour étude des sources aéroacoustiques en mouvement relatif.

La soufflerie comporte d'amont en aval : un silencieux de section carrée (9x9m) un convergent partiellement amovible conduisant à une veine d'air cylindrique de 2 m de diamètre et 8 m de long, ou de 3 m et 9 m de long. Cette veine libre, située dans un quart de sphère à parois absorbantes, de 9,6 m de rayon, est prise par un diffuseur approprié de 20 m de longueur. Un silencieux de même section que le premier isolé acoustiquement la salle d'essai du ventilateur d' entraînement de la soufflerie, à vitesse de rotation variable.

Les moyens de mesure sont adaptés à la double fonction acoustique et aérodynamique : microphones fixes et mobiles dans toute l'enceinte anéchoïque, filtres, corrélateurs, etc... ordinateur d'acquisition et traitement de signal en temps réel ; mesures aérothermodynamiques, reliées également aux calculateurs, moyens de visualisation, etc...

Des études sur maquettes à échelle 1/10 ont été menées pour compléter les calculs aérodynamiques et acoustiques faits par ailleurs. On a déterminé ainsi les profiles de vitesse dans la veine, les vitesses de recirculation d'air dans la chambre et les qualités anéchoïques de celle-ci.

D'autre part des études théoriques et expérimentales en vraie grandeur de diffusion et de réfraction par les couches de cisaillement de la veine fluide ont été faites pour connaître les corrections à apporter aux mesures acoustiques.

Les performances escomptées sont : vitesse d'air de 100 m/s (plus tard 130 m/s) dans la veine de Ø 2 m, 60 m/s dans celle de Ø 3 m ; fréquence de coupure de l'ordre de 200 Hz.

Il est prévu que les essais de réception dureront jusqu'en septembre 1977 après une mise en route en janvier 1977.
MINIMUM NOISE OF AXIAL FLOW FANS

In reducing the aerodynamically generated noise of sound sources, as e.g. of an axial flow fan, both acoustic and aerodynamic aspects have to be considered. A widely used empirical formula /1/ with \( V \) m/s flow rate and \( \Delta p \) mm w.g. system loss

\[
L_p = 60 + 10 \log V + 20 \log \Delta p \quad /\text{dB}/
\]

serves to estimate its radiated power. Assuming a "rotor only" construction the dominating sound source will be the presently unavoidable vortex shedding. It can be proved that the noise of the turbulent blade boundary layer and the oncoming turbulence is negligible.

The radiated sound power of \( N \) blades is /2/

\[
P_{vsh} = \left( \frac{\rho V^2}{120 a^3} \right) \int_{R_t}^{R_h} \frac{l w^6}{Re} \frac{\Delta p}{a} \, dR
\]

where \( \rho \) is the density, \( a \) the velocity of sound, \( l \) the chord, \( w \) the relative velocity, \( Re \) the Reynolds number, \( R \) the radius, \( t \) indicates the tip and \( h \) the hub. Approximating \( w \) by the peripheral speed \( u_t \), taking \( l \) constant and introducing \( D = 2R_t \) as reference dimension

\[
P_{vsh} = N p (1/D)^{0.6} V^{0.4} D^{1.6} u_t^{5.6} \left[ 1 - \left( \frac{R_h}{R_t} \right)^{6.6} \right] /5.10^3 a^3
\]

where \( \nu \) stands for the kinematic viscosity. Neglecting \( (R_h/R_t)^{6.6} \) and introducing the head and flow coefficients

\[
\psi = \frac{\Delta p}{(\rho u^2/2)} \quad \text{and} \quad \phi = 4\nu u_t D^2 \pi
\]

then

\[
P_{vsh} = KV \Delta p^{2.3}
\]

where \( K \) contains all parameters \( \psi, \phi, N, 1/D, \zeta, \nu, a \).

SYSTEM IMPROVEMENT

With average figures from axial flow fan engineering practice and introducing sound power level

\[
L_p = 50 + 10 \log V + 23 \log \Delta p \quad /\text{dB}/.
\]

It is the minimum level of broad band noise generated by vortex shedding, indicating that the reduction of the system loss is the best way of noise reduction.

In reducing the system loss by e.g. rounding off the front of the splitters in silencers, contraction loss and a part of the self-noise may be lowered. The same time reflection and so the insertion loss reduces as well. Although fan noise will be lower, calculation shows that it will not compensate the reduction of the insertion loss. Aerodynamic improvement usually means acoustic improvement as well. There are exceptions, hence careful consideration is recommended.

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INTRODUCTION

Circular power wood cutting saws have been shown to cause annoying and high level noises. An attempt, to reduce the noise emanating from such saws, has been made by using a thin layer of viscoelastic material as a constrained damping medium forming a three layer laminate. Such a blade was fabricated by properly adhering two standard 7 1/4" diameter 40 teeth circular blades with a viscoelastic material.

The method of constrained damping has been shown to effectively damp vibrations in beams, plates and rings. A simple test, striking the standard blade and the laminated blade, showed a marked reduction in 'ringing' of the laminated blade as sensed by the ear.

THEORETICAL CONSIDERATION

The sounds emanating from a power saw are in part attributable to:

1. Aerodynamic noise of the teeth moving through turbulent air and passing the opening in the table twice per revolution.
2. When cutting, the wood - blade - and its attachment have a different set of natural frequencies than when the blade is free.
3. The noise level is affected by: a. The kind of wood (hard or soft); b. The direction of cut, rip or cross-cut; c. The speed of rotation of the blade; d. The rate of feeding the wood to be cut; e. Blade teeth form and sharpness.
4. The absorption and geometrical properties of the room affect the noise levels.

EXPERIMENTAL TECHNIQUE

The standard and experimental saw blades were tested as follows:

1. Supporting each of the blades with string, each was given an impulse and the sounds compared by the human ear and a spectral analyzer.
2. The blades were tested, one at a time, in the same manner when placed in position onto a typical wood cutting machine. When possible, spectral analysis was obtained for the noise. A (B&K) Precision Sound Level meter Type 2203 with a 1613 type octave band filter was used to measure noise on the overall linear and A scales along with octave band readings. In all tests 5/8" pine wood was cut and the noise was measured when: a. Rotating blade - not cutting; b. Cutting in the rip direction of the wood; c. Cutting in the cross direction of the wood.
3. The above tests were conducted in a machine shop and outside in an open area.

RESULTS AND CONCLUSIONS

The blades, suspended by string and given an impulse, showed dramatic differences between the standard and experimental blade. The standard blade would 'ring' for some time whereas the experimental blade had no 'ringing' sound at all.

The tests indoors and outdoors gave similar results. When operating in a non-cutting mode the standard blade had an annoying pure tone noise whereas the experimental blade eliminated this sound and showed an overall reduction of 3 to 5 dBA.

Small differences in noise levels were observed between the two blades when wood was cut.

The preliminary conclusion to be drawn is that the experimental blade does reduce noise during that period when wood is not being cut, but that further work is needed to better understand the noise generated when cutting wood so that this aspect of noise may also be reduced.
Les scies circulaires sont des machines très bruyantes et répandues. A vide comme en charge, le bruit émis par une scie circulaire est dû en grande partie aux vibrations latérales de la lame.

Pendant la rotation à vide, la lame, excitée par les tourbillons alternés, entre souvent en résonance suivant une de ses fréquences naturelles ; le bruit émis peut atteindre alors à un mètre des niveaux allant jusqu'à 108-110 dB lin, référence 2.10-5 N/m². En faisant varier la vitesse de rotation de la lame, on excite successivement les différentes fréquences naturelles de la lame. A chaque fréquence naturelle correspond une plage de vitesses pour lesquelles la scie est très bruyante. Ce phénomène de résonance en rotation à vide se constate particulièrement sur les lames avoyées. La totalité des 53 lames avoyées essayées se révèlent comme résonantes et bruyantes. Par contre, sur 14 lames à denture rapportée, dont le corps et les dents sont plus épais, seules 3 sont résonantes.

Au cours du sciage, à l'excitation due aux tourbillons s'ajoute l'interaction avec le bois scié. Ce dernier excite la lame, mais peut également jouer, dans certains cas, le rôle d'amortisseur. Au cours d'une passe, le niveau de bruit varie de façon assez aléatoire. A un mètre, il atteint en moyenne des valeurs allant de 93-95 dB lin à 105-107 dB lin selon la lame, la vitesse d'amenage, l'essence du bois, etc... Les faibles valeurs (93-95 dB) correspondent aux lames à denture rapportée. Indépendamment de ces niveaux moyens déjà élevés, des pointes stridentes atteignant 110-112 dB lin ont été souvent constatées.

Les moyens destinés à réduire le bruit tels que : frotteur mécanique, lame sandwich, présentent des inconvénients. Un dispositif simple, ne changeant en rien la façon de travailler, permet de réduire de façon notable le bruit des scies circulaires. Il se compose d'une plaque de bois couvrant à peu près le tiers de la surface de la lame. Cette plaque est située, grâce à un support adéquat, à 1 mm de la lame sans la toucher et face à la partie non travaillante de la lame (figure 1).

En rotation à vide, ce dispositif permet de supprimer la résonance sur toutes les lames. Le gain obtenu atteint 19 à 20 dB si la lame était très bruyante au départ. En moyenne il est de l'ordre de 15 dB pour les lames résonantes (figure 2).

Au cours du sciage, le gain n'est que d'un ou deux décibels sur les lames à denture rapportée qui sont déjà relativement silencieuses ; par contre, sur les lames avoyées, il peut atteindre jusqu'à 8 dB.

En conclusion, on peut dire que l'utilisation de lames à denture rapportée non résonantes ou celle d'une plaque jouant le rôle d'amortisseur, constituent des solutions raisonnables pour réduire le bruit des scies circulaires.

![Fig. 1 : Dispositif silencieux](image1.png)

![Fig. 2 : Niveaux de bruit émis par des lames résonantes en rotation à vide](image2.png)
Supersonic jets undergo strong changes in their gasdynamical and acoustical properties when surrounded by a burning coaxial jet (1), (2).

EXPERIMENTS

The jets were investigated by acoustical, optical, pressure- and temperature-testing methods. Exit diameters of the supersonic nozzles were in the mm-regime and the jets discharged into the free atmosphere. Heating was provided by propane or acetylene flames arranged coaxially to the supersonic nozzle.

RESULTS

The length of a supersonic part of the jet may be extended up to a factor of 10 by coaxial combustion. At the position where the temperature of the jet core rises the supersonic jet breaks down and mixes with the ambient flow medium. For this flow configuration, two clearly separated noise sources can be distinguished. One in the immediate neighbourhood of the nozzle exit is due to combustion; the other coincides in space with the point of the supersonic jet breakdown (3). The noise radiation of the heated jet generally shows a tendency to become more uniform in angular distribution. This is due partly to the suppression of screech noise as shown in Fig. 1 and partly to combustion-noise. The overall level may be reduced by appr. 6 dB with proper adjustment of the combustion.

Fig. 1: Frequency spectra of the noise of a cold (a) and of a heated (b) supersonic oxygen jet. (Nozzle diameter \( d = 1.3 \) mm, stagnation pressure \( p_0 = 4 \) bar, stagnation temperature \( T_0 = 20^\circ \)C, angle between microphone and jet axis \( \alpha = 90^\circ \), distance between microphone and nozzle \( r = 500 \) mm, filter bandwidth \( \Delta f = 1 \) kHz.)

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SILENCING OF THE CSR-105 CANNON

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INTRODUCTION

In the surrounds of an Artillery Arsenal, there is a town in which some buildings show damages from the effects of blasts generated by CSR-105 cannon shots during factory tests which have caused serious community reactions.

To solve this, we have made many measurements and analysis of signals and developed a model silencer with the object of designing a full scale silencer at the factory.

THEORETICAL BASIS

There is very scanty bibliography on this classified subject. The theory of blast propagation and attenuation is too extensive but the spectral shape and intensity of blasts can be reasonably predicted from artillery data.

Some attenuation of the blast by the atmosphere and natural barriers was observed, but it is well known that infrasound travels long distances unattenuated, damaging structures and generating standing waves and vibrations in buildings.

EXPERIMENTAL TECHNIQUES

Measurements were made of the Sound Pressure Level around the cannon up to distances of 1500 m.

For this, a Pressure Peak Meter and Oscilloscope were used, with a frequency response of 0.1 Hz to 7 KHz.

Fourier Transform was used to study the Frequency Spectrum of the Oscillographs. The typical observed form was that of an "N" wave with the main components in the 8 to 12 Hz zone (Fig. 1). The Sound Pressure Level was over 125 dBp in all measurements.

CONCLUSIONS

A muffling device based on the active and reactive principles has been designed with a common exhaust for the gases from the front and rear of the CSR type cannon.

To study the behavior of the projected structure we made scaled models and they have been tested at the Anechoic Chamber of CIAL.

We continue our research on a subject about which so little is known at present.

Fig. 1
INTRODUCTION

In applying admittance concepts in vibration problems, it is necessary either to calculate the required admittances, or to measure them. This paper presents formulas for calculating the drive-point and point-to-point admittances for an Euler beam.

APPROACH

The solutions for the vibration of two free, uncoupled, end-driven, long, thin, viscously damped beams of circular cross-section, $B_1$ and $B_2$, are combined to obtain the solution for the vibration of a free beam driven at an arbitrary point by an arbitrary force and moment. (See Figure 1.)

The solution is obtained by the principle of superposition noting that the applied force and moment are divided between the two sections, $B_1$ and $B_2$, of the composite beam and that both the deflection and slope of the beam must be continuous at $\xi_0$. The physics that must be recognized is that the application of a force, in general, generates a moment at the cross-section of the beam where the force is applied; the application of a moment, a force.

RESULTS AND CONCLUSIONS

Formulas are presented for calculating the drive-point and point-to-point admittances of the composite free beam in terms of the characteristics of the two component beams before connection.

The results are applied in calculating the vibration of a beam driven at one point when loaded at a second point by arbitrary admittance.

REFERENCE

Vibration - damping treatments and sandwich materials are used for reducing vibrations and noise, increasing mechanism and transport reliability. It is possible to control of dynamic characteristics of the constructions using these technique, and to reduce vibrations in points of their occurrences and along paths of their spreading.

Simultaneously the noise radiated by the construction, is decreased. When the output mechanical impedance of the vibration source exceeds the input impedance of the construction, it is necessary to reduce the input impedance of the construction. If the output resistance of the vibration source is lesser than the output impedance of the construction, the vibration-damping treatment should increase the dynamic rigidity and the construction mass. Thus, applying of the treatment can be used as a mean for changing the transfer function of the construction. Construction rigidity decreasing leads to the wave propagation velocity decreasing. This can improve the effect of vibration reduction.

Applying of the treatment enables us to change frequency characteristic of the radiation coefficient of the construction, shifting the frequency range near "wave coincidence", and to decrease the radiation in this frequency range. The treatment effects on the construction radiation at frequencies of "wave coincidence", due to the waveguide effect.

Control of dynamic characteristics of metallic latticed constructions is of great importance. The propagation of the vibrations in such constructions is connected closely with their vibration conduction.

Hydrofoil ship hulls are the examples of such type constructions ("Rocket", "Kometa" and so on). Vibration-damping treatments were applied for changing the dynamic characteristics of these ships. This allowed to reduce vibrations and noise by 10-15dBA, bringing them up to sanitary norms.

REFERENCE

To estimate and control the vibratory response of a complex structure one may need to understand the nature of the transmission of structureborne sound through the structural components that compose the structure. Large structures often incorporate components consisting of ribbed panels. The nature of the transmission of structureborne sound across ribs on a panel is, therefore, of interest. Basic to such a transmission is the transmission of sound in the panel across a single rib. Since there are occasions in which the panels are exposed to significant fluid loading, investigating the influence of fluid loading on the transmission is also of interest.

The transmission efficiency \( T_g \) of flexural waves on a panel across a rib was estimated (1). The panel was in the form of a membrane. The membrane was made to simulate the dispersive properties of a plate in that the tension in the membrane was assumed to be frequency dependent. A critical frequency \( \omega_c \) could thus be defined (1). The influence of fluid loading on the transmission efficiency was assessed and is typified graphically in Figure 1. Moreover, the modification in the transmission efficiency that can be achieved by introducing compliant coating was also assessed and is typified graphically in Figure 1. [The compliant coating is characterized by a resonance frequency \( \omega_r \) and a loss factor \( \eta_r \). The resonance is in reference to the compliance of the coating and the mass of the fluid loading.]

The transfer of vibratory energy across boundaries between adjacent structural components can be suitably estimated by means of the Statistical Energy Analysis (SEA) (2,3). Significant parameters of SEA are the coupling loss factors between structural components. A coupling loss factor is related simply to the transmission efficiency across the boundary between two structural components. Thus, the assessment of the influence of fluid loading and compliant coating on the transmission efficiency relates to the influence of fluid loading and compliant coating on the coupling loss factor. The implications of this influence in the case of energy transfer between two portions of a panel divided by a rib will be discussed.

**Fig. 1. Influence of Fluid Loading and Compliant Coating on the Transmission Efficiency of Flexural Waves Across a Rib.** Fluid loading parameter at the critical frequency is equal to ~ 0.1 and the transmission efficiency in the absence of fluid loading is equal to zero.

**REFERENCES:**
Contemporary vibration damping treatments and constructions could be successfully used for reducing the vibrations of engineering systems within limited temperature range. This range extension by means of known methods lead to the reduction of the maximum value of treatment effectiveness.

Author's investigation [1] made it possible to propose vibration damping treatments and constructions with effectiveness practically constant in any desired temperature range. In these treatments and constructions the thermoadjustable material (TAM) is used as a damping layer with temperature to be independable of ambient temperature.

TAM is developed on the basis of a material with the maximum effectiveness temperature $T_{max}$ below the upper limit of the working temperature range not more than 10-15°C.

When ambient temperature becomes below $T_{max}$, heating of TAM is performed. It is performed by means of:

a) electrical current flowing through TAM if it is electroconductive one, or through special electroheater (for instance carbonic textile) built in nonconductive TAM;

b) heat carrier (for instance exhaust gases, heat water etc.) circulating along special channels made in TAM.

The loss factor of a steel plate covered with unconstrained TRM treatment as a function of temperature is shown in fig. 1. Curves 1 and 2 correspond to the cases when TAM is heated or not heated respectively.

To reduce the heat energy loss is useful to place the layer of a light rigid nonheatconductive material (for instance cellplastik) between TAM and the structure damped.

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INTRODUCTION

It has been recognized for many years that military track laying vehicles have been a source of intense noise which adversely affects the operating personnel of these vehicles. These effects range from permanent and temporary hearing loss to degradation of intelligibility for both direct and electronically aided communications. Due to the seriousness of this noise problem the Army has embarked on a study in which the manufacturer of the M113 family of vehicles with the aid of a prominent acoustical consulting firm will reduce the noise of these vehicles. A goal of 100 dB(A), which requires a reduction of 15 dB, primarily at 250 Hz, has been set for this effort. The purpose of this investigation is to develop a thorough understanding of the noise sources, the acoustical and vibratory paths through which energy enters the hull structure, and the mechanism by which noise arrives at personnel locations.

METHOD

This paper discusses the initial portion of the study which consisted of a theoretical and experimental analysis of the primary noise source of the vehicle, i.e. the track and suspension system. This consisted of three phases:

1. The design of a computer program to simulate the track and suspension.
2. The isolation of the noise produced by the sprocket, idler and roadwheels in order to determine the contribution of each of these sources.
3. The measurement of vibration levels at the suspension system, and force-to-noise transfer functions for predicting interior noise levels.

This study indicates that at speeds above 5 mph the engine is not a major contributor to noise since towing the vehicle produced the same level as the self-powered vehicle; also, the roadwheels produce significantly less noise than the sprocket and idler since towing the vehicle with the track around the roadwheels alone produce noise levels about 10 dB lower than towing the vehicle in the normal configuration. Towing the vehicle without tracks further reduced noise by 10 dB indicating that the irregular top surface of the track is responsible for a significant portion of the noise. The isolation study indicated that the noise of the sprocket is about 5 dB greater than that of the roadwheels, and that the idler noise is about 5 dB greater than that of the sprocket.

The computer model of the suspension system as used with the force-to-noise transfer function provided reasonably accurate prediction of the interior noise of the vehicle. Further work will be accomplished, however, to refine this model.

RESULTS AND CONCLUSIONS

The results of the study indicate that the greatest potential for noise reduction lies in providing a softer compliance between the idler and the track. This should be followed by lowering the compliance of the sprocket and finally the roadwheels. The idler compliance can be lowered either at the hub or at the rim, however rim compliance poses fewer design problems.

In order to pursue the theoretical findings of this study, an experimental idler will be designed and tested to determine the actual noise reduction achievable when measuring, in isolation, the noise of a low compliance idler. In addition, the compliance and shape of the inner track will be modified to assess potential noise reduction.

Further analysis will include the determination of coupling efficiency between the suspension system and the hull and the potential noise reduction achievable by structural changes to the hull.
INTRODUCTION

Ground-borne noise and vibration originating at the wheel/rail interface has, for many decades, been a source of noise intrusion in buildings near underground rail transit facilities. With modern lightweight vehicles and continuous welded rail the vibration is seldom of sufficient amplitude to be felt, but is heard as low frequency rumbling noise. The vibration isolated or "floating slab" trackbed provides for effective reduction of this noise to lessen intrusion and allow placement of rail transit tunnels in closer proximity to buildings. A number of types of vibration isolated trackbed have been devised - from heavy bridge-like structures with damping and thick rubber support pads to relatively lightweight concrete slabs on resilient pads and without damping. This paper presents the basic design parameters for lightweight floating slab trackbeds and the results from one installation.

DESIGN

The lightweight floating slab design is based on the simple single-degree-of-freedom mass on spring vibration isolator concept with appropriate limitations on permissible rail (slab) deflection and with resonance frequency low enough to give reduction of transmissibility in the low frequency audible range. To control slab motion and give significant reduction of ground-borne noise the slab mass is made at least equivalent to the train mass and at least three times the bogie unsprung mass distributed over the length of the vehicles.

With loaded vertical resonance $f_{nv}$ of 15 Hz, maximum permissible rail deflection of 3 mm and appropriate mass, the design consists of concrete slabs 375 to 300 mm thick and 3.0 to 3.5 meters wide supported on 75 mm thick elastomeric pads. Since the slabs must be completely isolated from the subway structure, lateral and longitudinal supports are also elastomer pads. Lateral stiffness is designed for lateral natural frequency $f_{nl}$ to be less than the vertical resonance and such that $f_{nv} > f_{nl} > f_{ly}$. The dynamic vertical stiffness of vertical and lateral support pads and any entrained air must all be included in design calculations.

RESULTS

Trains have been operating on continuous floating slabs at the Washington, D.C. Metropolitan Area Transit Authority Metro facilities since early 1975. The slabs are 380 mm thick, 3.4 m wide reinforced concrete without added damping and have a loaded vertical fundamental resonance of 15 Hz. Rail fixation is via resilient rail fasteners as used with rigid concrete trackbed.

To determine the effectiveness, comparative measurements were made with trains running in subway structures of similar configuration and depth with and without floating slab. The reduction of ground-borne noise achieved is shown by Figure 1. The result is 10 to 15 dB reduction of the low frequency rumbling noise.

Fig. 1 Insertion Loss Performance of the Floating Slab Trackbed as Measured with 2-car Trains - Average of 32, 48 and 64 km/hr tests.
INTRODUCTION

Many methods used in the stone processing industry cause hearing damage and inconvenient vibrations. In this investigation the following types of machines have been studied in detail.

- Rock drilling machines (for quarrying)
- Stone saws (for processing of the stone blocks)
- Carving machines (for engraving on stone)

Sound levels of 95-117 dB(A) have been registered near the machine operator's ears when drilling. When using stone saws, the corresponding sound levels are 94-105 dB(A) and when working with carving machines the sound levels are 100-105 dB(A).

METHODS TO REDUCE NOISE AND VIBRATIONS

The noise disturbances from the rock drilling machines have been reduced considerably by using exhaust silencers which also screen part of the sound radiation from the machine. The vibration velocity of the handles of the carving machines has been reduced to a great extent by using vibration isolation. The handle design includes rubber isolators in a rocker arm construction with the isolators working in shear.

The noise from stone saws with circular diamond saw blades has been reduced considerably mainly by acoustic optimizing of parameter data as circumferential velocity, table feed and cutting depth and also by using vibration damped blades. The damped diamond saw blades have a patented design based upon increased losses in the web of the blade.

The noise reduction measures concerning carving machines have among other things included exhaust silencing and covering of the machine and the tools with rubber. The result of these measures have been comparatively good. The rather inconvenient hand vibrations which arise when working with stone have been reduced by an enclosing structure which is vibration isolated from the machine.

RESULT

The noise reduction measures on rock drilling machines mentioned above have reduced sound levels near the machine operator's ears to 91.5-102.5 dB(A). The major noise reduction, approx. 10 dB(A), has been reached on the less powerful machines.

The tested vibration isolated handles used on rock drilling machines have reduced the vibration velocity level 10-15 dB at the beat frequency which is the most annoying frequency.

On stone saws the sound level near the machine operator's place has been reduced by approx. 10 dB(A) by using suitable parameter data and damped diamond saw blades. The major noise reduction has been achieved in the highest octave bands, i.e. 4 and 8 kHz, where the reduction is 6-13 dB. The parameter data chosen for minimized sound radiation have also involved an increased cutting per time unit.

The noise reduction measures on carving machines have reduced the sound level near the operator's ears to 87-96 dB(A). By using vibration isolated machines the annoying hand vibrations have been reduced with at least 10 dB at the beat frequency.

CONCLUSION

This investigation shows that fairly moderate measures on the machines mentioned can reduce the sound level with approx. 10 dB(A) and also reduce the vibration level with approx. 10 dB which must be considered as satisfying.
IMAGING AND DIAGNOSIS OF MACHINE SYSTEM USING BISPECTRAL ANALYSIS OF NOISE

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New techniques, such as the interpolation of holograms, the precise estimation of holographic information by using multiple combinations of frequency components from fairly short data, and the new definition and estimation method of nonstationary processes are introduced.

These methods contribute to reduce the time required for the imaging and as the result even nonstationary states of the machine system can be imaged.

A complete system is constructed. It consists of the scanning mechanism covering 1.5m x 1.5m, the mini-computer arranged to get bispectra and to reconstruct the image, and the color display. The schematic construction is shown in Fig. 1.

This system can be used to get detailed data required for the design of more quiet machine systems or to diagnose machine systems which are under the transition from the normal state to the abnormal one.

As an example of the diagnosis of machine element by bispectral analysis of the noise, the change of the condition of the gear surface under over the loaded operations was estimated from the change of the bispectral characteristics of the detected noise. The results show the detectability of the growth of scorings on the surface without any physical contact to the machine, while conventional power spectral analysis fails to detect it.

Images of the noise distributions on the machine surfaces were also obtained under several different operating conditions.

Fig. 1. Construction of the System
LE PROBLEME DE LA REDUCTION DU BRUIT DANS LES TISSAGES

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INTRODUCTION

En présent la plupart des tissages sont équipés avec des métiers à tisser à navette, qui pendant le fonctionnement prussent un bruit de grande intensité qui dépasse la limite admissible (90dB). Ce bruit a des effets extrêmement nocifs sur l'organisme humain, qui conduit à un abaissement de la productivité du travail.

SOLUTIONS

a) Dans les tissages actuels équipés avec métiers à navette, pour réduire le bruit à la source, on a remplacé le système d'insertion de la trame à l'aide de la navette avec un autre système à l'aide de projectiles à renforcement.

b) À l'avenir, dans les nouveaux tissages on préconise le remplacement des métiers à tisser, où l'insertion de la trame se fait par navette, avec des machines à tisser ayant d'autres systèmes d'insertion (par projectile, avec crochets rigides ou flexibles, à jet d'air ou hydraulique et à foule ondulée).

Par la dotation des tissages avec des machines à tisser à jet hydraulique et foule ondulée le niveau du bruit dans ces tissages peut être abaissé sous la limite admise.

Dans les tissages équipés avec les autres types de machines à tisser et métiers à tisser ayant remplacé le système d'insertion de la trame, on obtient une réduction considérable du bruit, mais il dépasse quand même la limite admise. Si on applique un traitement phonocapsant au plafond, le niveau du bruit sera abaissé jusqu'à la limite admise.

CONCLUSIONS

Par l'application d'autres systèmes d'insertion de la trame ainsi que par le remplacement des métiers à tisser on obtient une réduction considérable du bruit mais on n'atteint pas la limite admise qu'en appliquant des traitements phonocapsants au plafonds. Dans les tissages dotés avec des machines à tisser à jet hydrauliques et foule ondulée, le niveau du bruit se trouve sous la limite admise.
It was shown as early as 1965 that one of the major causes of urban traffic noise is that due to lorries (1). More recently an extensive social survey in Britain (2) indicated that under urban conditions the proportion of heavy lorries in a traffic stream is the dominant parameter in determining noise nuisance, and can in fact be used to predict this. Lorries can have a detrimental effect on the environment, not only when moving in a general traffic stream on a busy urban road but also as individual vehicles being driven down quiet residential roads, particularly at night. In order to ameliorate the problems of lorry noise, a combination of two procedures is required. Firstly, the noise of individual vehicles must be reduced as far as possible at source and, secondly, lorry traffic must as far as possible be segregated from noise sensitive areas. In the United Kingdom, the noise emission of vehicles is controlled both for newly constructed vehicles and for vehicles in use on the road (3). The latter in particular is very difficult to enforce and as limits are set 3dB(A) above those for the construction test, the Regulation has little effect. It is, however, of particular importance to have a satisfactory test which can be carried out on vehicles in normal use on the road as vehicle noise can increase due to worn engines, defective exhausts, and body rattle. A screening test is therefore required by which vehicles can be referred to a testing station. A further very important defect in existing vehicle noise regulations is their dependence on A-weighted measurements, as these do not take account of annoyance likely to be caused to people inside dwellings. Because of the relatively poor attenuation of low frequency noise by windows, and the possibility of resonances being set up in rooms, lorries giving an acceptable level of noise when measured solely in dB(A) can nevertheless give a disturbingly high level of noise indoors. It would therefore seem reasonable to suggest that noise limits for lorries should be specified not only in dB(A) but also in a form to take account of low frequency noise emission.

Although it can be expected that, following successful research work on quieter engines (4), lorry noise will be appreciably reduced in the future, this cannot by itself solve the problem of lorry noise in towns. In considering areas as a whole, considerable environmental benefit can be obtained by the concentration of heavy lorry traffic onto suitable road. This can be accomplished either by the use of specified lorry routes or lorry bans for residential areas on a 24 hour or night-time only basis. The increases of traffic on already busy roads would lead to only marginal increases in noise levels, while giving substantial benefits to roads freed from lorry traffic, particularly at night. The diversion of individual lorries from residential roads gives even greater environmental benefits, with no perceptible disbenefits. An equitable compensation and insulation scheme for people affected by lorry routes would nevertheless be required.

This paper is presented by permission of the Scientific Adviser to the GLC but the views expressed are those of the author and not necessarily those of the Council.

REFERENCES

2. F.J. Langdon J.S.V, 1976 47(2) 265-282
1. Einleitung

Für die Lärmbekämpfung an komplizierten Maschinen und Aggregaten ist es erforderlich, die beteiligten Mechanismen der Schallentstehung und -abstrahlung getrennt zu erfassen und durch gezielte Maßnahmen zu beeinflussen. Ein wesentliches Hilfsmittel ist hierbei der kraftbezogene Schalleistungspegel.

2. Physikalische Grundlagen

Der kraftbezogene Schalleistungspegel $L_{pp}$ ist eine frequenzabhängige Übertragungskenngröße des zu untersuchenden Maschinensystems. Sie gibt an, welchen Schalleistungspegel $L_p$ die Maschine im ausgeschalteten Zustand abstrahlt, wenn sie am Ort der Schallentstehung künstlich durch eine Wechselkraft $F$ periodisch oder stochastisch angeregt wird:

$$L_{pp} = L_p - 20 \times \lg \left( \frac{F}{F_0} \right) \text{dB} \quad \text{(1)}$$

Mit $L_p = 10 \times \lg \left( \frac{P_{ak}}{P_0} \right) \text{dB}$ Schalleistungspegel

$F$ Effektivwert der Wechselkraft

$F_0 = 1 \text{ N}$ Bezugswert der Kraft

Als Anregungssignal hat sich Schmalband- oder Terzrauschen veränderlicher Mittenfrequenz bewährt. Durch Anregung unterschiedlicher Punkte und schrittweise Veränderung des Prüfobjektes lassen sich die für die Schallemission verantwortlichen Einflussgrößen übersichtlich erfassen, eine wesentliche Hilfe bei der Optimierung konstruktiver und funktioneller Maßnahmen der primären Lärmbekämpfung.

3. Meßeinrichtung

Die Meßeinrichtung des kraftbezogenen Schalleistungspegels besteht aus
- einem sendeseitigen Teil zur Erzeugung der Wechselkraft (Generator mit Regelverstärker, Leistungsverstärker, Schwingungserreger, Kraftmeßelement, Schwingungsmeßgerät) und
- der üblichen empfangsseitigen Meßkette zur standardgerechten Ermittlung des Schalleistungspegels.


Die grundlegenden Untersuchungen führte das Zentralinstitut für Arbeitsschutz, Leitstelle für Lärmschutz und Schwingungsabwehr, Dresden, DDR, in enger Zusammenarbeit mit dem Meßgerätehersteller durch.

Literatur:
INFRASONIC NOISE COMPONENT ON BOARD SHIP

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INTRODUCTION

In addition to noise and vibration as unwanted but unavoidable phenomena resulting from ship's machinery operation, there are also high levels of infrasonic noise. Whereas the allowable limits for noise and vibration are regulated, the phenomenon of the infrasonic noise for the time being is still not thoroughly analyzed including its effect on men; therefore, no criteria for its allowable limits exist.

MEASUREMENTS

Measurements on board show the presence of high sound pressure levels within very low frequency and infrasonic range. The levels and spectrum content depend upon ship size and structure, power of main and auxiliary engines, etc. Fig. 1 shows the noise spectrum recorded in engine room of a ship. Dashed line means the infrasound ear sensitivity threshold (1,2).

CONCLUSION

The measurement results on board prove the high level infrasonic noise. This noise component appears and effects the men simultaneously with audible noise and vibration. Accordingly, in addition to standard measurement of noise and vibration, the infrasonic noise component should be measured and combined effect of all these components on men investigated, in order to determine the complete limiting criteria.

REFERENCES

(1) B. Somek, Some problems of infrasound transmission and effect on men, Doctoral thesis, University of Zagreb, 1972
(2) W. Tempest, M.E. Bryan, Appl.Acoustics, (1972)5, 133
The control of environmental noise is the subject of considerable public and government concern throughout the world. Control techniques, however, often involve heavy expenditure and so in the present economic climate much interest is being centred upon measures which, in addition to being effective, are 'low-cost'. Such techniques, by stimulating privately financed measures, and by assisting and supporting voluntary efforts, may then be used to defer heavier costs, and spread investment over a longer period.

The purpose of the present study is threefold. Firstly to review possible low-cost techniques for the control and abatement of environmental noise. Secondly to assess the effectiveness of these techniques, and determine their implications in terms of manpower and budgetary requirements. Thirdly to highlight the techniques which require further development and suggest new measures.

For the study, the possible control techniques were conveniently categorised under four headings: legislation enforcement, education, product labelling, and co-ordination and planning.

(i) Legislation Enforcement: Legal sanctions to control and abate noise nuisance have been enacted in most countries, however their most remarkable feature is that they generally have not worked. One reason for their failure concerns the level of enforcement, for, like other environmental laws, noise control measures have been greatly under-enforced. Therefore one option open to governments is to increase the level of effective enforcement. The success of such a step will clearly vary from country to country, but in addition to this, the success will also depend upon a number of basic decisions regarding noise descriptions, the noise limits imposed on various sources, and so on.

(ii) Education: A great deal of community noise nuisance is avoidable and unnecessary for it stems from inconsiderate and often irresponsible behaviour by citizens. In this respect, education techniques can be used to increase awareness of noise as an environmental problem and a risk to health and welfare, to stimulate involvement in noise control, and to inform the public of noise control regulations and complaint procedures. Indeed, awareness of noise as a risk, and the desire for its abatement should extend to all levels: government, industry, and the individual. This degree of awareness can only be generated through large-scale dissemination of information on the subject.

(iii) Product Labelling: The concept of product labelling is to label manufactured products as to their noise emission characteristics, or their effectiveness at reducing noise. The requirement of accurate emission labels for products should influence consumer choice, assuming a market for quieter equipment and appliances, and thereby influence manufacturers' policy by creating healthy incentives for quieter technology. In this way product labelling could be an alternative or supplement to a comprehensive system of noise emission standards.

(iv) Co-ordination and Planning: Co-ordination and planning are the keys to noise control, both in the long-term and in the short-term. Effective use of resources, which is imperative for low-cost measures can only be achieved through cooperation between administrations concerned with all aspects of noise control. Furthermore, planning, through the planned use of resources, and by avoiding noise-disturbance creating situations can result in remarkable savings, in both time and money. The three main noise problems encountered by co-ordinating and planning bodies, that is, noise from road traffic, from airports, and industrial installations, are considered. Examples of the use of the above techniques will be described, and their success and implications assessed. Problems associated with their use will be discussed, and recommendations will be made both as to their improvement, and their application in other countries.
INTRODUCTION

The measurement code describing the required physical measurements is key to the success of a product noise control program. The measurement code should be either written into a regulation or specification, or reference should be made to the appropriate international or national standards. Detailed information on methods of measurement of sound pressure levels is given in American National Standard S1.13-1971. For characterizing the noise emissions of stationary sources, the A-weighted sound power level is the preferred quantity as described in American National Standard S1.23-1976. Alternative methods for determining the sound power levels of stationary noise sources, both A-weighted and in frequency bands, are described in a new series of international standards which is in the process of being issued by the International Organization for Standardization (ISO).

MEASUREMENT CODES

A check-off list (in the form of questions) has been prepared to assist in evaluating the adequacy of a measurement code (1). An abbreviated version of this check-off list is given in Table I.

TABLE I

CHECK-OFF LIST FOR MEASUREMENT CODES

1. ACOUSTICAL QUANTITY. What is the acoustical quantity to be determined?
2. NATURE OF SOURCE AND NOISE. What kinds of sources and noise emissions are covered by the measurement code?
3. ENVIRONMENT. What acoustic environment is prescribed for the measurements?
4. INSTALLATION. Where will the source be installed and mounted during its acoustical evaluation?
5. OPERATION. How will the source be operated during its acoustical evaluation?
6. INSTRUMENTS. What test equipment is to be used for the acoustical evaluation?
7. CALCULATIONS. Are the calculation procedures described in detail?
8. DATA. What data are to be recorded?
9. REPORT. What data are to be reported?

CONCLUSION

Those test codes and specifications that provide satisfactory answers to the questions of Table I have the best chance of exerting a positive influence on the reduction and control of the levels of product noise. The importance of an unambiguous, clearly-presented measurement code cannot be overemphasized.

REFERENCE

(1) W.W. Lang, Proceedings of INTER-NOISE 77 (ETH, 8006 Zurich, Schweiz)
The aim of the present research is to form a basis to the complete understanding of the noise generated by non-free flowing traffic under any conditions. Present research has been limited to vehicles accelerating from a stationary condition at a controlled intersection under the action of the green phase of the traffic signals, the vehicles progressing forward across the junction, and to vehicles decelerating from free-flowing to a standstill when presented with the red phase of a set of traffic signals. To comply with existing prediction methods for the noise generated by road traffic, vehicles have been grouped into two categories, light and heavy.

A predictive method has been proposed to determine the noise emitted by an accelerating queue of vehicles based on performance characteristics of the average heavy vehicle (7.8 litre diesel engine) and the average light vehicle (1.6 litre petrol engine), the power unit being the major source of noise and being treated as a point source, with the inclusion of rolling noise at higher vehicle speeds. Using measured delay times for the commencement of each vehicle motion in a queue, the noise received at any location around an intersection can be determined for a queue of accelerating vehicles. The validity of this prediction method was tested during the present research by comparison of measured noise patterns with those predicted for queues of given vehicle composition accelerating at a controlled intersection, giving a pooled estimate of the correlation coefficient between measured and predicted results of 0.91.

The manner in which vehicles decelerate to a standstill when presented with the red phase of a set of traffic signals was measured by timing the progress of individual vehicles at a controlled intersection, deceleration patterns for the average light and heavy vehicle then being obtained. Using this information and by direct measurement of the noise generated by single vehicles when decelerating to a standstill from a free-flowing condition, the results being grouped into the two categories of vehicle type, noise patterns were obtained for the average light and heavy vehicle decelerating at a controlled intersection. By using these results it would then be possible to predict the noise received at any position from a queue of traffic of given composition decelerating at a controlled intersection.
The Exposure to Noise at Homes in England

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In 1972 a survey of the impact of roads and traffic on 7200 randomly chosen homes was undertaken by SCPR Social and Community Planning Research and OPCS Office of Population Censuses and Surveys on behalf of the Department (1). This was to be supplemented by objective measurements of noise and traffic at 1200 of the homes in the sample. These measurements were attempted during September and October of 1972 and 1973 and measurements were achieved at the facades of 1040 of the 1200 sample. (For the remainder, the measuring teams were not able to get permission from the occupant to make a facade measurement or the home could not be traced or had been demolished). The measurements are considered representative of the conditions experienced by the UK population at their homes.

Fig. 1 indicates the exposures to average hourly Leq levels calculated for 24 hours, the period 0600 to 2400 hours (day) and the period 2230 to 0730 hours (night). It can be seen that 12% per cent of the population are exposed to average day-time levels of not more than 47 dBA while towards the other extreme 11.3% per cent are exposed to average day-time levels higher than 65 dBA. Night-time levels are from 10 dBA (in quiet areas) to 5 dBA (in busy areas) below day-time levels.

An analysis by population density revealed that outside the conurbations exposure was weakly dependent on population density. The table shows the observed differences at the median level.

<table>
<thead>
<tr>
<th>Population Group (people per ha)</th>
<th>Median level of average Leq 0600 to 2400 hours</th>
</tr>
</thead>
<tbody>
<tr>
<td>25 to 96</td>
<td>57</td>
</tr>
<tr>
<td>1.5 to 24</td>
<td>54</td>
</tr>
<tr>
<td>less than 1.5</td>
<td>50</td>
</tr>
</tbody>
</table>

London was on average more than 5 dB(A) noisier than the other conurbations. Within the population groups sound levels varied with the logarithm of traffic flow.

REFERENCE
In a pilot survey of aircraft noise nuisance near London (Heathrow) Airport, 600 residents were each asked 45 questions during a 30-minute interview. The questionnaire was designed to compare various techniques for assessing the magnitude of the disturbance. To test the hypothesis that the direct disturbing or intruding effects of the noise would be more highly-correlated with noise exposure than would the annoyance indirectly caused by these effects, the questions were arranged into two categories. The direct effects were measured through questions concerning the frequency and duration of disturbance whilst the indirect effects were scaled through questions related to annoyance, bother and monetary costs.

In the event, this hypothesis could not be verified; the correlation between disturbance and noise was not significantly different from that between noise and annoyance. However, whether or not the hypothesis is incorrect, or whether the measurements of the two effects were inadequate could not be established. Certainly the correlation between measurements of disturbance and annoyance was sufficiently low to indicate that different responses were indeed being measured.

The relationship between annoyance reactions and noise exposure (measured in NNI) agreed closely with those found in earlier surveys. Measurements of disturbance frequency were equally well correlated with noise and offer a viable alternative or complement to annoyance measurements as well as a meaningful response scale which can be more precisely related to specific events or time periods.

Little evidence was found that people think about noise nuisance in monetary terms either spontaneously or after persuasion to do so. Only 1% of respondents spontaneously suggested that monetary compensation might provide a solution to the aircraft noise problem and few respondents could be persuaded to suggest a suitable payment.

Of numerous noise exposure scales investigated, none was found to offer any practical advantages over NNI as currently used in the U.K. for aircraft noise assessment. It is convenient to measure and predict and it exhibits what might be considered an ideal relationship to annoyance. Of potential alternatives, only equivalent continuous sound level (Leq) exhibited similar favourable relationships with community response.

In order to assess the magnitude of public concern about noise, respondents were questioned about eighteen other possible sources of dissatisfaction. Of all items, noise ranked second to road safety and was followed by atmospheric pollution. Predominant sources of noise nuisance were, in rank order, aircraft, road traffic, children and neighbours. Surprisingly, although half the respondents lived within half a mile of a railway, trains were less bothersome than any other nuisance, acoustic or otherwise.
THE EFFECT OF INTERIOR NOISE OF ROAD VEHICLES ON THE DETECTION OF EMERGENCY VEHICLE SIRENS

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INTRODUCTION

The driver of a motor vehicle in motion is subjected to a variety of noises from his own and other vehicles; these noises can mask the sound of an emergency vehicle siren. The published data on vehicle interior sound levels concentrates on new automobiles travelling at steady speeds with the windows up and the radio off. To examine the interior noise of older cars, a truck, a bus, and an ambulance, and to determine the effect of speed, open windows, and an operating radio on the interior sound levels, measurements were made using a tape recorder and an automatic 1/3-octave band analysis system. The results of these measurements were used to prepare a set of typical-situation design curves for a detailed analysis of siren detectability, reported elsewhere.

RESULTS

The measured interior sound levels were generally higher for older vehicles, when the influence of mechanical noise was significant; sound levels of 84 dBA were observed inside a small American car at 55 mph (90 kph) with a gearbox whine. The condition of the vehicle was also significant in determining the increase in interior sound level with speed. Again, the result depended on the significance of the mechanical noise compared to the tire and aerodynamic noise at different speeds. Opening the windows produced an increase in sound level that was greatest for the higher frequencies. Operating the radio produced an increase in interior sound level, with the greatest effect in the 1000 to 2000 Hz frequency range.

DISCUSSION

The results show that the high levels of interior noise of road vehicles will mask the sounds of emergency vehicle sirens which are typically in the 500 to 2000 Hz frequency range. This is particularly so for the high-speed radio-on case, where the sounds of the radio coincide in frequency with the sounds of the siren. The detection distances that were calculated ranged from 440 ft (133 m) for a slow-speed window-open, no-radio, head-on example to 3 ft (1 m) for a high-speed, window-closed, radio-operating, crossroads example. It was concluded that the drivers of emergency vehicles should not rely on their audible warning devices (sirens) alone to ensure detectability by other road users.

(This work was supported by the Society of Automotive Engineers.)
INTRODUCTION

Traditionally, rapid transit system stations, particularly underground stations, have been highly reverberant, noisy spaces where the patrons are subjected to very intense noise from transit train operations. The application of acoustical treatment to the interior surfaces of transit stations and to the underplatform area adjacent to the transit cars makes it possible to substantially reduce the noise due to all sources in the transit stations and particularly to reduce the noise due to transit train operations in underground stations.

The application of acoustical treatment in the design of a transit system station accomplishes four major purposes:

1. Control and reduction of noise from transit train operations
2. Provision for good intelligibility of announcements from the public address system
3. Control of general crowd noise generated by patrons talking and walking, and
4. Assistance in the control of noise from the station air conditioning system and other mechanical equipment

For modern transit system subway stations the use of sound absorption material installed on the underplatform area, the train room walls and ceilings, and the ceilings and walls of mezzanine or concourse areas is required for control of noise and reverberation in the station. Similarly, enclosed areas of above-grade stations will have ceiling and, possibly, wall mounted absorption materials. These design features are essential in order to provide a satisfactory and attractive acoustical environment for the transit system patrons.

The most flexible and probably the most economical material which can be used for station acoustical treatment is glass wool material either in flexible, semi-rigid, or rigid board form. In order to have sufficient absorption coefficient in the low frequency ranges (125 to 250 Hz), it is essential that 2" or 3" thick glass wool treatment be used. Facings of perforated metal or metal slit-and-slat are used for both protective and architectural purposes.

RESULTS

The San Francisco Bay Area Rapid Transit System (BART) and the Washington, D.C. Metro System (WMATA) subway stations all have extensive application of sound absorbing material on the ceilings and in the underplatform spaces. The result is much more acceptable noise levels from the transit train operations than found in older systems which have completely untreated highly reverberant stations. The reverberation time measured in treated BART and WMATA stations are typically 1.3-1.5 seconds at 500 Hz, as compared to 7-9 seconds for untreated stations. For trains passing by at 60-65 km/hr, the noise levels measured at BART and WMATA platforms are in the range of 87-89 dBA. Similar measurements made at untreated stations indicate train noise levels of 100-108 dBA on the platform.

Measurements at two BART stations also indicated that the average train operation noise level is 5 dBA less when the station is treated with underplatform acoustical treatment. The Toronto Transit Commission (TTC) has had good success with acoustical treatment and at one station where a suspended ceiling was installed on the entire ceiling and sound absorption was placed on the underplatform space, the noise reduction obtained was about 13 dBA average compared to the untreated station, a substantial improvement.
Repeatability of measurements is of major importance for administration of noise laws and regulations, particularly when situations under investigation differ marginally from critical values specified therein.

A series of studies have been done to determine the effect on repeatability of various instruments, operational techniques and test environments in the fields of vehicle and community noise control.

**VEHICLE NOISE**

The first study examined the ability of trained and semi-trained observers to read the pointer deflections of two makes of precision sound level meter using recordings of vehicle noise signatures. Sub-studies covered the performance of various automatic level recording systems and comparison between field measurements and recordings made simultaneously through a common microphone system.

Appropriate precautions were taken to avoid subjective identification of sources and to cover various segments of meter scales and these will be outlined in the main paper but briefly the results showed that:

(a) Trained observers reported their results in 0.5dB intervals and replicated very well at this resolution.

(b) Semi-trained observers reported results to 0.1dB resolution and replicated reasonably well.

(c) The "impulse-hold" mode of sound level meter operation could be read to high resolution with good replication and was better than either graphic level recorder or real-time analyser.

(d) There was a very small systematic difference between observer judgements of meter scales which were linear or logarithmic.

(e) The field measurements agreed substantially with trained observer interpretations of the recordings.

Further work on the vehicle noise field will investigate the possible advantages of using directional versus omni-directional microphones for on-street measurements.

**COMMUNITY NOISE**

The community noise studies involve comparison of a number of commercially available instruments using recordings of typical city noise situations, the variation in noise levels from site-to-site and also over successive intervals ranging from seconds up to weeks and correlation with traffic distributions. These studies have been made possible by development of an instrument which samples noise at millisecond intervals, integrates these over a wide range of periods and reports the resultant distribution of levels every five minutes for more than two weeks while unattended in the field.

Again the test procedures and results will be given in more detail in the full paper but briefly they show that:

(a) A minimum range of 70dBA is needed to cover typical distributions of noise at city and suburban sites.

(b) The lower limit is set by system electrical noise.

(c) The more processing of results within an instrument the greater the chance of "garbage" combined with less ability to detect it.

(d) No pattern has yet emerged for $L_{eq}, L_1, L_{10}, L_{50}, L_{90}, L_{99}$ variations over the intervals so far investigated which implies that short-term sampling to determine any of the above measures is fraught with problems.

(e) Mathematical models relating vehicle movements to noise exposure so far tested have not correlated well over a wide range of sites.
EXISTENCE D'UN INDICE DE CRÊTE

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INTRODUCTION

La gêne de jour due au bruit engendré par le trafic automobile est assez bien décrite par son niveau équivalent Leq exprimé en dBA. Il semble par contre que pendant la période de repos du soir et la période de sommeil les populations soient plus dérangées par les crêtes de bruit.

BASE THÉORIQUE

Le bruit de crête sera mesuré par un indice statistique Ln, niveau atteint ou dépassé pendant n % du temps.

PROCESSUS EXPERIMENTAL

L'étude présentée décrit parmi la série des indices statistiques un certain nombre d'indices de crête depuis les plus pointus (L0,1) jusqu'à ceux qui prennent en compte un pourcentage de temps plus long (L5). Cette étude est faite à partir de bruits provenant de différents types de trafic de nuit :

- 50 véhicules/heure dans deux rues de LYON
- 500 véhicules/heure (dont 40 % de poids lourds) sur la Nationale 6 en traversée de MACON
- 90 et 140 véhicules/heure sur voie rapide urbaine lyonnaise.

Ces indices de crête sont étudiés sous l'angle de leur stabilité dans le temps sur une période de 1 heure

RESULTATS ET CONCLUSIONS

L'étude montre d'une part que les indices de crête trop pointus sont trop sensibles au véhicule le plus bruyant; que d'autre part les indices d'ordre trop élevé (> à 1 ou 2) deviennent trop sensibles aux faibles variations de trafic nocturne.

Au terme de cette étude il semble donc que L1 requiert les qualités d'un bon indicateur de bruit concernant la période nocturne. Il ressort également qu'une période de mesure de 15 à 20 minutes suffit à la détermination de cet indice.
THE USE OF THE A-WEIGHTING SCALE IN INDOOR NOISE CONTROL PROBLEMS

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INTRODUCTION

The A-weighted scale is already internationally standardized for noise measurements dealing with the evaluation of nuisance from noise exposition.

Owing to the simplicity with which one may perform this kind of measurement, it would be interesting to know if one may so evaluate not only sound fields, but also the effects of absorbing and insulating materials on them.

THEORETICAL BACKGROUND

Recent researches (1,2) have attempted to define limits of validity of a measuring technique of sound insulation that uses A-weighted scale.

If we accept the well-known Sabine formula for the reverberation time, we have:

$$L_{10f}/10 - W_{A_f}/10$$

$$\Delta L(A) = 10 \log \left( \frac{\sum_{n_f} L_{10f}/10 - W_{A_f}/10}{\sum_{n_f} L_{10f}/10 - W_{A_f}/10} \right)$$

which analytically relates $\Delta L(A)$ not only to the characteristics of the acoustic materials fitted up and to the weighting curve, but also to the spectrum of the sound field to be controlled.

Having established the dimensions of a typical room (this is useful but not necessary) equation (1) was utilized to evaluate the decrease in the A-weighted sound level obtainable with the most common acoustic materials for different sound spectra of industrial noise.

CONCLUSIONS

Since, in the planning phase, an evaluation of the results obtainable with different applications of sound absorbing materials undertaken with precision greater than ±1 dB has little significance (owing to uncertainty in knowledge of materials' characteristics and in readings on measuring instruments), such calculations have allowed us to state the limits within which it is permitted to initially evaluate the degree of efficiency of planned treatments by means of the simplified equation:

$$\Delta L(A) = 10 \log \left( \frac{1}{n_f} \sum_{n_f} \frac{A_{2f}}{A_{1f}} \right)$$

REFERENCES

(1) D.H. Stephens - Applied Acoustics (1973) 6, 151; (1976) 9, 131
(2) J. Pujolle - Revue d'Acoustique, (1976) 35, 38

NOMENCLATURE

- $A_{1f}$ - absorbing units of the room before and after acoustic treatment, each evaluated for each frequency band utilized (1 octave, 1/3 octave, etc.);
- $A_{2f}$ - absorbing units of the room before and after acoustic treatment, each evaluated for each frequency band utilized;
- $W_{A_f}$ - A-weighting factor, evaluated for each frequency band utilized;
- $L_{10f}$ - sound pressure level before acoustic treatment, evaluated for each frequency band utilized;
- $\Delta L(A)$ - difference between the A-weighted sound level before and after treatment;
- $n_f$ - number of frequency band considered.
ESTIMATION OF LOUDNESS LEVEL BY WEIGHTING CURVES

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INTRODUCTION

The normal equal loudness contours for pure tones show how the physical sound pressure level should vary to create the same sensation of loudness at various frequencies. Taking this into consideration for estimation of loudness level in case of noises it is required to use a set of frequency weighting curves or several kinds of computing procedures. However, in some cases the results obtained by different methods can be considerably different.

In the framework of a Round Robin test we have determined the sound pressure levels for 20 types of different noises which were judged to be equally loud as a reference signal, i.e. 1000 Hz, 1/3 octave, 75 dB.

The measuring method used in these experiments was elaborated by ISO /1/.

RESULTS AND CONCLUSIONS

Results obtained from these experiments were already published /2/. This paper deals with the connections between the subjective judgements and the estimations obtained from different methods /dB(A), dB(D), phon(GF)/.

1. A better correlation was found with the subjective judgements when using the D-weighting curve instead of the A curve. The dB(A) values underestimate considerably and the dB(D) values give an acceptable estimation mainly for noises of high frequencies. It was assumed to be the reason of this difference that the D curve approximates to a greater extent to the correct frequency characteristics of the ear. The frequency characteristics of the ear - marked as "W" curve - can be plotted by using the normal equal loudness contours for pure tones. Comparing "W" curve to A and D curves, it appears that the A curve deviates considerably from "W" curve and the D curve shows resemblance to it only at high frequencies. By means of a computer the calculations were carried out using the "W" curve but the results achieved in this way were not correct enough either. This deviation can be explained by the fact that the "W" curve refers to pure tones only.

2. Knowing the equal loudness levels and the spectra of the examined noises, a computer program - successive approximation - was made to calculate the loudness levels for these noises with different weighting curves and to vary the weighting curves as long as the deviations reach a minimum in accordance with the subjective judgements. When the best approximation was achieved the maximum deviation was less than 3 dB. This curve was marked "R". See Table I.

Although these calculations concern only the examined 20 types of noise spectra and the reference level but the industrial noises and the noises in everyday life are more or less similar to some of the examined noises in spectral composition. Therefore the obtained "R" curve can be considered to be valid for the noises in practice, too.

Table I. Results for all of examined noises calculated by different methods.

<table>
<thead>
<tr>
<th>db(A)</th>
<th>db(D)</th>
<th>phon(GF)</th>
<th>W curve</th>
<th>R curve</th>
</tr>
</thead>
<tbody>
<tr>
<td>&quot;R&quot;</td>
<td>&quot;W&quot;</td>
<td>&quot;W&quot;</td>
<td>&quot;W&quot;</td>
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</tr>
<tr>
<td>dB</td>
<td>dB</td>
<td>phon(GF)</td>
<td>dB</td>
<td>dB</td>
</tr>
<tr>
<td>-19</td>
<td>-40</td>
<td>+8</td>
<td>-44.1</td>
<td>-3.6</td>
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<td>1.0</td>
</tr>
</tbody>
</table>

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EINE VEREINFACHTE METHODE ZUR SCHALLDÄMM-MASS BESTIMMUNGEN

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1. EINLEITUNG

Das gewöhnliche Verfahren zur Bestimmung Schalldämm-Mass R der Sperre ist nach der Gleichung:  
\[ R = L_1 - L_2 + 10 \log \frac{S_{12}}{A_{22}} \]  
(1)

wobei bedeuten:  
- \( L_1 \): Schallpegel im Senderaum,  
- \( L_2 \): Schallpegel im Empfangsraum,  
- \( S_{12} \): Fläche der Sperre,  
- \( A_{22} \): Äquivalente Schallabsorptionsfläche des Empfangsräumes.

In dieser Arbeit wird eine vereinfachte Methode beschrieben: es ist nicht notwendig Äquivalente Schallabsorptionsfläche zu messen.

2. THEORETISCHE GRUNDLAGE

Für zwei gekoppelte Räume, die auf der Abb.1 dargestellt sind, gelten folgende Gleichungen:

\[ cA_{11}E_1 - cS_{12}E_2 = 4P_1 \]  
(2)

\[ S_{12}E_1 - A_{22}E_2 = 0 \]  
(3)

dabei ist  
- \( P_1 \): Schalleistung der Quelle,  
- \( c = 300 \text{ m/s} \),  
- \( E_1 \): mittlere Energiedichte im Senderaum,  
- \( E_2 \): mittlere Energiedichte im Empfangsraum,  
- \( A_{11} \): gesamte Äquivalente Absorptionsfläche im Raum 1,  
- \( A_{22} \): Äquivalente Absorptionsfläche im Raum 2 ohne \( S_{12} \), ohne Öffnungsfäche.

Wenn die Öffnung zwischen den Räumen mit der Sperre geschlossen wird, Abb 2, Gleichungen (2) und (3) bekommen folgende Form:

\[ cA_{11}E_1 - \tau cS_{12}E_2 = 4P_1 \]  
(4)

\[ \tau S_{12}E_1 - A_{22}E_2 = 0 \]  
(5)

wo \( \tau \): Transmissionskoefizient ist. Aus der Gleichungen (2),(3),(4) und (5) wird:

\[ E_2/E_2 = (1/\tau)(A_{11}A_{22} - \tau \cdot S_{12}^2)/(A_{11}A_{22} - S_{12}^2) \]  
(6)
ausgerechnet.

3. MESSENGMETHODE

Wenn die Bedingung \( A_{11}A_{22} > S_{12}^2 \) (7)

ausgefüllt ist bekommt Gl(6) folgende Form:

\[ E_2/E_2 = 1/\tau \]  
(8)

Jetzt wird Schalldämm - Mass bestimmt als

\[ R = 10 \log 1/\tau = 10 \log E_2/E_2 = L_2 - L_2' \]  
(9)

4. ZUSAMMENFASSUNG

Also man soll nur Schallpegel im Empfangsraum in zwei Fällen messen: einmal wenn die Öffnung nicht zugesperrt ist und zweiten Mal wenn die Sperre an der Öffnung angestellt ist.

Diese Methode ist besonders zur Bestimmung der Schalldämm-Mass von Türen und Fenstern gunstig. Mit dieser Methode vermindert man die Zahl der Messungen (40 Messungen weniger) und die Apparatur zur Nachhallmessung ist nicht notwendig.

5. REFERENZEN

INTRODUCTION

An injection pump test machine is used to test diesel fuel pumps which have an impulsive torque characteristic. High levels of noise are produced which have been analysed and reduced, utilising the source-path-radiator concept.

DIAGNOSIS

The significant noise radiators were identified using progressive exposure from a shielded condition, a remotely-driven 'Quiet rig', the close microphone versus vibration velocity technique and modal analysis.

Primary noise radiators are machine surfaces below 2 kHz, and above 2 kHz, both machine and pump surfaces, with the table, bed and coupling predominant in the machine group, see Fig. 1.

SOLUTION

The path stiffness at B, the mounting bracket, could not be reduced without exceeding the allowable pump speed fluctuation. Modifying the path C-D by a direct earth or by introducing isolation were both considered, but rejected as uneconomic.

However a retrofit kit has been tested which eliminates the secondary source G, introduces isolation in path L-M (for CAV blocks only), and reduces radiation at D, E, K, L and M by close shielding. Overall noise reductions are of the order of 5-7 dB(A). Future work includes an investigation of the effect of partially earthing the bracket B.

REFERENCES

La prévision des niveaux sonores est très importante pour la lutte contre le bruit. Il a été souvent souligné que les calculs basés sur l'hypothèse du champ diffus conduisent à des résultats qui ne sont pas toujours vérifiés dans la pratique ; par contre, les calculs détaillés effectués sur ordinateur et basés sur les concepts d'acoustique géométrique (méthode des sources images, méthode des rayons) peuvent permettre une meilleure prévision.

Le but de la présente étude consiste à rassembler, grâce à des mesures effectuées in-situ, des informations concernant les caractéristiques de propagation d'un grand nombre de locaux industriels, à effectuer les calculs prévisionnels des niveaux sonores et à en confronter les résultats avec ceux qui ont été obtenus lors de la campagne de mesures.

Pour les études in-situ, nous avons utilisé une source sonore de référence de grande puissance \( L_w = 114 \text{ dB réf. } 10^{-12} \text{ W} \) et un dispositif enregistreur sur roues que l'on éloigne progressivement de la source ; ce dispositif permet d'enregistrer simultanément le niveau de bruit et la distance séparant le récepteur de la source. On obtient ainsi des "courbes de décroissance" qui permettent de caractériser, de façon adéquate, les locaux au point de vue acoustique. Cette étude a été effectuée pour plus d'une trentaine de locaux variés dont le volume est compris entre 1274 m\(^3\) et 129 000 m\(^3\).

Parallèlement à cette étude expérimentale, des calculs ont été effectués selon la méthode des sources images afin de prévoir les courbes de décroissance en fonction des dimensions des salles, des coefficients d'absorption estimés des parois et de l'humidité de l'air. Dans cette méthode de calcul, tout se passe comme si le niveau sonore en chaque point résultait de la combinaison de l'énergie sonore provenant de la source réelle et des images de la source qu'on verrait si les parois étaient remplacées par des miroirs. Dans nos calculs, nous avons tenu compte des 600 premières sources images (images du 1er au 7ème ordre) et des 100 plus importantes images allant du 8ème au 10ème ordre.

Les courbes de décroissance ainsi obtenues servent ensuite à l'établissement de cartes de bruit complètes qui tiennent compte de toutes les sources de bruit existant dans l'atelier.

La confrontation des résultats montre, par exemple, que dans les cas des courbes de décroissance, à 15 m de la source, la différence entre les résultats des calculs et ceux des mesures, est inférieure à 3 dB pour 27 sur 30 locaux ; elle est de 4 dB pour les 3 derniers locaux. Une concordance analogue est obtenue pour les cartes de bruit.

De façon générale, la concordance est moins bonne pour les locaux très encombrés avec une grande densité de machines ; ces dernières comme les matériaux peuvent avoir une influence sur la propagation du son en agissant comme absorbeurs, réflecteurs ou diffuseurs. De même, la précision des cartes de bruit dépend de la précision avec laquelle on connaît la puissance acoustique des machines et de la validité de l'approximation consistant à remplacer une machine par une source ponctuelle. Ces différentes questions seront abordées lors d'études ultérieures.

REFERENCES

(1) M. MOULIN, C.R., F.A.S.E., 1975, PARIS
INTRODUCTION

The validity of the 125 msec. time constant in the "fast" position for acoustics level meters and microprocessors used for Community Noise Evaluation, though used at present seems questionable in certain cases.

It is also very important to choose a suitable sampling technique. This aspect has in general not been taken into due consideration. The less gaussian is the noise and the bigger the standard deviation, the more significant the sampling intervals become.

EXPERIMENTAL PROCEDURE

Noise produced by an aircraft flyover or a vehicle passby can be approximated by a triangular pattern of level over time since it exhibits a gradual increase to its maximum level followed by a gradual decline.

Experiments were conducted to find out the significance of the sampling period and to get defined and repeatable signals. A series of triangular pattern signals with different amplitude and slope were recorded in magnetic tape to simulate the different vehicle passbys. These signals were computed statistically to get $L_{eq}$.

Two instruments were used: 1) A Statistical Analyzer with measurement intervals of 0.1 sec. and 1 sec.; 2) A modified 1945 Community Noise Analyzer.

The standard version of this analyzer has two integration time constants: "fast" position with 125 msec. and "slow" with 500 msec. with a maximum sampling time of 0.22 sec.

We had it modified at the factory so that the integration time constant was reduced to 50 msec. and a switch was added to triplicate the maximum sampling speed.

We chose as representative, a laboratory sampling time of 1 hour.

THEORETICAL BASIS

The equivalent continuous sound level $L_{eq}$ is supposed to represent the effects of noise on people since it integrates the absorbed acoustical energy. In practice:

$$L_{eq} = L_{max} - 10 \log 2.3 \Delta L/10$$

when $\Delta L = L_{max} - L_b > 10 \text{ dB}$. Where $L_b$ means background noise level.

This value was obtained from oscillographic data.

CONCLUSIONS

The statistical noise analyzer readings turned out not to be fast enough to reproduce the modelled event, whereas the microprocessor came closer to the theoretical value.

We plan to check this assumption by field experiments now in progress.
The problem of active absorption of arbitrary stationary sound field in waveguide was solved in work (1). Absorption is performed using monopole and dipole sources continuously distributed in waveguide cross-section. Source amplitudes are calculated from the indications of monopole and dipole receivers, distributed in another cross-section. Such method allows to achieve complete field absorption, however it is not realized in practice, because it is required infinite number of receivers and sources.

Another path to solve the problem of sound absorption in the waveguide is suggested in work (2). Sound wave energy is transferred by propagating normal waves. If sound frequency $\omega$ is limited ($\omega < \omega_k$), then the finite number $N$ of such propagating waves exists. As a rule, one can neglect exponentially vanishing normal waves of upper numbers. In this case the problem of absorption is solved using $2N$ sources and $2N$ receivers (in fact, the method of solution reduces "N-dimensional" problem to N "onedimensional" of type, described in works (3, 4).

Experimental system, developed on this principle, performed field absorption in the waveguide at frequencies, where two normal waves were propagated. We achieved field absorption near 15-20db in wide frequency range, independently from excitation method and relations between amplitudes of propagating normal waves.

REFERENCES


L'absorption acoustique active ou la réduction des ondes sonores par interférométrie présente des avantages pour les fréquences répétitives comprises entre 30 Hz et 1 kHz. Après des travaux [1, 2, 3] d'acoustique théorique et d'électroacoustique, préliminaires indispensables, il était opportun de faire appel aux théories et aux techniques nouvelles de la commande et du filtrage en vue du pilotage et de l'automatisation complète du dispositif anti-bruit.

**THEORIE**

Pour déterminer théoriquement un système d'asservissement physiquement réalisable, il est nécessaire de faire les hypothèses simplificatrices suivantes : onde sonore guidée par un conduit rectiligne et longueur minimale de la gamme de fréquences à réduire assez grande devant les dimensions latérales de la conduite (4 à 5 fois). La propagation devient alors un phénomène unidimensionnel auquel on peut appliquer la théorie des systèmes scalaires, linéaires et invariants.

Le problème se réduit à la recherche du filtre optimal $F(s)$ (Fig.1) qui minimise la pression acoustique $P_R(s)$ en aval de la zone à insonoriser, soit:

1. $E \left\{ \left[ P_R(t) \right]^2 \right\}$ minimum où $E$ représente l'espérance mathématique.

Soit encore :

2. $E \left\{ \left[ P_B(t) + \int_{-\infty}^{+\infty} W(\beta) \cdot P_A(t-\beta) \cdot dB \right] \cdot P_A(\varepsilon) \right\} = 0$

avec $W(s) = C(s) \cdot F(s) \cdot A(s) \cdot T(s)$ et $w(t)$ sa réponse impulsionnelle.

On obtient :

3. $C_{BP}^P A(t) + \int_{-\infty}^{+\infty} W(\alpha) \cdot C_{PA}^P A(t-\alpha) \cdot d\alpha = 0 \quad \forall t > 0$

où les $C_{ij}$ représentent des fonctions de corrélation.

L'équation (3) se résout par la méthode classique de Wiener-Hopf qui donne la transmittance $W(s)$ physiquement réalisable.

**CONCLUSIONS ET PERSPECTIVES**

L'application de ces résultats théoriques à un montage expérimental simple (gaine de climatisation) donne des résultats satisfaisants. Ce nouveau procédé de réduction des bruits permet d'envisager la protection de l'environnement de toutes sortes de conduits industriels. En liaison avec les théories et techniques traditionnelles, cette nouvelle méthode permet de réaliser des sources anti-bruit originales et d'en contrôler leur efficacité.

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Figure 1
REDUKTION VON GASPULSATIONEN DURCH REAKTIVE AKUSTISCHE FILTER

Jaeschke, M. Ruhrgas AG, Essen, Bundesrepublik Deutschland

EINLEITUNG

EXPERIMENTELLE ANORDNUNG

ERGEBNISSE
Das gemessene statische Einfügungsdämmaß bestätigt in den wesentlichen Punkten für die verschiedenen Dämpfertypen das nach (1) berechnete bzw. abgeschätzte Dämpferverhalten. Das dynamische Einfügungsdämmaß ist für Strömungsgeschwindigkeiten von bis zu 20 m/s im allgemeinen nur geringfügig niedriger als das statische Einfügungsdämmaß. Innerhalb der Reproduzierbarkeit der Messergebnisse von +4 dB besteht aber Übereinstimmung. Bei höheren Strömungsgeschwindigkeiten ist die Wirkungsweise der Dämpfer durch die von ihnen erzeugten Strömungsgeräusche begrenzt.

LITERATUR
(1) G. Kurtze et al.; Physik und Technik der Lärmbekämpfung, G. Braun Verlag, Karlsruhe (1975)
INTRODUCTION

Mufflers usually are symmetrically lined, in the practical design as well as in most theoretical work. In that case the absorber linings of the duct on opposite sides (at \( y = \pm h \)) of the duct are identical to each other (wall admittance \( G(y = +h) = G_+ ; G(y = -h) = G_- \)), and the deliberate choice between the symmetrical wave mode \((p(x, +y) = p(x, -y))\) and the antisymmetrical wave mode \((p(x, +y) = -p(x, -y))\) is made in favour of the symmetrical solution due to its much lower attenuation as compared to the antisymmetrical solution. Both of the modes are uncoupled of each other.

THEORY

In ducts with unsymmetrical linings \((G_+ \neq G_-)\) the wanted propagation constant \( \Gamma \) of the sound pressure

\[
p(x, y) = (p_s \cdot \cos \gamma y + p_a \cdot \sin \gamma y) e^{-\Gamma x}
\]

is calculated from the solution \( E = (\epsilon h)^2 \) of

\[
(\sqrt{E} \cdot \tanh \sqrt{E - jU_s}) \cdot (\sqrt{E} \cdot \cosh \sqrt{E + jU_s}) = U_a^2
\]

where \( E = \sqrt{E - (k_o h)^2} \); \( U_+ = k_o h \cdot Z_o G_+ \) (absorber function)

\[
U_s = \frac{1}{2} (U_+ + U_-) ; \quad U_a = \frac{1}{2} (U_+ - U_-)
\]

\((k_o = \text{free field wave number} ; Z_o = \text{wave impedance}).\)

The characteristic equation (1) for \( E \) contains the two equations for the symmetrical mode (1st factor) and the antisymmetrical mode (2nd factor) as special cases if the antisymmetrical component \( U_a \) of the absorber function vanishes: \( U_a = 0 \). The coupling between the two modes, if \( U_a \neq 0 \), will change the propagation constant of the least attenuated wave form. An approximate solution of (1) on the basis of developments by continued fractions will be presented.

EXPERIMENTAL RESULTS

Measurements with absorber splitters of different thicknesses on opposite duct sides confirm the theoretical finding that the attenuation at low frequencies is about that of a muffler with only thick splitters, whereas at medium and high frequencies the unsymmetrical muffler behaves like a silencer with only thin splitters.
Généralités

On se propose de déterminer la meilleure utilisation possible d'une surface absorbante donnée pour obtenir un effet prédéterminé dans un guide acoustique de forme géométriquement simple.

On sait que les effets de la diffraction sur des morceaux de produits absorbants peuvent être très favorables dans certaines bandes de fréquences, compte tenu des couplages existant avec les morceaux absorbants voisins.

On envisage ici des exemples simples qui permettent une vérification expérimentale aisée ; mais la technique employée à un caractère général, les limitations intervenant seulement au niveau du calcul numérique sur ordinateur. En principe, on peut considérer simultanément les variations de section et d'impédance des parois.

Méthode utilisée

La technique de calcul est la théorie modale ; l'hypothèse de paroi à réaction localisée est donc faite ; elle est presque toujours raisonnable. Par une manipulation convenable des produits de matrices généralisées qui traduisent les propagations et les changements d'impédance le problème est résolu de proche en proche en limitant le nombre des mémoires nécessaires et, pour l'onde plane, à 7 ou 8 l'ordre des matrices à inverser.

Exemples d'application

Pour la simplicité, on choisit un guide à section rectangulaire ; l'onde incidente est plane, les discontinuités intéressent seulement l'impédance normale des parois.

Trois exemples sont proposés d'un traitement discontinu par morceaux sur une paroi, deux parois opposées, deux parois adjacentes.

Pour toutes les applications numériques et les essais, le matériau utilisé est une laine de verre dont l'impédance est très bien approchée par une loi type KOSTEN-ZWICKKER, JANSSSEN légèrement modifiée.

La figure ci-contre montre le gain d'affaiblissement obtenu par le découpage en éléments égaux, découplés, d'une longueur donnée de laine de verre tapissant deux parois opposées du guide.

\[ L = \text{longueur des éléments (cm)} \]
THE RATIONAL DESIGN OF AUTOMOTIVE MUFFLERS

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INTRODUCTION

The design of engine exhaust silencing systems for automotive application involves a compromise between conflicting requirements. These include environmental constraints, e.g., statutory noise and smoke levels; engine performance constraints, e.g., back pressure; space and positional constraints imposed by the vehicle layout with weight and cost constraints.

Traditional design methods are based on an empirical approach depending on past experience and simplified acoustic principles. Such methods can involve testing up to forty or fifty different exhaust systems before an acceptable design is obtained. The trend towards improved engine performance, giving a corresponding increase in exhaust noise, with the trend towards more restrictive noise legislation together make a more rational approach to design increasingly desirable.

A RATIONAL DESIGN PROCEDURE

A systematic design procedure requires a practical prediction method for acoustic energy propagation through the exhaust system, based on rational models of component performance (1). In assessing acoustic performance provision must be made for the effects of temperature gradients, high speed gas flow and flow acoustic coupling. A reliable procedure is also required for measuring or specifying the engine noise in terms of a driving signal and a source impedance (1,2).

It has been shown in (1,2) that appropriate linear acoustic models do provide satisfactory predictions of acoustic energy transport through the exhaust system. Uncertainties arise due to inadequate data on flow temperature and to flow acoustic interactions which generate flow noise within the exhaust system. Adequate descriptions of the noise source characteristics also present problems, since the insertion of a silencer or change of the exhaust duct modifies the source coupling (1).

Within the limitations just noted, a simplified systematic design procedure has been developed which gives consistent results. The exhaust noise driving signal and gas flow conditions are determined first, using test bed measurements. An initial layout of the system is then made. This follows a systematic procedure to take due account of noise levels, component lengths and diameters, back pressure, flow noise and flanking transmission, to identify a good compromise within the installation constraints.

The layout specifies the length and diameter of pipe runs and of silencer boxes with the general internal arrangement of the latter. The next step is to calculate details of the silencer internal dimensions and to predict system performance. An iteration between this step and the system layout is performed, to produce an optimum design. The final stage is the construction of a prototype and its evaluation.

A recent example is provided by the exhaust system designs for the Quiet Heavy Vehicle project, where an insertion loss of over 40 dB was required with the back pressure limited to 45 mm of mercury. The first prototype, though acoustically satisfactory, required detail modification to further reduce back pressure and to control flow noise. The third and subsequent prototypes all surpassed the design specification, reducing exhaust noise from a 350 H.P. turbocharged engine to levels below 69 dBA at 7.5 m.

REFERENCES

(2) E. K. Bender and A. J. Brammer 1975 JASA 58 22-30
INTRODUCTION

In modern automobile muffler designs the flow is usually reversed. The acoustic attenuation of such flow-reversing chambers is difficult to predict using classical methods. A technique based on the finite element method was developed for this purpose. The method was first tested on a simple muffler expansion chamber, then on flow-reversing chambers. Finally, the acoustic attenuation of a complete muffler was determined by using electrical network theory to connect together the transfer parameters of each muffler component.

THEORY

A finite element approximate description of the variational formulation of the acoustic system is first produced. Then using different boundary conditions at the outlet of the system, the system transfer parameters are calculated by solving a set of algebraic equations. A computer program was written to calculate attenuation. Two dimensional rectangular finite elements of variable thickness were used so that the volume of real elliptical section chambers could be approximated.

EXPERIMENT

Experimental results were obtained so that comparisons could be made with the theoretical results for expansion chambers, flow-reversing chambers and complete mufflers comprised of straight pipes, flow-reversing chambers and Helmholtz resonators. Using an acoustic driver and examining the standing wave ahead of the muffler, with a traversing microphone, the incident wave can be extracted. Comparing the incident wave with the transmitted wave measured with a second microphone gives the muffler attenuation. A steady 'quiet' air flow was also used in some experiments to find the effect of flow on attenuation. The flow was supplied from a blower and passed through a plenum chamber to the muffler. Fig. 1 shows the attenuation (i.e. transmission loss) of a typical reversing chamber.

CONCLUSIONS

Good agreement was obtained between experiment and theory. Flow was found to have little effect on attenuation except for short expansion chambers. The finite element technique is a powerful tool for the analysis of practical muffler chambers.

REFERENCES

Es werden zwei Klassen kreiszylindrischer Absorptionsschalldämpfer untersucht: solche, bei denen der Querschnitt in Kreisringe unterteilt ist (Figur a) und solche mit der Unterteilung in Sektoren (Figur b). Betrachtet man jeweils nur einen Modus und den eingeschwungenen Zustand, läßt sich das Schallfeld eines Kreisringes oder Sektors formal mit folgendem Produktansatz beschreiben:

\[ p = P(r)P(p)P(x) = \left\{ \begin{array}{ll}
A_r I_m(g_{r}r) + B_r K_m(g_{r}r) \\
A_p \exp(g_{p}p) + B_p \exp(-g_{p}p) \\
A_x \exp(g_{x}x) + B_x \exp(-g_{x}x) 
\end{array} \right. \]

Durch Gradientenbildung erhält man daraus die Schnellen. Für die Komponenten der Ausbreitungsmaße gilt:

\[ \sigma^2 = \sigma_{r}^2 + \sigma_{p}^2 \frac{r^2}{r^2} + \sigma_{x}^2 \]

Bei in x-Richtung "kassettierten" Kreisringen oder Sektoren ist \( \sigma_x = 0 \), alle anderen \( \sigma_x \) sind gleich.

Man hat nun mit den Rand- und Stetigkeitsbedingungen (Gleichheit der Schalldrücke und der Normalkomponenten der Schnellen beiderseits der Trennflächen \( r = \text{const.} \) bzw. \( p = \text{const.} \) benachbarter Schichten) gerade genug Gleichungen, um die "Verteilungsfunktionen" \( A, B \) zu berechnen oder zu eliminieren und die Komponenten der Ausbreitungsmaße zu ermitteln. Das sich ergebende Gleichungssystem ist ähnlich rekursiv lösbar wie bei rechteckigen Anordnungen /1/. Der Rekursionsalgorithmus wird erläutert, und berechnete normierte Dämpfungscurven werden diskutiert. Weiter wird gezeigt, daß es runde Dämpfer gibt, die im ganzen Frequenzbereich die gleiche Dämpfung haben wie bestimmte rechteckige Anordnungen.

Figur a

Figur b

/1/ WALSDORFF, J.: ACUSTICA 37 (1977), Heft 5
On fait l'analyse de sortes de silencieux circulaires avec des diamètres à partir de Ø 100 jusqu'à Ø 1000. On a étudié leurs constructions et caractéristiques et on a fait des conclusions en vue de leur application.

Les silencieux circulaires mis en œuvre sont construits de deux manières, à savoir:

a) les silencieux circulaires ne comportant pas de bulbe central

b) les silencieux circulaires comportant un bulbe central

De la première sorte sont élaborés les silencieux avec diamètre à partir de Ø 100 jusqu'à Ø 500 avec deux longueurs de 500 et de 1000 mm. De la deuxième sorte élaborés les silencieux avec diamètre interne à partir de Ø 150 jusqu'à Ø 1000 avec longueur 1 = kD avec k = 1,6. Concernant tous les types analysés la construction est analogue - un gaine extérieure en tôle d'acier et un cylindre interne en tôle perforée comportant une différence des diamètres de 200 mm.

La deuxième sorte contrairement à la première contient un bulbe. Et les deux éléments - la partie interne et la partie externe des silencieux sont partagées avec deux réflecteurs acoustiques en trois chambres séparées. Ici la matière absorbant ininflammable est installée avec un poids spécifique différent entre $\sigma = 50 \text{ kg/m}^3$ et $\sigma = 150 \text{ kg/m}^3$. Les caractéristiques techniques sont construits et étudiés par expérience et que représentent la attenuation $\Delta L (\text{dB/okt})$ et les pertes de charge $\Delta p (\text{kg/m}^2)$.

Conclusion: 1. Des silencieux avec des poids spécifiques différents avec section en longueur donnent une haute attenuation.

2. Les pertes de charge sont insinifiantes pour une vitesse de l'air jusqu'à 20 m/s.

3. Pour les silencieux avec diamètres intérieurs de l'ordre de Ø 150 mm le bulbe central n'améliore pas la caractéristique d'atténuation.
MEASUREMENTS ON REACTIVE TYPE MUFFLERS

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INTRODUCTION

The performances of acoustic filters to be used as mufflers in exhaust systems are commonly expressed in terms of Transmission Loss (T.L.) or Insertion Loss (I.L.). However, without precautions, the use of these quantities in practical muffler design may be imprudent, especially if reactive filter elements are to be used. (see Ref./1/)

DEFINITIONS

A more convenient way to formulate the problem is to define the Transfer Matrix T of a muffler. It relates the acoustic pressure p and the volume velocity q at one end of the filter to that at the other end:

\[ T = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \begin{bmatrix} p_1 \\ q_1 \end{bmatrix} = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \begin{bmatrix} p_2 \\ q_2 \end{bmatrix} \]

This formulation avoids the difficulties encountered with I.L. and T.L. In particular T is representativ of the muffler only and not of the system in which it is placed. Moreover the performance of a complete exhaust system can be calculated by making the product of the T-matrix es of its successive elements. Extensive measurements showed the validity of this approach (see also Ref./1/)

SYNTHESIS-METHOD

More difficult than calculating or measuring the performances of existing filters, is the inverse problem which consists in designing a filter in such a way that it satisfies some given requirements. This synthesis-problem has been resolved for certain expansion-type filters. The method, based on T matrix calculations, is analogue to synthesis-methods for electrical transmission-line circuits. (Ref./2/)

However, it had to be generalized and adapted to muffler design, mainly to take into account important influence of a continuous gasflow on the propagation of energy.

REFERENCES

/1/ P.STEENACKERS and H.MYNCKE Transfer matrix measurements on reactive filters - DAGA 76 - Heidelberg.
LA REDUCTION DU BRUIT PRODUIT PAR LES INSTALLATIONS DE CONDITIONNEMENT ET VENTILATION DANS LE MILIEU AMBIANT

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INTRODUCTION

Compte tenu des motifs technologiques d'usinage, quelques-unes des sections de l'industrie textile ont des conditions spéciales de microclimat.

Il est question d'assurer les paramètres d'humidité et de temperature de l'air dans les halles de fabrication.

L'assurance de ces paramètres en certaines limites est réalisée à l'aide des usines de conditionnement de l'air, ayant une capacité de 100,000 mc/h à 300,000 mc/h.

Le niveau du bruit, transmis dans le milieu ambiant, dépasse les limites admises par les normes.

SOLUTIONS

1. Le montage d'atténuateurs actifs à lame aux prises d'air

Pour réduire le niveau du bruit on a monté des atténuateurs aux prises d'air. Les dimensions utilisées sont: L = 1,5 m, H = 2,7 m, L = 2 m et d = 135 mm entre lames.

Les réductions de bruit réalisées avec ces atténuateurs se placent entre 17 et 24 dB(A).

2. Écrans acoustiques phonabsorbants

Devant les atténuateurs de bruit on a placé, à 2 m distance, un écran acoustique en tôle revêtu de matériaux phonabsorbants.

L'écran a les suivantes dimensions: 4000 x 6000 mm. Les atténuations de bruit réalisées à l'aide de cet écran se placent entre 10 et 16 dB.

CONCLUSIONS

Par l'application des mesures citées mentionnées nous avons réussi de réduire le niveau du bruit dans le milieu ambiant jusqu'à la limite admise par les normes.

Le niveau du bruit à 65 m de la source a été réduit jusqu'à 46,5 dB par rapport à 45 dB admis par les normes.
Le Principe de Huygens appliqué à l'Acoustique et la théorie des découpages ont permis de développer la théorie des sources secondaires : dans l'espace-temps existe un certain champ acoustique $C(M,t)$ produit par des sources $K$ ; si $O$ est l'opérateur liant le champ aux sources, on a : $O.C = K$. Partageons cet espace en 2 parties $V_1$ et $V_2$ séparées par une surface ou une zone d'épaisseur non nulle $S : V = V_1 + S + V_2$. Mathématiquement, cela se traduit par un opérateur de découpage $s$ tel que :

$$s(M \in V_1) = 0, \quad s(M \in V_2) = 1, \quad 0 < s(M \in S) < 1 \quad \text{(ou inversement)}.$$

Les sources $K$ se partagent de même. Le champ à l'intérieur de $V_2$ peut rester identique à lui-même si les sources de $V_1$ sont remplacées par une densité continue de sources sur $S$ répondant à l'équation : $K^h = (os - s0)C$. On démontre que ces sources sont de 2 sortes : des sources de débit $(q^h = v \cdot \text{grad } s)$ et des sources de force $(f^h = p \cdot \text{grad } s)$. Cette étude amène, après discrétisation des formules, à plusieurs applications :

- 1ère la restitution parfaite d'un événement sonore (concert...) c'est-à-dire l'Holophonie. La discrétisation peut porter sur une source (monophonie), 2 sources (stéréophonie), ou 4 sources ou plus (quadriphonie, tétraédrophonie, myriaphonie) [1],[2].

- 2nde si les sources $K^h$ émettent en même temps que les sources de $V_1$ la pression calculée mais changée de signe, le principe est celui des Barrières Acoustiques Actives : réseaux plans (si les sources secondaires sont coplanaires), ou non plans. La zone protégée $V_2$ peut être soit un domaine fermé (voir figure), soit un domaine ouvert : c'est la source de bruit qui est alors entourée par la barrière. Dans chaque cas, la discrétisation impose des sources particulières [2],[3],[4].

Exemple de densité continue de sources :

- cas du découpage d'une onde sinusoidale sphérique par un parallélépipède rectangle (Figure ci-contre).
EFFECT OF SOURCE LOCATION ON NOISE SHIELDING OF SHADOW ZONES BY WEDGE-SHAPED BARRIERS

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INTRODUCTION

The exact solution for diffraction of point source generated waves by a rigid wedge [dating back to Sommerfeld and MacDonald at the turn of the century] is typically given in the form of a sum of one or more definite integrals or in an approximate form valid when both kr ≫ 1 and kr₀ ≫ 1. The present paper reports a detailed analytical and numerical study of the solution when kr₀ not necessarily large. Possible eventual application is in analysis of wing shielding of engine noise.

REMARKS AND CONCLUSIONS

1. Revised Numerical Integration Scheme. As pointed out by Ambaud and Bergassoli (Acustica, 1972), the integration in their expression for the diffracted wave presents some technical difficulties. To overcome these, the variable of integration is changed such that integrand is uniformly bounded and limits are finite. Integration is performed along a steepest descents path of finite length (ending at essential singularity). Formulation amenable both to rapid computation and analytical approximation.

2. Source on Edge. In accord with related previous analyses of Rayleigh, Waterhouse and Tuzhilin, field is spherically symmetric, amplitude enhanced by 2π/θ, where θ is wedge exterior angle.

3. Total Acoustic Power Radiated into Shadow Zone to good approx. when kr₀ > 1, source not too near wall, is [.003/kr₀]₁/₂ of total free field power of source.

4. Power into Shadow Zone per Unit Area of Shadow Zone Boundary is (.35)I/(kr) where I is free field intensity at same point. Formula valid only for kr > 1; at small kr, expression depends in complicated manner on source location but is still singular (pressure finite, fluid velocity singular) although integrable, at r = 0.

5. Uniform Asymptotic Formula Valid to Excellent Approximation when source as close as (1/6)θ to edge.

6. Moving Source Back Slightly Behind Edge gives dramatic noise reduction in shadow zone: e.g. 3 dB when r₀ only θ/16.

7. Effect of Ambient Flow. For thin screen case, ambient flow tangential to surface, Candel's (JASA, 1973) solution for plane wave diffraction is extended to point source case. Solutions with and without flow related by a Galilean transformation.

8. Wedge Diffraction Solution is Building Block for solution of almost any radiation or diffraction problem involving planar surfaces and edges. Geometrical diffraction theory of Keller and others can be modified to avoid spurious singularities and yet preserve adherence to reciprocity requirements. Examples of three sided barrier diffraction (revision of 1974 JASA theory) and radiation from vibrating rectangular box are discussed.

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INTRODUCTION

When running at full speed, newspaper printing presses usually cause noise levels up to 105 dB(A) in places where the press crew works most of the time. At A/S Dagbladet Politiken in Copenhagen three Ampress Colormatic letterpress rotaries are installed. At a speed of 50,000 copies per hour the sound level along the control sides 1.0 m from the presses was between 95 and 105 dB(A) with the highest level near the folder. The ceiling of the press hall had a relatively good absorbent lining, but the walls were acoustically hard.

After discussion of different alternatives to reduce the sound level it was decided to build screens in the presses and to give some parts of the walls an absorbent lining.

SCREEN DESIGN

The screen, 2.6 m high, is built in front of the folder and between the printing units. There are several doors which permit good access to the area behind the screen. The screen has many windows in order to get good visual contact between the both sides. The control side of the printing unit is left outside the screen. See figure 1.

A great number of different ready-made wall elements and doors were considered but none of them could fulfill the very stringent ergonometric requirements. It was therefore decided to use a completely tailor-made design. The design selected makes it possible to remove any separate unit after removing four bolts. The units are made of steel profiles and 2.0 mm steel plates. The inner side is covered with an absorbent, 65 mm thick, surfaced with a thin aluminized polyester film. This makes the absorbent resistant to printing ink, oil, paper dust etc. The windows consist of 6 mm laminated glass.

RESULT AND CONCLUSION

The insertion loss was calculated to 4 to 12 dB(A). Measurements showed sound levels between 87 and 94 dB(A) and consequently the resulting insertion loss was 5 to 11 dB(A). See figure 2. The sound level behind the screen, between the printing units and near the folder did not increase. The press crews are very satisfied with the screens. The access to the press is still good and service and maintenance work is not obstructed.

The results concerning both noise control and ergonometrics show that screens are useful tools for noise control in existing newspaper printing halls.
ABSORPTION ACTIVE SPATIALE PAR TRIPLETS ACOUSTIQUES

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INTRODUCTION

L'absorption active des sons de basse fréquence dans l'espace nécessite l'utilisation de barrières, c'est-à-dire de réseaux de sources anti-bruit. Ces sources ont dans le cas général des caractéristiques de rayonnement complexes. Cependant dans le cas particulier d'une barrière sphérique, destinée par exemple à l'insonorisation d'une source fixe et d'étendue limitée (machine, transformateur...) ces sources peuvent être simplement des tripoles à rayonnement cardioïdal. Nous présenterons, à partir d'une étude théorique et expérimentale, les conditions imposées par la réalisation de telles sources.

ANALYSE DU PROBLEME

Pour réaliser une source tripolaire à rayonnement spatial cardioïdal nous avons envisagé l'utilisation conjointe de 3 sources monopoles; deux d'entre elles sont groupées en dipôle et l'ensemble forme ce que nous appelons un "triplet acoustique". Or comme on le sait, en basse fréquence, les sources monopoles courantes ont un rayonnement très peu sphérique. Cela nous permet de décrire ce rayonnement par une expression du type $A_0 e^{-jkr}$ ; nos expériences ont d'ailleurs appuyé cette hypothèse.

Par conséquent, le rayonnement global du triplet en un point de l'espace pourra s'écrire:

$$p(M) = \frac{A_0}{r_a} e^{-jkr_a} + \frac{A_0}{r_0} e^{-j(kr_0+\phi)} + \frac{A_0}{r_b} e^{-j(kr_b+\pi)}$$

Si l'on veut obtenir de ce triplet un rayonnement cardioïdal dans l'espace on pourra vérifier que deux conditions s'imposent :
- d'une part, en admettant que les 3 sources constituent le triplet ont des courbes de réponse fréquentielles identiques, et que les amplificateurs ont une réponse parfaitement linéaire, il faudra imposer au dipôle une correction électronique des niveaux et des phases telle que :

$$A_0 = \sqrt{2} \frac{2 \pi}{r_0^2} A_2^2 \left[ \frac{r_0^2 + d^2}{r_0^2 - d^2} \right] - \frac{2 \pi^2}{r_0^2 - d^2} \cos 2 kd$$

$$\phi_0 = \frac{A_0}{r_0} \frac{\sin k(r_0+d)}{\cos k(r_0+d)} - \frac{r_0+d}{r_0-d} \cos k(r_0-d)$$

- d'autre part il sera nécessaire d'agir sur la forme de l'onde émise par le monopôle central en imposant un rayonnement du type :

$$\frac{A_0}{r_0} e^{-j0(t-\phi(t,\theta))}$$

Cette notion de surface d'onde est d'une importance capitale en absorption active tridimensionnelle et conditionne, au même titre que le pilotage des sources, la qualité de l'absorption obtenue.

CONCLUSION

Les 2 conditions décrites ci-dessus ont été démontrées expérimentalement, et cette approche expérimentale du problème permet de situer certaines orientations de recherche actuelles en matière d'absorption active.
DETERMINATION DE L'EFFICACITE REELLE D'UN ECRAN ABSORBANT AU BORD D'UNE VOIE RAPIDE

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INTRODUCTION

On dispose actuellement de trop peu d'éléments pour prévoir l'efficacité réelle d'un écran absorbant installé au bord d'une voie rapide et le gain par rapport à un écran réfléchissant.

Nous avons tenté de mettre au point une méthode de mesure in-situ de l'absorption.

BASES THEORIQUES

Nous faisons une mesure face à l'écran réfléchissant et une mesure face à l'écran absorbant. La différence entre les niveaux de pression acoustique (calculés à partir d'une analyse statistique du bruit de la route, ref 1) est trop faible devant l'incertitude de mesure qui est de 1dB. En prenant des microphones directionnels on aura l'équation (1).

\[ L_{r} - L_{a} = 10 \log \frac{D + 1}{D+1-\alpha} \]

avec \( L_{r} \) le niveau face à l'écran réfléchissant, \( L_{a} \) le niveau face à l'écran absorbant, \( D \) le facteur de directivité (D<1) de la face arrière sur la face avant, \( \alpha \) le coefficient d'absorption.

RESULTATS D'EXPERIENCES

\( D \) a été déterminé en chambre sourde en intégrant la courbe de directivité. L'atténuation \( L_{r} - L_{a} \) mesurée est de 4dB pour un \( \alpha \) de 0,7. Le résultat est significatif et nous obtenons un \( \alpha \) mesuré in situ à ± 0,1.

Nous recherchons actuellement une relation simple avec les coefficients d'absorption mesurés en laboratoire.

CONCLUSIONS

Nous disposons d'une méthode simple de mesure de l'absorption d'un écran routier. Nous pouvons ensuite à l'aide d'un programme de calcul déterminer le gain obtenu lorsque l'on pose un revêtement absorbant sur un écran réfléchissant.

REFERENCES

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Noise control by means of absorbing materials is a very inadequate technique for sounds of low frequencies (say from 20 to 500 Hz) because the thickness of materials necessary to produce a constant attenuation increases with decreasing frequency. Another technique can be used by which destructive interference is applied to reduce noise level: it is active noise attenuation. We call active acoustical barrier (AAB) a set of antinoise sources distributed around a surface separating the controlled zone from noise sources. A theory of AAB is proposed, which is deduced from general results presented in recent papers (1) (2) (3).

**THEORY**

(The symbols used in this paper have been defined in ref. 1).

The acoustic pressure \( p(M, t) \) to be reduced may be written in the form:

\[
\mathbf{M} \mathbf{P}(M, t) = \frac{1}{4\pi} \int \frac{q(M, P, t)}{r} \, da
\]

where \( \mathbf{M} \) is the "cutting-out" operator and \( q(M, P, t) \) the anti-noise sources density. In any practical system, this continuous density \( q \) must be replaced by a set of discrete sources \( q(M, P_{jk}, t) \) and eq. 1 becomes:

\[
\mathbf{M} \mathbf{P}(M, t) = \frac{1}{4\pi} \sum_{jk}^{N} \frac{q(H_{jk}P_{jk}, t)}{p_{jk}} \Delta \sigma_{jk}
\]

where \( N \) is the total number of antinoise sources and \( \{P_{jk}\} \) a set of points of \( \Sigma \). Calling \( p^A(M, t) \) the resultant antinoise sources radiation and \( p^A(M, s) \) its Laplace transform, the transmittance of an AAB can be deduced from eq. 1 and 2.

\[
p^A(M, s) = -\frac{1}{4\pi} \sum_{jk}^{N} \frac{p(M_{jk}, s)}{p_{jk}} \left( \frac{\cos \varepsilon_{jk}}{p_{jk}} \right) \left( \frac{s}{c} (\cos \theta_{jk} + \cos \varepsilon_{jk})e^{-\frac{(\rho_{jk}+\tau_{jk})s}{c}} \right)
\]

where it is supposed that \( \tau_{jk} >> \lambda_M \), \( \lambda_M \) being the highest wave length to be attenuated. One can see that an AAB may be considered as a multichannel system including:
- first order transmittances \( a_{jk} \) with \( a_{jk} = \cos \varepsilon_{jk}/\rho_{jk} \) and \( \tau_{jk} = \rho_{jk}/c \)
- derivative and delay units connected to dipolar sources \( \cos \theta_{jk} \)
- flat response transducers and amplifiers.

The positions and the number of microphones and antinoise sources can be determined with computer assistance. It can be shown that the distance between two transducers must not be higher than \( \lambda/2 \) for a given frequency.

Theoretical and practical results will be presented in a next paper to be published in Noise Control Engineering.

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(3) CANEVET (G.), JESSEL (M.) & MANGIANTE (G.), Noise Control by means of active absorbers, 1976 Noise Control Conference, Warsaw.
Dans quelle mesure le modèle bidimensionnel de diffraction par un écran mince, plan, inﬁnitement long, réléchissant comme le sol sur lequel il est placé, détermine des résultats satisfaits pour le problème tridimensionnel analogue. Deux méthodes de calcul de l’efﬁcacité de l’écran sont comparées.

Enoncé du problème : déterminer le potentiel des vitesses $\Phi(M)$ qui vériﬁe sur l’écran et le sol, la condition aux limites de Neumann $\partial n \Phi(P) = 0$. Le champ incident est $\Phi_0(M) = \Phi_{OS}(M) + \Phi_{OS}(M)$, où $S'$ est le symétrique de $S$ par rapport au sol et $Z = 2\Phi_0$.

EFF (dB) = 20 log $\frac{\Phi(M)}{2\Phi_{OS}(M)}$

Première méthode : c’est une approximation analytique (1), (2), basée sur les résultats asymptotiques obtenus dans la diffraction d’un écran semi-inﬁni. On a quatre conﬁgurations possibles d’où :

$\Phi(M) = \frac{1}{2} \Phi_1(M)$.

Seconde méthode : on représente le champ diffracté $\Phi_d(M)$ par un potentiel de double couche, $\Phi_3(M)$ ; l’équation intégrale obtenue est résolue numériquement d’où $\Phi(M) = \Phi_0(M) + \Phi_d(M)$.

$\Phi_d(M) = 2 \Sigma_0 \frac{\mu(P) - \mu(P)}{3\pi n(P)} d\sigma(P)$ ; $\mu(P)$ la densité de la double couche, $G = \frac{1}{4} H^0_0(\alpha)$ avec $k$ le nombre d’onde, $R$ la distance à l’origine, $H^0_0$ la fonction de Hankel.

La source sur le sol : $S(0, Y_0)$

--- résultats de la 1ère méthode

+ résultats de la 2ème méthode

+ résultats pour une file de sources ponctuelles (7 sources) ; 0 résultats pour une seule source ponctuelle ; ● enveloppe des maxima de l’efﬁcacité calculée par la 2ème méthode, deﬁnie l’efﬁcacité utile.

Les écarts observés sur les résultats expérimentaux sont dus aux interférences entre les 7 sources. Le calcul par la méthode du potentiel de double couche donne une efﬁcacité minimale plus proche des résultats expérimentaux sauf au voisinage de l’écran. L’hypothèse émise se revèle très convenable.

(1) Z. MAEKAWA, Mémoires of the faculty of engineering-Kobé University (1965) 11,29.


Introduction

Le but de cette étude est la mise au point d'une méthode expérimentale, permettant d'obtenir "in situ", l'isolement acoustique d'un écran mince, en fonction de la fréquence.

Bases théoriques

Un microphone, placé derrière un écran, ne permet pas, par une mesure directe d'évaluer son isolement. La pression efficace mesurée est égale à la somme quadratique des pressions efficaces dues aux ondes se propagant à travers l'écran, et diffractée par son arête.

La solution envisagée repose sur l'utilisation de la fonction d'intercorrélation entre le signal émis et le signal reçu ; soit :

\[ C_{xy}(\tau) = \frac{1}{T} \int_{0}^{T} x(t) y(t - \tau) \, dt \]

Cette fonction elle possède une transformée de Fourier

\[ F[C_{xy}(\tau)] = S_{xy}(\nu) \]

dont le module nous donne l'interspectre.

Technique expérimentale

On réalise dans la grande salle sans écho, la maquette d'un écran mince ; une source placée d'un côté de l'écran, émet un bruit à spectre étendu. On dispose de part et d'autre de l'écran, deux microphones qui prélevent le signal émis et le signal reçu. Un corrélateur permet de calculer l'intercorrélation des deux signaux. La fonction obtenue présente des maxima, dont le premier correspond au trajet le plus court, donc à l'onde traversant l'écran. Cette fonction est ensuite échantillonnée ; un programme sur ordinateur permet d'en calculer la transformée de Fourier et d'obtenir l'interspectre \( S_1 \).

La mesure est répétée sans la présence de l'écran et l'on calcule l'interspectre \( S_2 \) :

\[ A = 20 \log \frac{S_2}{S_1} \]

on obtient directement l'isolement acoustique en fonction de la fréquence.

Conclusion

La méthode permet, en enregistrant "in situ", les signaux en fonction du temps, d'évaluer la transparence d'un écran, en ne considérant que l'onde le traversant.
SOUND DIFFRACTION BY A SCREEN (PRACTICAL METHOD BASED ON MACDONALD'S RIGOROUS SOLUTION)

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MACDONALD'S RIGOROUS SOLUTION AND EXPERIMENTAL RESULTS

In predicting and controlling the propagation of noise in the open air, the theory of the free-field diffraction of a spherical sound wave by a thin half-plane is very important.

From Fig.1, it is clear that Macdonald's rigorous solution and experimental results agree extremely well no matter where the sound source and the observing point are located.

NEW APPROXIMATE FORMULA AND PRACTICAL METHOD

Macdonald's solution expressed in terms of integrals of Hankel functions need long calculating time in a computer and is hardly used in practical cases.

Then we present the followed simple approximate formula.

\[ \phi = U(\theta_0) + U(-\theta_0) \]

\[ U(\theta_0) = n(-N) e^{\frac{1}{k R}} + \text{sgn}(-N) \frac{2}{R_1 (R_1 + R)} A(R) \]

Where

\[ A(R) = \frac{1}{2 \pi |N|} \left[ 1 - \left( \frac{|N|}{|N|+0.4} \right) \right] e^{i (k R - 0.8 \pi)} \text{ for } 0.8 < |N| \]

\[ (C+1)^2 e^{1/k R} \text{ for } |N| < 0.8 \]

\[ C = \begin{cases} 
0.5 - \sqrt{|N|}/1.68 & \text{for } |N| \leq 0.1 \\
1.01(|N| - 0.75)^2 - 0.17 & \text{for } 0.1 < |N| < 0.8 \\
S = \sqrt{|N|} (1 - |N|/0.75) & \text{for } |N| \leq 0.5 \\
1.16 \sqrt{|N|} (1 - |N|/0.8) & \text{for } 0.5 < |N| < 0.8 \\
\end{cases} \]

\[ n(\theta) = \begin{cases} 
1 \text{ for } \theta > 0 \\
0 \text{ for } \theta \leq 0 \\
\text{sgn}(\theta) = 1 \text{ for } 0 > \theta \\
-1 \text{ for } 0 \leq \theta \end{cases} \]

Level difference between attenuation by a screen by the above Eq. and that by Macdonald's solution is smaller than 0.5dB only if \( \lambda/4 < R_1 \).

Finally we propose a practical method based on Eq. (1), by which the detailed peak-dip due to interference between several waves are not drawn in SPL distributions but only monotonous gradual variations are drawn, as shown in Fig.1.

Fig.1 Equi-SPL contours when 500Hz. —— by practical method, —— by Macdonald's, —— by Maekawa's chart, —— measured by 1/3 oct. band noise.
SOUND DIFFRACTION BY TRAPEZOIDAL BARRIERS

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INTRODUCTION

The diffraction of sound by trapezoidal barriers has been investigated with the goal of providing a means for predicting the sound-shielding effects of barriers such as earth berms. The surfaces of the barriers are assumed to be characterized by a (local reaction) point acoustic impedance. The increased insertion loss attributable to the finite surface impedance may thus be used in selecting surface facings once the barrier geometry has been chosen.

THEORETICAL BASES

We consider a source of sound such that near the source the sound pressure field may be represented as spherical waves emanating from the source. The source-barrier-listener configuration is depicted in Figure 1. An approximation for the pressure diffracted into the shadow of the barrier \((\theta_L, \theta_L - \pi)\) has been developed by Pierce and Hadden (1) by a juxtaposition of the approximate solutions of the problem of single-edge diffraction by an obtuse wedge with finite surface impedance (1,2). The juxtaposition is effected by considering the pressure arriving at the receiver to have originated at a fictitious source located above the actual source on the extension of the plane of the top of the barrier, with the fictitious source strength adjusted so that the pressure at the diffracting edge nearest the receiver is the same as that due to the actual source.

RESULTS

The results are most conveniently expressed in terms of the insertion loss for the barrier. Only a general description and a few typical results are listed here. For large separations of both source and receiver from their respective diffracting edges, the insertion loss contains contributions from geometrical spreading losses and terms dependent on the surface impedance. The latter terms may be divided into two categories: one which shows the dependence on the angles \(\beta, \theta_s\) and \(\beta, \theta_L\); and another which depends on the barrier width \(w\). Of these two, the latter is more important: for \(\beta = \beta_s = 270^\circ, \theta_s = \theta_L = 30^\circ, r_s = r_L = 50\lambda\) and for an impedance typical of turf at 1000Hz, the angle-dependent term is 0.9dB while the width-dependent term is 6.8dB for \(w=10\lambda\) and 8.6dB for \(w=20\lambda\). Thus the inclusion of the finite-impedance effects is seen to be significant in predicting the insertion loss for a barrier.

REFERENCES

AN APPLICATION OF THE INTEGRAL EQUATION METHODS TO DIFFRACTION EFFECTS BY NOISE BARRIERS

Furue, Y. Kyoto University, Kyoto - Japan
Terai, T.
Matsu'ura, K.

INTRODUCTION

Although many works have been made on effects of noise barriers, most of them are on two dimensional ones. In this paper an approach to estimate approximately the diffraction effects of any shaped rigid barrier based on the surface integral equation method with some assumptions is described. Numerical and the corresponding examples are also presented.

THEORETICAL BASES

We consider any rigid and thin closed surface S, which is consist of two unclosed surfaces $S_1$ and $S_2$, point source $P_s$, and receiving point $P$ in free space.

Supposing the velocity potential on the surface $S_2$, which is dark side acoustically, is small negligibly, the velocity potential $U(P)$ in the region exterior to or on $S$ is given by Helmholtz's formula [1] as shown in eq.(1) or (2), respectively.

$$4\pi U_d(P) + \int_{S_1} U(P)\frac{2}{n_Q}(e^{-ikr_PQ/r_PQ})dS = \begin{cases} 4\pi U(P) & P \text{ is outside of } S \\ 2\pi U(P) & P \text{ is on } S \end{cases}$$

(1)

(2)

where $U_d(P)$ is direct component from the source, $n_Q$ is outward normal to $S$ at $Q$, $r_PQ$ is the distance from $P$ to $Q$, $S_1$ is illuminated surface and $k$ is wave number.

Eq.(2) is a Fredholm integral equation of the second kind, and could be solved approximately by converting it into a system of algebraic equations.

$$4\pi e^{ikr_{Ps}} = \sum_{j=1}^{N} \frac{U_j(1+e^{-ikr_{ij}})}{r_{ij}} 
\cos(n_{ij}r_{ij}) \Delta S_j = 2\pi U_i \quad (i=1,N)$$

(3)

Substituting the solution of eq.(3) into eq.(1), which may be also approximated similar to eq.(3), the velocity potential at any point can be calculated.

In the case $U(P)$ on $S_2$ is not assumed so small, another integral equation must be provided in addition to eq.(2). [2]

NUMERICAL AND EXPERIMENTAL

As an example of noise barrier a rigid and thin box-like one with open top (35cm wide, 20cm depth and 15 cm height), at the center of which a point source is located (1kHz), is considered. The calculated diffraction effects ($10\log[U(P)/U_d(P)]$ dB) at the points all of which are 80cm distant from the source and the corresponding measurements carried out in anechoic room are shown in Fig.1.

Agreement between calculations and measurements at the points of our interest (upper half) is satisfactory.

REFERENCES


Fig.1 Calculation(solid line) and Measurements(°)
INTRODUCTION.

A worldwide growth of interest in hearing protection has given rise to a growing number of national standards for the measurement of real-ear attenuation of hearing protectors. There is a high degree of similarity between many of the standards in current usage and the majority are based on a Real-Ear Auditory Threshold technique. However, amongst the new standards which have appeared in recent years is one notable exception. This, the South African Standard SABS 572-1973, is based on a loudness balance technique which is a modified version of a technique described in an earlier publication.

The advantages of using a loudness balance technique rather than a threshold technique are immediately obvious. The low frequency physiological noise problem is obviated and, because of less stringent criteria, the cost of the test environment construction is drastically reduced.

In this laboratory there is a continuing interest in the methodology of hearing protector attenuation measurement. Although contemporary measurement methods represent a significant improvement on previous techniques it is recognised that a completely satisfactory answer to the problem of protector attenuation measurement is yet to be found. Therefore, as part of a programme of research into protector attenuation, it was thought worthwhile to carry out an assessment of the South African Standard. Accordingly a series of measurements have been performed following the instructions contained in SABS 572-1973.

EXPERIMENTAL TECHNIQUE

The subject is seated in a diffuse sound field in a semi-reverberant room. An earphone is inserted in one ear and the same ear is occluded by one cup of a circumaural ear protector. The other ear may or may not be occluded by the hearing protector, plug or muff, being tested.

The subject controls the level of the diffuse sound field and balances it against a reference tone from the earphone. The difference in level between the two balance conditions, occluded and unoccluded, is deemed to be the attenuation of the hearing protector.

RESULTS AND CONCLUSIONS

The experiments are not yet completed and a full discussion of results is therefore withheld until the presentation of this paper. Preliminary experiments have generated attenuation values markedly lower than those normally produced by Real-Ear Auditory Threshold tests and this is in agreement with Meij et al. However, standard deviation values have generally been higher than the 5dB deemed to be acceptable by SABS 572-1973.

REFERENCES.

THE EFFECTIVE ACOUSTICAL PERFORMANCE OF HEARING PROTECTORS

Whittle, L. S. Acoustics Unit, National Physical Laboratory, Teddington, Middlesex, England

ACOUSTICAL STANDARDS FOR TESTING HEARING PROTECTORS

The acoustic attenuation of a hearing protector must be sufficient to meet the relevant occupational noise exposure limits; but there are disadvantages in over-protection, including communication difficulties, decreased comfort and increased cost. A reliable knowledge of acoustical performance is therefore important.

Noise exposure limits are commonly expressed in terms of A-weighted $L_{eq}$ since this measure is directly linked to the risk of hearing handicap following repeated exposure. The attenuation of a protector may, however, vary as much as 20 dB(A) depending on the noise spectrum, and to calculate its performance in a given noise one must combine the spectrum with the measured attenuation at different frequencies, eg at octave intervals. Also, the attenuation at a given frequency, though specific for a given type of protector, varies with the head anatomy of the wearer. In choosing protectors for use in a particular plant one must allow for this variability by adopting a conservative value for the effective attenuation such as mean minus standard deviation, or lower quartile, of measurements on typical heads. A complicating factor is that the dispersion is inflated by subjective error in the measurements since no standardised procedure has yet been devised to ensure valid results by a purely physical test on an artificial head.

There have been recent improvements in the real-ear threshold shift test methods. The USA standard Z24.22:1957 is replaced by ASA-STD-1 of 1975 which, like British Standard BS 5108 of 1974, now uses 1/3-octave bands of noise as test sounds. Both require diffuse field conditions, as in a typical factory, and both state ambient noise limits to avoid threshold elevation due to masking, BS 5108 being the more stringent. The test conditions in BS 5108 derive from investigations at the NFL (1,2) where the audiometric advantages of a free-field room were combined with the diffuse field requirement by the use of a regular tetrahedral array of uncorrelated sound sources, with the subject seated at the geometric centre.

TEST RESULTS AND CONCLUSIONS

A test programme with this rig on 5 types of earmuff and 1 type of earplug gave the results below. As in BS 5108 subjects made 2 visits, in a balanced design using 18 subjects instead of the 15 specified.

(a) overall standard deviations of attenuation were 4 to 7 dB for the earmuffs (depending on frequency) and 8 to 11 dB for the earplugs. Similar standard deviations were obtained by the Z24.22 method.

(b) 2-factor analysis provided estimates of the component of variance representing subjective threshold error plus acoustical fit-refit variability. For earmuffs, this was of the order 13-24 dB² (depending on frequency) consistent with the results observed in ordinary audiometry using earphones. For earplugs the variance was greater, doubtless due to the less repeatable acoustical fit of the plugs (though each subject retained his own pair).

(c) intersubject variance for the earmuffs ranged from 7 to 19 dB² and is not the major factor. The contrary was found for earplugs, for reasons which are uncertain, values ranging from 39 to 70 dB²; in this case intersubject variance was the greatest component and the same amount of testing would be more effectively distributed by testing twice as many subjects once only.

REFERENCES

(1) L S Whittle and D H Evans, J. Sound Vib. (1972) 23, 63-76.
(2) L S Whittle (1977), NPL Acoustics Report Ac 79.
INTRODUCTION

All the ear protectors used by people working in areas which the sound pressure level exceeds the established limits (S.P.L. expressed in dB(A)) must reduce the effective perceived noise level so that it doesn't exceed the daily noise dose value during the exposition time.

To have one or several numeric values which summarize the behavior in all of these personal protection devices, would be very useful in order to determine the effective perceived noise level, because when one knows only the attenuation curve, the calculations are complex and it is also necessary to know the octave band frequency level of the problem noise.

EXPERIMENTAL TECHNIQUES

This communication tries to collect the methods, (1), (2) and (3), that had been realized on this subject up today by making a comparative study of the obtained results which each one of the referred to methods, with twelve ear protectors and several Spanish industrial noises that exist nowadays.

RESULTS AND CONCLUSION

The results that we have obtained on one of these ear protectors using each one of the methods are summarized in Table 1 in which we can see the different attenuation values obtained according to the method used.

According with these results, we think that it would be very interesting to adopt only one method to determine dB(A) reduction of ear protectors, as it would be very useful not only for occupational safety and health technicians, but also for working people and for managers.

<table>
<thead>
<tr>
<th>ATTENUATIONS</th>
<th>METHOD</th>
<th>≤ 0</th>
<th>0'1 - 2</th>
<th>2'1 - 4</th>
<th>4'1 - 9</th>
<th>&gt; 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>NIOSH</td>
<td>(dB(A))</td>
<td>23</td>
<td>16'8</td>
<td>14'3</td>
<td>10'3</td>
<td>6'7</td>
</tr>
<tr>
<td>WAUGH</td>
<td>(dB(A))</td>
<td>33</td>
<td>26'8</td>
<td>24'3</td>
<td>20'3</td>
<td>16'7</td>
</tr>
<tr>
<td>BOTSFORD</td>
<td>(dB)</td>
<td>31'9</td>
<td>27'6</td>
<td>26'6</td>
<td>26'4</td>
<td>27'8</td>
</tr>
</tbody>
</table>

REFERENCES

INTRODUCTION

After the promulgation of the second proposal of standard ISO/DP/4869, referring to a new operatory method for determining the attenuation of threshold in hearing protectors, this comparative study has been made so as to prove the efficiency or inefficiency of the method with regard to the one already existing.

STUDY BASE

This study is based on the determination of the attenuation curves at the threshold of hearing protectors following the methods mentioned below:

A compulsory and Technical Standard MT-2 "Hearing protectors"

The acoustical field of aleatory incidence showed in the last method has been obtained in the same way as indicated in the article "Acoustical Field of Aleatory Incidence" (M. Montes, 1976, not published yet).

This study has been made on 9 equipment of personal protection among which the following types are included: Earmuff and earplug.

RESULTS AND CONCLUSIONS

For each one of the hearing protectors that has been studied it has been computed the existing variation among the values obtained according to ISO/DP/4869, on bands of one third of octaves whose central frequencies correspond to 125, 250, 500, 1000, 2000, 3150, 6300 and 8000 Hz., and the standard MT-2 for pure tones of 125, 250, 500, 1000, 2000, 3000, 4000 and 8000 Hz.

From all these differences has been determined the mean value \( m_i \) and the standard deviation \( \sigma_i \) for each frequencies. Table 1.

As shown in Table 1 and having in mind the subjective character of the two operatory methods we can conclude that: the statistical values obtained show that the use of any of these two methods give positive results at the time of determining the attenuation at the threshold of hearing protectors.

<table>
<thead>
<tr>
<th>TABLE 1. THE STATISTICAL VALUES</th>
</tr>
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<tbody>
<tr>
<td>Frequencies (Hz)</td>
</tr>
<tr>
<td>125</td>
</tr>
<tr>
<td>m_i</td>
</tr>
<tr>
<td>( \sigma_i )</td>
</tr>
</tbody>
</table>
A COMPARISON OF THE METHODS OF MEASUREMENT USED IN THE DETERMINATION OF THE ACOUSTIC ATTENUATION OF HEARING PROTECTORS AND FLYING HELMETS

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The "Real Ear at Threshold" method for determining the acoustic attenuation of hearing protectors and helmets, has many advantages, but also several disadvantages, especially where research and development of such devices is being pursued. To test BS 5108 involves major costs in setting up the necessary quiet rooms and the equipment required to generate the noise; and the 30 subject-runs necessary (15 subjects x 2 replications) involves a minimum of some 15 hours of testing. The use of ten discrete one-third octave bands of noise, generally at intervals of one octave, has disadvantages where hearing protectors are combined with communication sets, since poor attenuation in certain one-third octave bands could remain undetected but could cause problems in the masking of speech or warning signals.

Testing methods, other than those using the human in an 'active' mode as in HEAT determinations, are either fully objective methods where the human is not used at all, (such as in artificial ear or dummy head measurements), or they use the human in a 'passive' role, thus combining some of the advantages of HEAT methods with fewer of the disadvantages of the fully objective method.

Such semi-objective methods use the human with a miniature microphone mounted at the entrance to the external auditory meatus and measure the attenuation of hearing protector devices by either insertion or transmission loss.

Current research at RAE Farnborough Human Engineering Division is concerned with developing such semi-objective measures with the aim of producing a testing schedule which gives a consistent, reliable and accurate determination of the attenuation characteristics of hearing protectors or flying helmets.

This paper describes recent work along these lines. Comparisons have been made between several methods of testing using the BS 5108 rig at the Institute of Sound and Vibration Research. Real Ear at Threshold measurements to BS 5108 have been compared with both semi-objective and objective methods, using one-third octave bands of random noise, pure tones and broad-band random noise making use of both insertion and transmission loss techniques. Semi-objective methods made use of miniature microphones and objective measures used a dummy head with a Zwischenki artificial ear and also a standard artificial ear with a flat plate coupler.

The results of pilot studies and of the main studies are discussed and compared, and recommendations are made as to methods which may be used reliably for research and development purposes.
INTRODUCTION

The control of noise from woodworking machinery has often been tackled in a rather ad hoc fashion in the past. Measures have been applied to various parts of the machine on a trial and error basis without knowledge of the location of the source of the noise. The results have, not surprisingly, often been disappointing. For example, damping pads applied to circular saw blades are only partly successful. The work reported here has shown that the wood being processed can be an important radiator of noise, and that, except where the cross-sectional area of the wood is small, the application of damping pads is unlikely to yield noise reductions greater than about 3 dB(A).

EXPERIMENTAL MEASUREMENTS

Near-field sound pressure level measurements were made for a number of woodworking machines, engaged in a variety of tasks, as part of a survey to establish noise control requirements for a large woodworking shop. The (partial) sound power levels of the component parts or surfaces of a machine, including the wood being processed, were computed using the equation

\[ PWL = SPL + 10 \log A \]

The errors involved in this computation are greatest at low frequencies (1) and thus may be minimised by use of the A-weighted sound power level.

The measurements were made using a conventional 25mm condenser microphone provided with a hood in order to discriminate against extraneous noise sources.

WOOD-RADIATED NOISE

Figure 1 shows the near-field sound level measured at increasing distances along planks of softwood, hardwood, and hardwood ply being cut by a 550mm diameter circular saw. Results are expressed relative to the level measured at 0.3m from the cutting process.

With larger timber cross-sectional areas, noise radiated from the wood was found to assume increasing importance, and with 300 x 25mm softwood it was found to be the dominant noise source.

REFERENCES

ACOUSTIC TREATMENTS GIVEN TO HIDROELECTRICA ESPAÑOLA'S EMBAJADORES TRANSFORMER SUBSTATION IN MADRID

M. Requena, J. J. Service Studies and Works of H.E.

INTRODUCTION

This paper presents the acoustic treatment given to Hidroeléctrica Española's Embajadores Transformer Substation located in the central zone of Madrid. The project for the acoustic treatment of this urban transformer substation is described, together with the results obtained by the treatment with regards to the reduction of noise emitted outside.

SOUND LEVELS AT THE S.T. EMBAJADORES

The following measurements, before and after the acoustic treatment, have been made.

BEFORE THE TREATMENT

A = Hot air exits
B = Cold air entrances
C = Hot air chimney

Relaxation damper placed in the Substation transformer of Embajadores

<table>
<thead>
<tr>
<th>Hz</th>
<th>31.5</th>
<th>63</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
</tr>
</thead>
<tbody>
<tr>
<td>A.</td>
<td>dB</td>
<td>72</td>
<td>74</td>
<td>75</td>
<td>74</td>
<td>71</td>
<td>66</td>
<td>64</td>
</tr>
<tr>
<td>B.</td>
<td>dB</td>
<td>71</td>
<td>72</td>
<td>72</td>
<td>66</td>
<td>61</td>
<td>54</td>
<td>45</td>
</tr>
<tr>
<td>C.</td>
<td>dB</td>
<td>74</td>
<td>74</td>
<td>80</td>
<td>74</td>
<td>66</td>
<td>64</td>
<td>58</td>
</tr>
</tbody>
</table>

AFTER THE TREATMENT

Making a comparison of these measurements it is possible to observe that the following reductions of noise levels, have been obtained with the introduction of the relaxation damper:

<table>
<thead>
<tr>
<th>Hz</th>
<th>63</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
</tr>
</thead>
<tbody>
<tr>
<td>dB</td>
<td>5</td>
<td>12</td>
<td>19</td>
<td>26</td>
<td>32</td>
<td>36</td>
<td>32</td>
</tr>
</tbody>
</table>

(*)Measurements made between 5 and 6.30 p.m. of June the 11th. 1976
II. Noise and vibration

G. Shock and vibration
ON A DIRECT METHOD FOR ANALYZING THE FORCED VIBRATIONS OF CONTINUOUS SYSTEMS HAVING DAMPING

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Ohio State University 
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INTRODUCTION

The classical procedure of analyzing the forced vibrations of a continuous system having damping is well known—one expresses the forced motion in terms of the eigenfunctions (or mode shapes) of free vibration, and determines the amplitude and time response of each mode. However, exact eigensolutions are seldom known and, if known, the classical procedure is usually a long and laborious one.

The Galerkin method is a well known approximate method for solving static equilibrium and eigenvalue problems. It has also been occasionally used for the solution of forced vibration problems without damping (cf., (1)). The present paper shows how the method can be used directly (without first determining the free vibration eigenfunctions) to solve forced vibration problems with damping.

ANALYSIS

Numerous classical equations of motion for continuous systems can be written in the form

\[ \mathcal{L}(w) - \rho \frac{\partial^2 w}{\partial t^2} - c \frac{\partial w}{\partial t} + f = 0 \]  

(1)

where \( \mathcal{L}(w) \) is a differential operator of even order depending upon the continuum being analyzed (e.g., taut string, straight beam, isotropic membrane, orthotropic plate, circular cylindrical shell) and the remaining terms arise, in order, from inertia, damping and exciting forces. If the forcing function \( f \) is periodic in time it can be regarded as the superposition of sinusoidal functions, each having its own amplitude. Similarly, the displacement response \( w \) can be represented as a corresponding sum of modes, each having a corresponding sinusoidal time response, its own amplitude, and its own phase angle. Each mode is taken as the sum of a set of trial functions having undetermined amplitude coefficients and satisfying the boundary conditions of the problem. Substituting into eq. (1) yields a set of ordinary differential equations which, in general, the trial functions cannot exactly satisfy. The Galerkin procedure is used to best fit the trial functions to the differential equations.

NUMERICAL RESULTS

The method is demonstrated on several problems of damped, continuous systems subjected to periodic exciting forces including the one-dimensional problems of the taut string fixed at both ends and the simply supported beam, and the two-dimensional problem of a simply-supported plate. The accuracy of the method is compared with known exact solutions for these problems. It is demonstrated that, in some cases, the present method yields accurate solutions more easily than the classical, exact procedure.

REFERENCES

INTRODUCTION

Primary interest in bridging-wire realizations has centered around the need to reduce the number of resonators. The first designs were introduced about 15 years ago and involved simple bridging across two resonators for the purpose of realizing two attenuation poles at the \( j\omega \) axis. The distance between neighboring resonators was chosen one-quarter wavelength line.

In this paper, we designed one attenuation pole, below or above the passband through the use of wire bridging across one resonator. The distance between the adjoining resonators is \( \frac{\pi}{4} \).

THEORETICAL BASES

The basic section of an mechanical filters consists of three resonators Fig. 1. The pole of attenuation above the passband is realized by coupler \( K_1 > 0 \) and the pole below the passband is realized by coupler \( K_1 < 0 \) between the first and third resonator. From the short and open circuited impedances of half section, one can derive eq. (1), which serve for computing of the coupling coefficients.

\[
v = (f^2 - f_0^2) / f \cdot f_0 \quad \ldots (1)\]

For an equivalent LC bandpass Fig. 3 with edge frequency of passband at \( f_1 = 128 \text{kHz} \), \( f_2 = 131,2 \) and the frequency infinite loss at \( f_\infty = 133 \text{kHz} \) are the values of elements:

\[
\begin{align*}
K_1 &= 1,29 \times 10^{-2} \quad K_p &= 1,23 \times 10^{-2} \\
L_1 &= 1,71 \times 10^{-4} \quad L_p &= 2,61 \times 10^{-4} \\
C_1 &= 4,96 \times 10^{-7} \quad C_p &= 4,94 \times 10^{-7} \\
C_R &= 4,98 \times 10^{-7} \quad L_R &= 3,10 \times 10^{-6} \\
C_A &= 4,94 \times 10^{-7} \quad C_B &= 4,91 \times 10^{-7}
\end{align*}
\]

There are numerous other ways of realizing attenuation poles such as through use of ladder-arm poles and zeros as well as lattice structures. The realization by the coupling between non-adjacent resonators is very simple for manufacturing. In this way, more poles may be realized.

REFERENCES

INTRODUCTION

Determination of the amount of power flow through elastic mountings of machinery and equipment gives an indication of the structure-borne sound isolation effectiveness which seems to be the most appropriate from the physical point of view. It is shown here that such a determination can be achieved via certain measurement procedures by using a rather basic measuring equipment.

FORMULATION OF THE PRINCIPLE OF MEASUREMENT

The net power flow generated by axial vibration of the boundary surfaces of an elastic mounting can be evaluated to be:

\[ W = c \langle v_1 \xi_2 \rangle \]

where \( v_1 \) and \( \xi_2 \) are the vibration velocity and displacement on the opposite surfaces of the mounting, \( c \) is its axial stiffness, and \( \langle \rangle \) stands for time averaging. The variables \( v_1 \) and \( \xi_2 \) can be arbitrary time dependent functions.

Furthermore it can be shown that the wave effects occurring within the mounting reduce the accuracy of measurements, underestimating the result by the factor \( \sin(kh)/kh \) in the regions far from the mounting resonances (\( k \)-wavenumber, \( h \)-thickness).

In the regions closer to its own resonances, vibrations of the mounting are controlled by damping which can be very high for common-type rubber elements (1). The measurement results obtained for such cases have to be corrected by some values which can be calculated analytically.

EXPERIMENTAL RESULTS AND CONCLUSIONS

Tests were performed under laboratory conditions in regions sufficiently below the first resonant frequency. Good agreement between the results obtained by direct power flow measurements and those obtained in the way described were observed.

It can be expected that the applicability of the basic method (i.e. without making corrections) will be satisfactory in the large number of cases where machinery subjected to low-frequency excitation forces energy into the surroundings via elastic mountings. In the frequency region around and above the first resonance, frequency dependent corrections are necessary for the appropriate interpretation of results. For such corrections to be made, more data about the mounting, apart from the value of stiffness, have to be known.

REFERENCES

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INTRODUCTION

When mounting vibration transducers on the ground, care must be taken so that they faithfully follow the ground motion [1,2,3]. The transducer mount must be properly coupled to the ground, should not load the ground so as to cause a resonance effect and should be small in size compared to one wave length. Ground motion averaging may take place if the transducer is mounted on a long stake driven into the ground. Ref [4] shows that the horizontal component of Rayleigh waves is down -10 dB at $\lambda/10$ below the surface. For transducers mounted on plates or disks, coupling with the ground will be important.

EXPERIMENTAL DATA

The vertical response of the ground mountings shown in Fig. 1 was compared for sine wave inputs on grass-covered loam. The measurement results, all compared to Mount A, are also shown in Fig. 1. Mount A compared to itself (A/A) gave reasonable results out to 160 Hz. Differences in the responses of the two identical mounts may be due to propagation path anomalies as well as coupling effects. Similarly, Mount C showed remarkable agreement with A for frequencies up to 160 Hz. Mount B, whose theoretical response was calculated after Bycroft [5], was particularly troublesome, deviating substantially from Mount A (B1/A). The problem was due to root material under the disk which provided uneven support and substantially reduced the ground stiffness. Impressing the disk into the ground greatly improved its response (B2/A); however, a serious problem still existed above 100 Hz.

CONCLUSION

These short experiments show that serious transducer mounting problems may exist due to poor coupling to the ground. The problem is accentuated when vibration above 100 Hz is of interest. Other experiments are needed to determine standard mounting procedures.

The author thanks Mr. F.A. Prahl and Dr. L.E. Wittig, both of BBN, for their assistance with the vibration measurements.

REFERENCES

ON EFFECTIVENESS OF FORCE AND DISPLACEMENT VIBRATION ISOLATION IN MECHANICAL SYSTEMS

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INTRODUCTION

One of the possibilities of the vibration decreasing in mechanical systems is using so called vibration isolators. Putting the isolator between the primary systems namely source of vibration and vibration receiver, one can obtain the new system called briefly "source-isolator-receiver" system.

THEORETICAL CONSIDERATION

In our consideration every subsystem is characterized by its so called receptance [1]. That means, in case of force isolation problem, in order to describe the whole system dynamical properties one can need: the direct receptance of the source \( \beta_{ii} \), the direct end cross receptance of the isolator \( \beta_{i0} \), and \( \beta_{0i} \); and finally the direct receptance of a foundation \( \beta_{00} \). In case of displacement isolation and isolator symmetry assumption \( \beta_{i0} = \beta_{0i} \) it is possible to reverse subsystem properties, that means for source - \( \beta_{00} \) and receiver - \( \beta_{ii} \). Now, if we define force and displacement isolation effectiveness \( E_F \) and \( E_Z \), as forces or displacements amplitudes quantient, between those without isolator to the same quantity but with isolator system, one can obtain the common vibration isolation effectiveness measure in the form:

\[
E_F = E_Z = E = \frac{(\beta_{ii} + \beta_{00})(\beta_{ii} + \beta_{00}) - \beta_{00}^2}{(\beta_{ii} + \beta_{00})\beta_{00}}
\]  

(1)

For the massless isolator, or the general type but low frequency band, it is easy to prove that for \( E \gg 1 \), must be \( \beta_{ii} \gg |\beta_{ii} + \beta_{00}| \), where \( k \) is the reduced stiffness of the isolator. The same result but in terms of mobilities was obtained by Ungarn [2]. That means receptance of the isolator \( \beta_{ii} \) must be greater than modulus of a receiver and foundation receptances summ. But if we take the previous result as a generalized assumption in the forms \( \beta_{ii} \ll 1 \) and \( \beta_{00} \ll 1 \), one can obtain \( E \gg 1 \) only if

\[
\left| \frac{\beta_{ii} + \beta_{00}}{\beta_{00}} \right| \gg |\beta_{ii} + \beta_{00}|.
\]

(2)

This means in turn that for general case isolator and frequency, the cross receptance of the isolator \( \beta_{i0} \) is of primary importance.

CONCLUSION

1/ For the symmetric isolator system the isolation effectiveness \( E \) is the same both for force and displacement isolation problems. 2/ For the low frequency band or massless isolator its receptance must be greater than modulus of a sum for receiver and source receptances. 3/ In the general case of isolation problem the isolator cross-receptance is of primary importance. It must be as low as possible.

REFERENCES

INTRODUCTION

As a consequence of mass unbalance, rotating machinery generate harmonic forces. In the case of small bench grinders, commonly used in any machine shop, the problem of mass unbalance is particularly serious since those mechanical systems cannot be balanced (the unbalancing varies with the uneven wear of the grinding elements). When the generated harmonic forces are transmitted to the machine support structure and the rest of the environment the resulting vibration and noise cause physical discomfort to human beings and malfunction and even failure, of other mechanical systems located in the dynamic environment (1).

This paper describes an elastic mount for such a mechanical system, which can be attached to the work bench or to the wall (see figure 1).

RESULTS AND CONCLUSIONS

Since the motions of the system are six (one vertical and two horizontal translations and three rotational motions) the principal design criterion consists in developing a mounting system which will provide very low fundamental frequency values in those four possible modes of vibration.

The elastic mount described herewith is quite simple, efficient and economic.

Flexible cantilever bars attached to the wall.

Fig. 1: Elastic mount

REFERENCES

VIBRATION ANALYSIS OF STRUCTURAL-AcouSTIC SYSTEMS USING FINITE ELEMENTS

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INTRODUCTION

Current interest in low-frequency vibrations (0 to 200 Hz) of structural-acoustic systems includes the analysis of the interior noise environment of automobiles and aircraft [1-5]. Finite element analysis utilizing a modal approach is attractive for these systems of complicated geometry where the acoustic wavelength is the same order of magnitude as the cavity dimensions.

THEORETICAL BASIS

The equations of free motion for a structural-acoustic system are [6]

\[
\begin{bmatrix}
M_{ss} & 0 \\
M_{fs} & M_{ff}
\end{bmatrix}
\begin{bmatrix}
u \\
p
\end{bmatrix}
+ \begin{bmatrix}
K_{ss} & K_{sf} \\
0 & K_{ff}
\end{bmatrix}
\begin{bmatrix}
u \\
p
\end{bmatrix}
= \begin{bmatrix}
0 \\
0
\end{bmatrix}
\]

(1)

where \( u \) represents structural displacements, \( p \) represents fluid pressures, subscript 'ss' refers to structural properties, and 'ff' refers to fluid properties. Fluid-structural coupling is accounted for by the off-diagonal terms in this equation. External forcing terms can be added in the usual way. For analyzing cavity acoustics, various simplifications of eq. (1) may be of interest. These are (a) \( p = 0 \) for purely structural behavior and determining in-vacuo structural modes, (b) \( u = 0 \) for purely acoustic behavior and determining rigid wall cavity modes, and (c) prescribing \( f_i \) to determine the forced acoustic response to known structural vibrations. The full set of equations is solved if there is strong fluid-structure coupling.

APPLICATIONS

The finite element capability of the NASTRAN (NASA Structural Analysis) computer program has been used to model and analyze 2- and 3-dimensional structural-acoustic systems including those with multiple, coupled cavities [7]. If there is strong interaction between structural and acoustic modes, computational savings may be achieved by using a modal synthesis approach (Wolf, to be published). Applications to the automobile include (a) computation of acoustic modes and frequencies of the passenger compartment and (b) computation of the acoustic response to harmonic, random, and transient excitation of the vehicle structure. Laboratory tests of a vehicle were used to determine the accuracy of the analysis (Nefske, to be published). Figure 1 shows a comparison between the predicted compartment sound pressure response and the measured noise for harmonic excitation of the surrounding vehicle structure.

REFERENCES

Pour réduire le bruit dans les industries, il faut auparavant chercher d'intervenir sur les sources principales du bruit. Seulement si cela n'est pas possible, il faut intervenir sur le milieu, ou dans le cas dé precable que cela ne suffise pas, sur l'auditeur.

Les systèmes principaux de source et de transmission sont les structures en particulier celles ayant des grands parois.

Pour pouvoir donc intervenir en manière satisfaisante, il faut connaître les caractéristiques dynamiques, c'est à dire la valeur des fréquences de résonance, de la rigidité statique et des amortissements, ainsi que des fréquences perturbatrices.

A ce but il faudrait analyser le comportement dynamique de la structure d'une machine pendant le fonctionnement et cela dans toutes les configurations prévues et possibles: mais tout cela signifie que la valeur de l'excitation est indéterminée et le temps relatif nécessaire devient vraiment long. Au but donc de garantir majeure fiabilité à les données et pour réduire le coût des essais, il est nécessaire avoir recours à l'excitation artificielle, ou les données d'entrée sont bien connues. Mais il faut distinguer aussi entre les types de ces essais artificiaux, qui peuvent être très différents spécialement pour ce qui concerne le type du signal d'entrée (périodique, aléatoire, choc).

Jusqu'à présent on a utilisé le signal périodique, plus facile à obtenir, en variant la fréquence entre une gamme choisie à plaisir.*

L'analyse du signal de réponse de la structure était faite avec des instruments d'origine analogique.

Mais dans les dernières années les appareillages qui utilisent des techniques digitales, ont présenté un très important développement, en donnant la possibilité d'utiliser aussi des méthodes de calcul mathématique, pas possibles jusqu'à présent.

La nouvelle méthode d'analyser les signaux de réponse dans le domaine de la fréquence, reduit considérablement le temps du calcul.

Aujourd'hui dans les laboratoires de recherche on connaît bien l'utilisation de ces systèmes avec signaux d'excitation artificielle aléatoire et impulsionnelle.

Au cours de leur utilisation normale, les structures des machines sont soumises à une excitation transitoire ou choc, donc une excitation de ce type et la relative analyse, sont très utiles et il faut les rendre les plus possibles économiques et rapides, du point de vue pratique.

Il est donc nécessaire vériifier les résultats de plusieurs applications et déterminer les critères des tests d'environnement qui de vront être satisfaits.

En particulier il faut développer la recherche de la valeur des paramètres d'excitation nécessaire pour garantir les mêmes résultats, en utilisant différents signaux d'entrée.

Dans ce but chez le Laboratoire CEMU on est en train d'étudier en particulier l'excitation impulsionnelle qui est la plus simple à obtenir, aussi du point de vue économique.
INTRODUCTION

In a number of industrial situations a rapid test to screen components for defects - particularly cracks - would be valuable. For this reason the response of a test piece to a short duration mechanical impact has been examined for information related to mechanical integrity.

BASIS OF TECHNIQUE

A sample's impact response can be analysed into a series of decaying sinusoidal waveforms. The frequencies correspond to sample modal frequencies and the damping is due to the loss mechanisms for each mode. A crack changes these response frequencies and their damping so these quantities are used as test parameters (1,2).

EXPERIMENTAL TECHNIQUES

Aluminium alloy test samples 150mm x 25mm x 3mm were constructed with defects in a plane 15mm from one end. One set contained cuts of various depths 0.79mm wide, while others contained fatigue cracks grown from a 1mm deep notch. These samples were supported on fine thread and struck on an end face. The responses were picked up with a B & K ½ inch microphone and analysed on a small computer system.

RESULTS

Particularly interesting results were obtained for a flexural mode of vibration with large amplitude at the defect (at 17.35kHz for a sample without defect).

Similar modal frequency changes were found for samples with cuts and cracks and the changes were most sensitive to large defects. Conversely damping changes were most sensitive to small surface opening defects. Again changes for cuts and cracks followed the same trend but the presence of the notch from which the cracks were grown dominated (2).

CONCLUSIONS

Experiments have shown that the impact response is sensitive to the presence, position and size of a defect, and could form the basis of a useful testing technique.

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MODELLING OF DYNAMIC STRUCTURES USING COMPLEX NORMAL MODES

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INTRODUCTION

Identification of mathematical models from experimental data to represent the dynamic behaviour of structures or systems has been widely investigated in recent years (1,2). A favoured approach is the use of lumped-element models together with a normal mode method of identification. This method can be simplified further if the structure being modelled is assumed to have uniform damping so that the mode shapes (eigenvectors) are purely real and the modal responses effectively uncoupled.

However, the presence of non-uniform damping implies that the modes are both complex and coupled (3,4). This paper describes a method whereby a more general model is set up using complex normal modes without the restrictive assumptions mentioned above.

THEORETICAL BASES

For a discrete system of \( N \) degrees of freedom with arbitrary hysteretic damping, the receptance at a point \( j \) due to a single harmonic excitation of frequency \( \omega \) applied at a point \( k \) can be expressed as

\[
R_{jk}(\omega) = \sum_{r=1}^{N} \psi_{rj} \psi_{rk} \frac{m_r}{m_r^2 (\omega^2 - \omega_r^2) + 2i \omega \omega_r (\omega^2 - \omega_r^2 - \omega_r^2)}
\]

where \( \psi_{ij} \) are elements of the complex modal vectors, \( m_r \) is a complex modal mass, \( \eta_r \) is a real modal loss factor, and \( \omega_r \) is a natural frequency. Graphical curve-fitting techniques can be used to calculate these parameters by analysing each mode separately and allowing for off-resonant contributions. If the modal masses are fixed at unity, the identified real mass, stiffness and damping matrices follow thus:

\[
\]

\[
[K] + 1[H] = [\psi]^T \left[ \omega_r^2 (1 + i \eta_r) \right] [\psi]^{-1}
\]

EXPERIMENTAL TECHNIQUES

An interactive computer program has been written to perform the analysis. Its validity has been checked initially using simulated data from a lumped element model. Experimental data from a simple beam structure has been analysed. By adding localised damping treatments the effective loss factor of the beam was varied over a series of experiments.

RESULTS AND CONCLUSIONS

The results from the experiments although encouraging illustrated the shortcomings of the method. In particular the damping matrix was found to be susceptible to error, mainly as a result of small but significant inaccuracies in the identified eigenvectors. The identified mass and stiffness matrices, however, were consistently accurate.

REFERENCES

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300
INTRODUCTION

A structure's response maximum and energy are often critical parameters in predicting likely damage by transient loads. The history of the response and, hence, gross characterisations may be evaluated explicitly if the properties of the load and structure are known. In practice, however, the form of any one loading (e.g., earthquake, sonic boom, wind gusts) is seldom unique and the structure is subjected over its life to different combinations of loads. Such considerations led a number of investigators (e.g., 1, 2) to determine the excitation amongst all possible loads which produced the greatest maximum (maximax) response. Unfortunately, this maximax will often be highly inaccurate, especially for the usual situation of poorly damped structures, because the corresponding "worst" excitation is physically unrealizable (2). Another, equally significant problem related to practical applications is that the theory implicitly requires all the structure's properties to be known. An alternative theory will be proposed and developed which usually will produce better estimates and certainly will require less knowledge of the structure.

THEORETICAL BASES

Instrumentation limitations suggest that a structure's response may reasonably be assumed limited to band \( \pi g \) in the frequency domain. It will be shown that the maximax displacement then may be written as:

\[
\text{maximax } g(t) = \sqrt{E_g} \text{max } \psi_0 \left( t + \frac{T_g}{2}, \Omega_g T_g/2 \right)
\]

where \( E_g \) and \( T_g \) are the energy (1) and duration, respectively, of the deflection, \( g(t) \). \( \psi_0 \left( t, \Omega_g T_g/2 \right) \) is the prolate spheroidal wave function defined in reference (3). The structure's maximax velocity, acceleration or jerk may be derived similarly. \( E_g, T_g \) and \( \Omega_g \) can be calculated in terms of their more easily measured load counterparts but additional limited information is still needed in the form of structural damping.

COMPUTATIONS

Results have been computed digitally for a simple vibrating oscillator and will be presented in graphical format. The better accuracy will be shown by comparisons with sample calculations obtained more conventionally from response spectra. Extensions to multi-degree of freedom systems will be outlined.

CONCLUSIONS

An improved, more practically orientated technique has been developed to determine the maximax response of structures to transient loads. The application of the theory will be demonstrated for fairly realistic situations.

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DYNAMIC BEHAVIOR OF STEEL AND NYLON ROPES SUBJECTED TO IMPACT LOADING

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INTRODUCTION

Cable systems are used in a multitude of engineering applications which range from suspended bridge structures to salvage recovery of large objects in the ocean to transmission lines. When dealing with the dynamics of cable systems one finds that the external forces acting on a cable are longitudinal (generated by pulling, viscous drag, etc.) and transverse (caused by interaction between cable and fluid medium, impact like in the case of an arresting gear in an aircraft-carrier, etc.).

Nylon cables are used in a multitude of applications because of lightweight and relatively high strength. They also possess high internal damping and capacity to store a large amount of strain energy in view of their high strength and deformability properties.

The beneficial effect of nylon cable complementing a mechanical steel rope system has been shown in [1] where a longitudinal impact situation was investigated. Dynamic stresses were mitigated significantly.

The present paper describes an experimental investigation which deals with steel and nylon cables subjected to transverse impact loads.

RESULTS

It is shown that under certain special conditions nylon cables behave considerably better than steel ropes when subjected to severe transverse impact excitation.

REFERENCES

EINFÜHRUNG

In Hamburg wurden an zwei S-Bahnen Brücken über die Hammerbrookstraße Schallschutzmaßnahmen durchgeführt. Zu diesem Zweck wurden nahezu alle abstrahlenden Flächen der Stahlhohlkastenbrücken (direkte Schienenauflage) mit Entdröhnmaterial und Konterblech versehen.

Zur Ermittlung der schalltechnischen Wirksamkeit wurden vor und nach der Beschichtung Luft- und Körperschallmessungen durchgeführt.

MESSUNGEN

Die Luftschallmessungen erfolgten unter Brücken und in 10, 25 und 50 m Abstand. Für die Körperschallmessungen wurden an jeder Brücke ca. 40 Meßpunkte verteilt auf drei Querschnitte angebracht.

Die Messungen wurden bei Vorbeifahrt einer S-Bahn mit 40 km/h durchgeführt.

ERGEBNISSE

Die Untersuchungen ergaben folgendes:

a) der mittlere Körperschallschnellepegel verringerte sich um 18 dB (Pegeldifferenz zwischen der Messung bei unbeischichteter Brücke und nach dem Anbringen des Entdröhnmaterials)


c) Der Vergleich der Messungen an den Brücken mit Ergebnissen an der freien Strecke (Meßabstand 10 m zeigt, daß nach der Beschichtung die Brücken der freien Strecke schalltechnisch gleichzusetzen sind (mit Ausnahme der tiefen Frequenzen bis ca. 200 Hz.

Abb. 1: Schalldruckpegel vor und nach der Beschichtung Brücke 2, Abstand 25 m
INTRODUCTION

Design engineers are frequently confronted with the problem of mounting different types of engines and motors on structural elements. In order to avoid dangerous resonance situations, the designer must be able to predict the natural frequencies of the complete mechanical system: structure-motor and its elastic mounting. Ultimately, he should also determine mode shapes and dynamic stresses induced by any dynamic disturbance generated by the exciting system (Fig. 1).

Fig. 1: Simply supported beam of variable cross section and moment of inertia which supports an elastically mounted mass.

APPROXIMATE SOLUTION

A rather general approximate method of solution is developed in the present study. Variable cross section and moment of inertia, translational and rotational flexibilities at the supports are taken into account.

It is quite convenient to express the displacement amplitude in the form:

$$W = 2\sum_{j=0}^{1} A_j \left( \alpha_j x^4 + \beta_j x^3 + \delta_j x^2 + \delta_j x + 1 \right) x^4$$

where the $\alpha_j$, $\beta_j$, etc. are obtained substituting each coordinate function in the governing boundary conditions.

RESULTS AND CONCLUSIONS

Galerkin's method is used to obtain the natural frequencies of the coupled system in the free vibrations case and the coefficients of the coordinate functions when the system is subjected to an external periodic excitation.

The experimental results compare favourably well with the mathematical predictions.

Das Verfahren beruht auf der Beziehung /1/

$$\frac{\alpha}{\beta} = \frac{\frac{2\pi}{k} f c}{4 \pi}$$

wobei $k$ die Wellenlänge der Luft, $f$ ihre Dichte und $c$ die Schallgeschwindigkeit bedeuten. Der Parameter $\alpha$ gibt an, wie groß die abgestrahlte Schalleistung $P$ ist, wenn eine Struktur durch die Wechselkraft $F$ angeregt wird ($P = \alpha F^2$). Der Parameter $\beta$ gibt an, wie groß die Schnelle $\vec{v}$ an der Stelle ist, an der die Kraft $F$ wirkt, wenn die Anregung durch ein statistisch verteiltes Schallfeld mit dem mittleren Druckquadrat $\bar{p}^2$ erfolgt ($\vec{v} = \beta \bar{p}^2$). Die Beziehung (1) wurde experimentell überprüft (siehe Bild 1), außerdem wurde mit Hilfe der leicht messbaren Größe $\alpha$ und $P$ die von haustechnischen Anlagen erzeugten Wechselkräfte bestimmt. Ein Beispiel zeigt Bild 2.

Die Vorteile eines solchen Meßverfahrens liegen auf der Hand:

1) In vorhandenen Bauten kann der geeignete Punkt zum Anbringen von Körperschallerregern gefunden und Aggregate nach maximalen Wechselkräften ausgewählt werden.

2) Über akustische Größen kann auf die punktförmige Wechselkraft von Erregern geschlossen werden.

Literatur:
INTRODUCTION

On établit les équations du mouvement de plaques multicouches en utilisant une formulation variationnelle à champ unique de déplacements et une technique de raccordement des champs aux interfaces, déjà utilisée par (1), que nous systématisons. Les équations des vibrations libres et les conditions aux limites naturelles sont formellement indépendantes du nombre de couches. La résolution est effectuée pour la plaque appuyée. Une comparaison théorie-expérience est enfin réalisée.

THEORIE

Le champ de déplacement, défini pour chaque couche, est du type (1). Aux interfaces nous raccordons les contraintes et les déplacements cf (2), (3)

\[
\begin{align*}
U &= \Psi^n + (R_n-\delta_2)\left(\begin{array}{c}
\Psi^n \\
\end{array}\right) \\
V &= \Psi^n + (R_n-\delta_2)\left(\begin{array}{c}
\Psi^n \\
\end{array}\right) \\
W &= W
\end{align*}
\]

\(R_n\) cote du plan médian de la couche n

La technique de raccordement des champs permet de relier les variables cinématiques de couches consécutives et ainsi par application successive du procédé, d'exprimer le champ dans toute la plaque avec les variables de la première couche. La fonctionnelle de Hamilton est construite puis minimisée; on obtient les conditions aux limites naturelles et l'équation de mouvement (4)

\[
\left(\begin{array}{c}
J \\
E
\end{array}\right) \{x\} = 0
\]

\(J\) et \(E\) sont des matrices d'opérateurs, \(\{x\}\) est le vecteur des variables cinématiques de la première couche. La forme de cette équation est indépendante du nombre de couches.

La résolution est effectuée pour la plaque appuyée; les formes propres gardent la forme simple de produit de fonctions sinusoidales. Une propriété importante apparaît: les modes peuvent se grouper par famille, C'est à dire qu'il existe plusieurs modes possédant le même nombre d'onde dans le plan de la plaque.

RESULTATS-CONCLUSION

L'étude numérique souligne deux types de comportement dynamique en fonction des caractéristiques mécaniques des couches: le 1er type est essentiellement dû à l'effet de flexion lorsque les constantes élastiques des différentes couches gardent des valeurs voisines; pour un grand nombre de couches, les fréquences propres tendent vers celles calculées avec la "loi des mélanges". Le deuxième type obtenu avec un empilement de couches dures et molles alternées, est dû au cisaillement des couches molles.

L'expérimentation a nécessaire la mesure préalable de caractéristiques mécaniques des matériaux utilisés. Trois dispositifs complémentaires ont été employés. La comparaison théorie-expérience s'avère correcte si l'on inclut dans le calcul les constantes élastiques des couches de liaison (collage).

INTRODUCTION

The analysis of transverse vibrations of membranes and plates is of fundamental interest in several fields of science and technology. Their existence is of paramount importance in musical instruments, hearing and practically all fields of engineering: transducer design and aeronautical, civil, mechanical and naval engineering applications.

Several applications make use of non-circular and non-rectangular shapes. One situation where non-conventional shapes are used takes place in printed-circuit board technology. Originally printed-circuit boards have been of rectangular shape supported around the edges either continuously or at a discrete set of points. However, increasing demands on electronics applications generate needs for more complicated shapes (1). Obviously, the first step towards avoiding resonant mechanical conditions which in general lead to malfunction of the system is determining the frequency at which the plate resonates.

However, exact solutions can only be obtained in a limited number of cases since they are limited to simple geometries. Since boundary configurations of interest are frequently more complicated it has been necessary to make use of approximate techniques to solve those problems.

The present paper deals with the determination of the fundamental frequency of vibration of plates of complicated boundary shape. Complicating conditions such as in-plane forces, concentrated masses, etc., are taken into account in the present formulation.

THEORETICAL BASES

It is shown that a unified and quite simple methodology for the solution of a rather large class of problems consist in making use of a combined conformal mapping-variational approach. The calculated eigenvalues seem to possess excellent accuracy and it is also possible to predict the fundamental mode shapes.

EXPERIMENTAL RESULTS

Several experiments on polygonal plates carrying concentrated masses have been conducted. The values of natural frequencies obtained experimentally are in reasonably good agreement with those predicted analytically.

REFERENCES

FREE AND FORCED VIBRATIONS OF PLATES OF VARIABLE THICKNESS

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INTRODUCTION

As stated by Leissa (1): "in the case of plates with variable thickness, the governing differential equation of motion is found to have variable coefficients, and this fact increases the difficulty of solution."

The goal of the investigation reported here was to develop a simple and yet sufficiently accurate technique for the dynamic analysis of rectangular and circular plates of variable thickness with edges elastically restrained against rotation and translation.

THEORETICAL BASES

This study may be considered as a sequel to earlier papers (2)-(5) in which a simple method for analyzing free and forced vibrations of rectangular and circular plates with edges elastically restrained against rotation was developed. Polynomial coordinate functions which identically satisfy the boundary conditions are used to approximate the plate response.

A variational method is then used to evaluate frequency coefficients in the free vibrations case and dynamic displacements and bending moments when the plate is subjected to a external \( p_0 e^{j\omega t} \) type excitation.

The same approach is used in the present paper.

A similar technique is followed in the case of stepped variation of thickness. The Rayleigh-Ritz method is used in this case.

RESULTS

The present approach is quite simple and straightforward. The calculated eigenvalues are, in some cases, more accurate than those considered as "exact" in the open literature. It is also shown that in the case of circular plates frequencies of higher, anti-symmetric modes can be calculated with a minimum amount of difficulty.

REFERENCES

DIE DRUCKVERTEILUNG AUF EINER SENKRECHT ANGESTROMTEN PLATTE UND IHR EINFLUSS AUF DEN ANGEREGTEN KÖRPERSCHALL

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Einleitung

Strömungsschallquellen besitzen im allgemeinen ein ausgeprägtes Nahfeld. Werden schwingungsfähige Strukturen in die Nähe solcher Quellen gebracht, so ist eine starke Anregung zu erwarten, die von der Frequenz- und Wellenzahlverteilung der anregenden Wechseldrücke abhängt.

Experiment

Es wurde eine Anordnung gewählt, bei der ein Luft-Freistrahl aus einer Düse von D = 6 cm in einer Entfernung von x = 10 D senkrecht auf eine Platte trifft. Die Platte war mit abdeckbaren Bohrungen versehen, so daß ohne Störung der vorhandenen Grenzschicht Druckmessungen an der überströmten Oberfläche vorgenommen werden konnten.

Zur Kennzeichnung der Strömungsquellen wurden ermittelt:
1. die normierten Druckspektren als Funktion der Strouhal-Zahl
   \( St = \frac{f \cdot D}{U} \)
2. die Kohärenzfunktion des Druckes in radialer und meridionaler Richtung
   \[ \gamma(St, r, r', \varphi, \varphi') = \frac{\phi_{r2}(St, r, r', \varphi')}{\phi_r(St, r, \varphi) \cdot \phi_{r2}(St, r', \varphi')} \]
   und hieraus die Kohärenzlängen
   \[ l_c(St, r) = \int \gamma(St, r, r', \varphi' = 0) \, dr' \]
   \[ \varphi_c(St, r) = \int \gamma(St, r, r' = 0, \varphi') \, d\varphi' \]
3. die Konvektionsgeschwindigkeit \( U_c \)
   Für verschiedene Platten wurden außerdem Körperschallmessungen durchgeführt.

Ergebnisse:

In dem für die Anregung maßgeblichen Plattenbereiche ergaben sich relativ große Kohärenzlängen, die etwa proportional dem Abstand \( r \) vom angeströmten Zentrum zunahmen. Entsprechend nahm die Konvektionsgeschwindigkeit \( U_c \) mit \( r \) ab. Die größte Kohärenzlänge war bei der Frequenz anzutreffen, wo auch das Druckspektrum sein spektrales Maximum hatte.

Es wird der Versuch unternommen, aus den gemessenen Daten ein theoretisches Modell zu konstruieren, das die gemessenen Körperschallwerte erklärt.
**INTRODUCTION**

L'emploi de matériaux viscoélastiques pour réduire les vibrations des tôles est assez ancien : les revêtements utilisés sont de simple couche ou de type dit "sandwich", c'est-à-dire une couche de matériau viscoélastique intercalée entre deux plaques élastiques. En utilisant les bases de la mécanique des milieux continus, certains auteurs ont pu étudier le phénomène d'un point de vue fondamental. Parmi ceux-ci nous citerons uniquement Di Taranto et J.R. Mc.Graw [1]. Cependant, le problème d'un traitement local de tôles ne semblait pas avoir été étudié du point de vue théorique et ceci fait l'objet du présent travail.

**ÉTUDE THÉORIQUE**

L'équilibre des forces et des moments appliqués à une plaque "sandwich" conduit à une équation du mouvement transversale $W$ du sixième ordre :

$$ A \Delta \Delta \Delta \Delta W + \left( B_1 + B_2 \right) \Delta \Delta W + C \omega^2 \Delta W - \rho \omega^2 W = 0 $$

(1)

$A$, $B_1$, $B_2$, $C$ sont des coefficients qui dépendent des épaisseurs des trois couches, des modules de Young des deux couches élastiques, ainsi que du coefficient de Poisson, du module de cisaillement $G$ et du coefficient d'amortissement $\beta$ de la couche viscoélastique. $A$, $B_1$, $C$ sont nuls lorsque la plaque n'est pas traitée, et l'on reconnaît bien l'équation classique des plaques nues : $B_2 \Delta \Delta W - \rho \omega^2 W = 0$.

Dans le cas d'une plaque partiellement traitée (plaque supérieure et matériau amortissant plus petits que la plaque inférieure), ces coefficients doivent comporter une discontinuité telle que les coefficients réels soient nuls pour la plaque non traitée, et égaux à $A$, $B_1$, $C$ pour la plaque entièrement traitée. Si nous représentons cette discontinuité par

$$ R_0 = \sum_{m,n} \sin \frac{mk}{2} \sin \frac{nk}{2} \sin \frac{mk}{2} e^{-\frac{im}{e}} e^{-\frac{in}{e}} $$

où $k = \frac{x}{e}$ rapport du côté de la partie traitée à celui de la plaque entière, nous reconnaissons cette hypothèse. L'équation (1) devient donc une équation à coefficients (trois) périodiques. Le choix d'une solution périodique conduit à des conditions d'identification faisant intervenir neuf constantes arbitraires qui forment un système linéaire dont le déterminant est nul. Ceci donne une équation algébrique complexe. L'amortissement global $\eta$ obtenu est donné par le rapport de la quantité imaginaire à la quantité réelle de la pulsation.

**ÉTUDE EXPERIMENTALE**

L'étude expérimentale est effectuée sur des plaques carrées d'acier de $20 \times 20 \times 0,3\text{cm}^3$ enduites d'"aquaplasse" et recouvertes d'une autre plaque de dimensions variables. On détermine le coefficient d'amortissement global par la méthode de résonance. La variation de ce coefficient avec les dimensions de la partie traitée de la plaque n'est pas linéaire comme le montraient les équations.

**REFERENCES**

INTRODUCTION - Orthogonally stiffened plates have served as the basis of many structural components. Because of this one can find in the literature both static and dynamic analyses of orthotropic plates. A number of these analyses use the concept of an equivalent plate with uniform thickness and orthotropic material properties. The purpose of this paper is to develop the procedure for analyzing orthogonally stiffened plates by considering directly the geometrical properties of the plate.

THEORETICAL BASIS - A set of governing equations with the associated natural boundary conditions are derived using the generalized Hamilton's principle. The stress resultants and the displacement relations are then established through the use of Reissner's stress potential in the definition of the potential energy of the system. In this procedure, one does not need to make any hypothetic assumption regarding the orthotropic nature in defining the stress-strain relationship. In the development of the framework the classical thin plate theory has been extended to incorporate the effects of transverse shear and rotatory inertia (Mindlin's plate theory) for the orthogonally stiffened plate. Three independent displacement functions are required to describe the deformation of the plate. Equations of motion have been developed considering the coupling effects of the initial membrane forces and the transverse bending on the dynamic characteristics of the plate. Solutions are obtained for an initially stressed orthotropic plate simply-supported to a transverse impulse pressure (or shock loading) which is a function of time and position. The natural frequencies and the modes of free vibration are properly calculated. The uniqueness of the modal responses has been established by a normalization process. The problem then is reduced to solving for the modal time-dependent coefficients as defined by a second order differential equation of a single degree of freedom system. Solutions are readily obtainable by the Duhamel integral.

RESULTS - The preceding formulation was applied to a rectangular plate composed of a series of tubes joined by a membrane that forms the furnace wall of a utility fossil steam generator. The inplane buckling loads and natural frequencies were determined for a simply-supported boundary condition. The results showed a noticeable influence of the inplane membrane forces on the frequency corresponding to the transverse displacement modes but little influence to the rotatory modes. In addition, plate responses were plotted for a pressure load having an exponential time decay with respect to different values of inplane membrane force. The results can be used for the optimization of buckstay spacing (distances between supports of a steam generator wall) as a function of the initial inplane force.
FREE VIBRATION OF ANISOTROPIC CIRCULAR PLATE SUBJECTED TO VARIOUS BOUNDARY CONDITIONS

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INTRODUCTION

A theory is presented for the determination of the free vibration characteristics of non-uniform, anisotropic circular plate. The formulation is based on a recently developed method for the analysis of cylindrical shells (1-2). It is a hybrid of the finite element and classical plate theories. The "finite-element" chosen is a circular frustum and the displacement functions are derived from the exact solution of the circular plate equations.

THEORETICAL ANALYSIS

The finite element is defined by two nodes, i and j, and nodal surface boundaries; the displacement functions are defined by

\[ \begin{bmatrix} U(r,\theta), W(r,\theta), V(r,\theta) \end{bmatrix}^T = [N] \begin{bmatrix} \delta_i, \delta_j \end{bmatrix}^T \]  \hspace{1cm} (1)

where \(U, V, W\) are the axial, circumferential and radial displacements, respectively, and \(\delta_i, \delta_j\) and nodal displacements at \(i\) and \(j\) associated with the \(n\)th circumferential wave number.

The strain and stress vectors are written as

\[ \varepsilon = [B] \begin{bmatrix} \delta_i \\ \delta_j \end{bmatrix}, \sigma = [P] \varepsilon \]  \hspace{1cm} (2)

where \([P]\), the elasticity matrix, characterizes the plate's anisotropy. To determine the elements of \([N]\) and \([B]\), Sander's equations are solved (3) and the displacement functions of thin plate are explicitly derived in terms of the nodal displacements \(\delta_i\) and \(\delta_j\).

With \([N]\) and \([B]\) determined functions of \(n, \theta, r\) and the elements of \([P]\), the general terms of the stiffness and mass matrices for one finite element were analytically obtained after lengthy manipulations, and hence the global mass and stiffness matrices assembled.

RESULTS AND CONCLUSIONS

Given the dimensions and properties of the plate, a computer program calculates the eigenvalues and eigenvectors of anisotropic plate for various boundary conditions.

The theory was tested by calculating the natural frequencies of a free plate, and it was found that the lowest two frequencies were essentially zero (rigid-body motions), and the corresponding strains vanished. Other calculations were conducted determining the natural frequencies of simply supported and clamped non-uniform plates. The results were in excellent agreement with those from other theories and with experiments; often being in better agreement with experiments than other theories. The effort involved in producing such complex theory is deemed to be justified. In this connection, it is noted that accurate knowledge of some of the high and the low frequencies is essential for the accurate determination of the response of plates to random pressure field, such as those generated by a flowing fluid.

REFERENCES

VERSCHIEBUNGS- UND SPANNUNGSÜBERTRAGUNG AN PLANPARALLEL-LEN ELASTISCHEN PLATTEN

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Die Überlegungen gelten für beliebige Plattendicken, sie sind nicht auf den Fall dünner Platten beschränkt. Auch werden keine Voraussetzungen über die Schwere der Platten (im Verhältnis zu den angekoppelten Medien) gemacht.

Der Übertragungsformalismus eignet sich ferner für die Berechnung der Schalldurchlässigenenschaften schichtartig strukturiertn Platten (Mehrfachwände, Sandwich-Konstruktionen); Dämpfungser- scheinungen werden vermöge der Verwendung komplexer Wellengeschwindigkeiten in dem Plattenmedium berücksichtigt.

Literatur
INTRODUCTION

Free and forced vibrations of shallow spherical shells have been studied by many researchers. The present paper deals with the axi-symmetric vibration of shallow shells supported along the outer periphery \( r = a \) and along an intermediate circle of radius \( r = b \). Shear deformation effects have been included in the differential equations. This makes the analysis of higher modes more meaningful.

THEORY

It is found that the differential equations of motion can be reduced to a simple form in terms of two deformation functions \( \theta \) and \( \psi \) which are defined as

\[
\theta = \frac{U_x}{x}, \quad \psi = \frac{V_x}{x}
\]

where the deformation \( U \) and rotation of the normal \( V \) are functions of the non-dimensional radial distance \( x \). The solution of the equations is generated and it is shown in the paper that it can be written as

\[
W = \sum_{\alpha=1}^{3} \phi_\alpha, \quad \theta = \sum_{\alpha=1}^{3} \eta_\alpha \phi_\alpha, \quad \psi = \sum_{\alpha=1}^{3} \Lambda_\alpha \phi_\alpha
\]

In the above equations the function \( \phi_\alpha \) is given by

\[
\phi_\alpha = A_\alpha J_0 (\mu_\alpha x) + B_\alpha Y_0 (\mu_\alpha x).
\]

The coefficients \( \eta_\alpha \) and \( \Lambda_\alpha \) are functions of frequency \( \omega \), the parameter \( \mu \), material and geometric properties of the shell. The parameters \( \mu_\alpha \) are to be determined from the characteristic equation.

COMPUTATIONS AND RESULTS

The paper shows results for three cases of shells: (a) Fixed edge condition at \( r = a \) (b) Simple supported condition at \( r = a \) (c) Free edge condition at \( r = a \). The results are generated for various \( b/a \) ratios by changing the location of the intermediate support. The paper also shows modal shapes for the three cases mentioned above. Results for some limiting cases can be worked out by setting \( \frac{b}{a} \rightarrow 0 \) or \( \frac{b}{a} \rightarrow 1 \).
THE FORCED VIBRATION OF A DAMPED CIRCULAR RING SEGMENT

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The use of laminated, three-layer beams or plates consisting of a layer of viscoelastic material sandwiched between two elastic layers is a common procedure in the design of highly damped structures. Design application of laminated flat structures is assisted by a well-developed theory and close agreement with experiment. The ability to design damped laminated curved structures is much less satisfactory.

A theory for the forced vibration of laminated complete circular rings has been presented by several authors, e.g. Ref. [1], and, based on these studies, a set of guidelines has recently been prepared (Nelson and Sullivan, to be published). The purpose of the present study is to extend the theory to the case of a circular ring segment with simply supported ends. The equations of motion are derived by a method similar to that used in Ref. [2] for flat beams. The equations are solved by the use of damped forced modes. The results are shown in Fig. 2 for the ring geometry of Fig. 1 with \( \alpha = 2\pi \).

Figure 2 also shows the results of a preliminary test on a ring segment with the geometry and material properties of Fig. 1. It is expected that better bonding between layers and adjustment of material properties for variation with frequency and temperature will improve the agreement between test and theory.

REFERENCES

INTRODUCTION

In many aircraft configurations, parts of the structure are subjected to high intensity random pressures arising from the acoustic environment of the jet engines. Pseudo-acoustic loads on the structure are also created by the turbulent boundary layer, regions of separated flow, or impinging flows. The box-like structures of stabilisers and control surfaces are excited into high frequency random vibration. This can cause fatigue damage of the internal structure. The purpose of this paper is to discuss the vibration of the fatigue prone internal support structure.

THEORY

The complete theory based on the normal mode approach can be used to describe qualitatively the behaviour of the structure. Simplified design procedures are available for the external plate like structure. However there is no adequate design method for the internal structure. A semi-empirical method is proposed and the predictions compared with some experimental results.

EXPERIMENT

A test structure Figure 1 was exposed to random acoustic pressures in a high intensity test facility. The box structure could be modified by changing the internal ribs. In this way the effect of design of the internal structure was investigated.

RESULTS AND CONCLUSIONS

The broadband acoustic loading Figure 2, caused a narrow band multi-modal response of the internal structure, Figure 3. It is not possible to interpret the fine detail of this response spectrum. However the major trends can be explained in terms of the characteristics of the internal plate itself - rather than in terms of the complete coupled structure. This allows a simplified model to be proposed and checked against the experimental results for the r.m.s. strain. Comparison is also made with results from typical aircraft structures. The overall agreement is of the order of + 6 dB.

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SOUND INTENSITY PATTERNS FOR VIBRATING SURFACES

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ABSTRACT

The spatial characteristics of the acoustic power flow at the surface of a vibrating body are studied to obtain an insight on the process of energy exchange between the structure and the acoustic medium, and to explore the possibilities of using surface intensity measurements as a means of source identification, ranking and quantification in research on machinery noise. Rayleigh's formula is used to compute the normal component of the acoustic intensity at the surface of a vibrating boundary, giving an expression for the local radiation efficiency as a function of position and frequency. This expression is used to compute sound intensity patterns at the vibrating surface, to show which parts of the structure are mostly responsible for the radiated noise, and to give a picture of the spatial distribution of energy exchange. A computer simulation is developed to study the intensity field at the surface of a circular piston radiator, and to plot the contours of equal radiation efficiency at the surface of a simply supported rectangular plate source under single mode and multimodal excitation. See Fig.V-8, [1].

An experimental procedure is presented to measure the normal component of the acoustic intensity at a point on a vibrating surface, from local measurements of acceleration and pressure. The cross-power spectral density of the acceleration-pressure joint process is used to estimate the local intensity, and the coherence function is used to monitor measurement quality. Analog and digital implementations of this technique are described, with an analysis of the expected accuracy. Experimental versions of a circular piston source and of a simply supported rectangular plate source were built and measured simultaneously with the surface intensity technique and with the reverberant room method. Local intensity measurements are compared with the theoretical solution, and space average values are compared with the reverberant room measurements. Good agreement is found in all of this comparisons.

The intensity measurement technique is then applied to the problem of identifying, ranking and quantifying the noise sources on a sewing machine with good results. A sampling strategy based on a Monte Carlo method is suggested to determine source sound power level and its variance, from surface intensity measurements on a few points on the source. The author's recommendation, based on this work, is that the surface intensity measurement be considered by researchers in machinery noise control.

INTRODUCTION

The fringes which appear on the reconstructed image of a vibrating object recorded using time-averaged holography are well known but their use for describing the surface motion of vibrating objects has had limited success until the present work. Interpretation is complicated by many factors which include the directions of illumination, observation, motion of the observed surface, and the time-averaging process itself. In general the motion of a surface may be expected to have components in three dimensions, especially if the surface is a thin curved shell. However, a single time-averaged hologram can only provide information about displacement in a single direction that is along the sensitivity vector defined by the directions of illumination and observation. Thus it is apparent that to define the motion of a vibrating surface uniquely, at least as many independent views of the surface are required as degrees of freedom of the surface motion and careful consideration must also be given to all other factors which affect the fringe formation.

The path of a point which vibrates harmonically in three dimensions simultaneously is an ellipse whose description requires the determination of five constants. Thus to determine the path of a point on a vibrating surface undergoing elliptic motion requires a minimum of five separate time-averaged holograms. If internal dissipation is small, however, the generalized ellipse reduces to rectilinear motion and the number of constants and required time-averaged holograms is reduced to three.

If the motion to be investigated has been previously described and experimental verification or perhaps the measurement of relative amplitudes in some plane is sufficient, then the required number of separate views may be still further reduced to two.

EXPERIMENTAL WORK

The motion of several cylinders has been investigated. For this purpose the cylinder is mounted with its axis coincident with the axis of rotation of a turntable. A self-excited feedback loop system is used to drive the cylinder in one of its resonant modes. A sequence of time-averaged holograms is then taken at regular intervals around the cylinder simply by turning the cylinder between exposures.

The holograms and their fringe systems are entered into a computer using a digital data tablet. A least squares best fit routine is then used to determine the components of motion at each point on the cylinder. In addition, confidence limits are readily computed and displayed in the final print-out.

RESULTS AND CONCLUSIONS

The usefulness of time-averaged holography as a means for studying complicated modal surface vibrations is demonstrated. It is shown that holography provides a means for studying subtle surface motions not possible at present by any other means. Even in the case where the motion has components in the surface as well as normal to it the fringe system can be unambiguously interpreted. However, for this purpose several independent time-averaged holograms are required and their number is determined by the complexity of the motion.
INTRODUCTION

Solutions of structure problems can be, in general, intractable according to the three-dimensional theory of continuum mechanics. Consequently, the method of series expansion and the asymptotic method are, for instance, used to extract one- and two-dimensional approximate theories of structures from the three-dimensional theory. By the use of the latter, to construct a dynamic theory for piezoelectric bars and plates is the topic of this paper.

THEORETICAL BASES

In this study, a parametric expansion technique (see, e.g., Goodier 1938, and Johnson and Widera 1971), used so far to derive elastic beam, plate and shell equations, is extended to establish piezoelectric ones. First, a suitable scaling for space coordinates and time is introduced. And the dimensionless field quantities and constants are defined. Next, all stresses, mechanical and electrical displacements and electric potential are expanded in power series of a small parameter. The parameter is taken to be a ratio of piezoelectric plate thickness and its characteristic length and that of a characteristic dimension of bar cross-section and bar length. Then the expansions of dimensionless field quantities are substituted into the field equations and the asymptotic integration of these equations is performed over the thickness of plate and a cross-section of bar. And equating the corresponding powers of the parameter and requiring that each system be integrable in a step by step manner, a successive system of approximate governing equations of piezoelectric bars and plates is constructed.

CONCLUSIONS

A set of one- and two-dimensional governing equations is established by the use of the method of asymptotic integration of the three-dimensional equations of piezoelectricity. The approximate governing equations accommodate all the types of motions of piezoelectric bars of uniform cross-section (cf., Dökmecci 1974) and plates of constant thickness (cf., Tiersten 1969, Mindlin 1972, and Dökmecci 1974 and 1976). As a result of the asymptotic method applied here, the lower order approximations reduce the number of material constants even though complete anisotropy is allowed. The higher order approximations incorporate the thickness and cross-sectional effects associated with those of transverse shear and normal strains and the rotatory inertia. The conditions sufficient to ensure uniqueness in the solutions of the governing equations are enumerated with the help of the classical energy argument. Also, some numerical examples are given. And the results are compared with those of earlier studies.

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SOME PROBLEMS IN THE FREE AND RANDOM VIBRATIONS OF STRUCTURES

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Introduction
An approximate method is given for the free and random vibration analysis of structures occupying rectangular region with any set of continuous boundary conditions. The frequency/wave-number relationship for the exact solution cases are used to construct approximate solution for general boundary conditions.

Analysis
The problem is formulated as follows: given the probability behaviour of the external forces, find that of the displacement and/or stress field in the structure with continuous boundary conditions, displacing with initial conditions. For this stochastic boundary value problem an approximated method is presented, the first step of which consists (as in the normal mode approach) in expanding the given and sought functions in stochastic Fourier integrals and in series in terms of the modes of the undamped structure. An approximate method for determining the eigenfrequencies and eigenfunctions was presented in Refs. (1) and (2). It consisted of two Voight-Lévy type problems and a postulated eigenfrequency/wave-number relationship. The set of possible eigenfrequency/wave-number relations is obtained by considering the problems capable of exact solutions and generalizing the resulting eigenfrequency/wave-number relations in such a way that the approximate solution will always reduce to the exact solutions for the appropriate boundary conditions. The next step consists in using Galerkin's method for the determination of the probabilistic response. This procedure, together with expansion of the displacements and external forces, results in a set of linear equations for the cross spectral densities of responses and, inter alia, yields in the second, "improved" approximation of the eigenfrequencies. The cross correlations between like vibration modes and those between unlike modes are calculated. The method is applied to thin, rectangular, fully clamped isotropic and orthotropic plates and to fully clamped cylindrical shells. The special attention is paid to the role of the cross correlations [Refs. (3)-(5)].

The essential advantage of the method presented is that it coincides with the normal-mode approach for the cases capable of exact solution.

References
ACOUSTIC TRANSMISSIBILITY STUDIES ON A SKIN STIFFENED PANEL

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INTRODUCTION

In this paper the acoustic transmissibility of a skin stiffened panel is studied using two different theoretical approaches viz., the Statistical Energy Analysis and the Generalized Harmonic Analysis and the results were verified experimentally.

THEORITICAL

For a multi dimensional system the output power spectral density may be shown as a function of two complex quantities by the normal mode approach with the aid of Generalized Harmonic Analysis.

\[ \Phi_{out}(f, \omega) = \sum_{mnrs} H_{mnrs}(f, \omega) A_{mnrs}(r, j\omega) \]

Where \( H_{mnrs} \) is the frequency transfer function containing the modal shapes of free vibration and rigidity of the structure. This was evaluated for the panel using the Lin's (1) equations for stringer torsion mode and the stringer bending mode. \( A_{mnrs} \) is the spatial distribution of the power spectral density of the input pressure and it was obtained experimentally.

Statistical Energy Analysis: The structure, its exciting and radiating sides were treated as three systems in this analysis. Power balance equations for each of the three systems were written in terms of the input, dissipated and transferred power. Modal energy formed the independent dynamical variable. A ratio of the energies on the output and input sides of the structure is a measure of transmissibility.

EXPERIMENTAL

The structure was excited by random acoustic excitation. The power spectra of the input and output pressures were obtained by taking the Fourier Transform of the auto correlation function using an FFT algorithm.

RESULTS AND CONCLUSIONS

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<th>GHA</th>
<th>SBA</th>
<th>EXPT</th>
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<tr>
<td>10 dB</td>
<td>9 dB</td>
<td>11 dB</td>
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</table>

The results show the good agreement between the theoretical and experimental values of the overall acoustic transmissibility for the skin stiffened panel.

REFERENCES

III. Psycho and physiological acoustics

H. HEARING
We will consider a two-chamber model of the cochlea, with the basilar membrane (BM) located along the partition (Fig. 1a) and make the following assumptions:

1. The chambers are identical rectangular boxes with rigid outer walls.
2. The fluid filling the chambers is incompressible and inviscid.
3. Nothing varies along the z-direction (i.e., perpendicular to the plane of the paper).
4. The input velocity at the stapes is independent of y.
5. At two points located symmetrically with respect to the BM, the y-components of the fluid velocity are identical and the x-components are equal but opposite in sign.
6. The mechanical properties of the BM can be described by an impedance \( Z(x, \omega) \) which is the ratio of pressure on the BM to its velocity, at a point \( x \) and radian frequency \( \omega \).

In view of assumption 5 the two chambers can be "opened out" as shown in Fig. 1b. With coordinates defined in that figure, and an input stapes velocity \( \exp(j\omega t) \), the steady state pressure \( P \) satisfies:

\[
\nabla^2 P = 0 \quad \text{in the chamber}
\]

subject to the boundary conditions that \( P = ZV \) on the BM (\( y = 0 \)) and that the normal component of \( \nabla P \) be 0 at \( y = \pm H \) at \( x = \pm L \), and \( +j\omega \) at \( x = L \).

The problem of BM motion has been formulated essentially as in eq. (1) by many authors since the 1950's. However most analyses have made the further simplification that the fluid velocity is independent of \( y \). Such "one-dimensional" motion is of course not consistent with BM motion. Two-dimensional motion has been considered in Refs. [1] and [2]; however the methods used in those articles require orders of magnitude greater computing effort than the one described here.

By using a conformal transformation which maps the rectangle of Fig. 1b. on to the real axis, it can be shown that the BM velocity \( V \) is a solution of

\[
\frac{Z(x, \omega)}{j\omega} V(x, \omega) + \int_{-L}^{L} F(|x - x'|) V(x', \omega) \, dx' = -x
\]

where

\[
F(\sigma) = \frac{1}{\pi \sinh \frac{\pi \sigma}{2H}} \left[ 1 - \frac{L - \sigma}{2H} + \frac{1}{\pi} \exp\left(-\frac{\pi \sigma}{H}\right) \right] + \delta(\sigma)
\]

In eq. (3), \( \delta \) is a Jacobian elliptic function, with the ratio of complete integrals \( (K/K') = (2L/H) \). The approximation that the remainder in eq. (3) can be regarded as a \( \delta \)-function for extremely reasonable values of \( L \) and \( H \). The structure of \( F(\sigma) \) enables us to write a pair of 2nd order differential equations which are equivalent to eq. (2). Note, that the \( \delta \)-function term can be accounted for by redefining the impedance \( Z \). Thus with \( Z = (Z + j\omega e) \) and \( P = ZV \) it can be shown that

\[
\frac{d^2 P}{dx^2} = j\omega \left( -\frac{3}{HZ} \hat{P} + \frac{\pi^2}{H^2} W \right)
\]

\[
\frac{d^2 W}{dx^2} = \frac{\pi^2}{H^2} W + \frac{2}{HZ} \hat{P}
\]

In eq. (4a,4b) \( W \) is an auxiliary variable with \( W(0) = 0 \) and \( W(-L) = 0 \).

The BM velocity computed from eq. (4a,4b) fits modern experimental measurements [3] much better than "one-dimensional" approximations. Input impedance at the stapes is almost identical for the one- and two-dimensional models. Graphs showing these computations and comparisons will be presented at the talk.

A three-dimensional model of the cochlea is examined. The model has the following features:
The geometry of the model is an uncoiled, rectangular tube consisting of two scalae which have constant and equal cross-sections. The fluid in the scalae is considered as incompressible and inviscid as a first approximation. The scalae are separated by a partly bony, partly pliable partition. The pliable part, named basilar membrane, in its turn has two parts (pars pectinata and pars tecta) with distinct motions; cfr (1). These parts are, contrary to (1,2), characterized by visco-elastic plates of variable width and thickness.

The mathematical formulation leads to a boundary value problem for the difference pressure, i.e. the pressure in the upper scala minus that in the lower scala. The difference pressure must satisfy Laplace's equation in three dimensions, and in addition boundary conditions at the six walls, one of which is the partition of the scalae. The condition at the partition is too complicated to allow a straightforward treatment. Therefore it is assumed that the lengths of the basilar membrane displacement waves are large compared to the width of the membrane. This assumption is confirmed theoretically in (3) and experimentally in (4). By means of Fourier series expansions in the direction along the width of the membrane combined with an asymptotic method, a feasible boundary condition at the partition is derived. The asymptotic method, the method of matched asymptotic expansions, moreover enables to reckon with the attachment of the basilar membrane at the basal and apical ends of the cochlea. An interesting result of the mentioned approach is that a physical significance can be given to the widely used point-impedance characterization of the basilar membrane.

The remaining problem is to solve a number of Helmholtz-equations in two variables, which are coupled through the boundary conditions at the partition. To find the solution, once more the method of matched asymptotic expansions is used; this time the method is based upon the assumed small ratio of scala height \( h \) and wavelength \( \lambda \). The conventional long-wave approximation (5) gives good results only in the region \( \lambda \gg 2h \). The present approach gives good results in the region \( \lambda > 2h \), which covers at least the major part of the cochlear waves (3,4,5). For values of the wavelength which are somewhat smaller than twice the scala height, probably still quite good results are obtained, as can be conjectured from the solution of a problem which bears some analogy to the problem of cochlear waves (5). The advantage of the present approach to that of other multi-dimensional models is the reduction of the model equations to a system of ordinary differential equations, so that the numerical work becomes relatively simple.

References:
In this study a simplification of the cochlear partition is made, so that attention may be focused on the gross aspects of the fluid motion.

Many investigators have utilized the one-dimensional (long wavelength) approximation for the cochlear fluid motion, while a few have, in recent years, worked with a two-dimensional description. The corresponding physical model can be thought of as resulting from straightening the cochlea and then distorting the cross-section into a tall, narrow rectangle. The equations give a transition from the long wavelength to short wavelength behavior. Much of the response indicated by such a model can be correlated with physiological data on the cochlea. One glaring exception is the rate of decay of the traveling wave. Several investigators have found the endolymph and perilymph to have about the viscosity of water. Without an artificial mass and dissipation given to the basilar membrane, the two-dimensional model gives virtually no decay of the traveling wave past the point of maximum amplitude.

A convenient method of analysis for the one and two-dimensional models is provided by the phase-integral method. We show that this can be extended to a model with a cross-section whose height is the same as the width, which is a good step closer to the actual mammalian cochlea. For this case, the fluid motion is fully three-dimensional. The interesting result is that the transition from long to short wavelength occurs much more sharply than in the two-dimensional model. The consequence is a more substantial damping of the traveling wave due to the cochlear fluids without much change of the place of maximum response. The rate of decay is almost equal to that observed at a long time postmortem but still much less than the in vivo measurements, either in the spatial envelope for fixed frequency (1) or in the tuning curve at a fixed point (2). The three-dimensional model does give the observed steepening of the low frequency envelope from 6 db/octave to around 25 db/octave just before the maximum (3).

The curvature of the cochlea can also be considered. The calculations show little effect in the first turn of the cochlea, for which the ratio of radii to outer and inner walls is typically about 2. A substantial decrease in the effective fluid impedance occurs in the apical portion where the ratio is large. Since the sharpest tuning occurs in the first turn, the tentative conclusion is that curvature does not have a sharpening effect, as far as the gross fluid motion is concerned.

REFERENCES


MÖGLICHKEITEN UND GRENZEN FÜR DIE ANWENDUNG DES INTRACOHLEÄREN INTERFERENZEFKTS FÜR DEN PERSÖNLICHEN SCHALLSCHUTZ

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EINLEITUNG


THEORETISCHE GRUNDLAGEN

Schall, der unser Innenohr auf dem Luft- und Knochenleitungswege erreicht, kann durch Interferenz innerhalb der Schnecke (cochlea) in seiner Wahrnehmbarkeit erheblich beeinträchtigt werden. Kommen Luft- und Knochenschall auf der Basilarmembran mit gleichgroßer Amplitude und entgegengesetzter Phase zur Wirkung, so können sie sich im Idealfall gegenseitig auslöschen.

VERSUCHSAUFBAU

Das Blockschaltbild der verwendeten Versuchsanordnung zeigt Fig. 1. Als Signalquelle wurde ein Tongenerator verwendet, an den ein binauraler Luftschall-Kopfhörer und über einen eigens für diese Untersuchung aufgebauten Knochenhörerverstärker - zwei Knochen- schallhörer angeschlossen waren. Die Knochenhörer waren an der Stirn und am Hinterkopf der Versuchsperson angebracht. Der Knochenhörerverstärker war mit zwei voneinander unabhängig bedienbaren Stellern ausgerüstet, mit denen die Versuchsperson den ihr angebotenen Knochenschall amplituden- und phasenmäßig so abgleichen konnte, daß die resultierend wahrgenommene Lautstärke ein Minimum ergab. Die größte erzielbare Lautstärkereinigung \( \Delta L_{\text{max}} \) in dB wurde durch den Hörvergleich ermittelt.

ERGEBNISSE UND SCHLÜSSELZÄHLERGEBNISSE


LITERATUR

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NOISE INDUCED HEARING LOSSES - MAY THEY BE EXPLAINED BY THE BASILAR MEMBRANE MOVEMENT?

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Krokstad, A.
Skarstein, Ø.

INTRODUCTION
The mechanism of noise induced hearing losses is not fully understood. Maximum losses are typically located in the range 4000 - 6000 Hz, irrespectively of the fact that common noises have most of their energy at lower frequencies.

ANALYSIS
It is near at hand to believe that noise induced hearing losses are due to mechanical stimulation and destruction of hair cells caused by the strong movement of the basilar membrane.

The movement of the basilar membrane has been measured for 7 human temporal bones, using the Mössbauer technique, and has also been calculated from a model of the inner ear, represented by a simple transmission network.

For 121 industrial noise spectra the correlation between various measures of the noises has been determined. These measures are, in addition to the displacement level at the characteristic basilar membrane point for approx. 4000 Hz, the A, B and C weighted levels and the noise rating number for the input noise to the ear.

The best correlation found was between the B-weighted noise levels and the displacement levels. This fact supports the hypothesis that noise induced hair cell destruction may be determined by the displacement of the basilar membrane.

A place-dependent relation between basilar membrane displacement and the resulting hair cell stimulation may be postulated. When we demand the normal threshold stimulation at characteristic frequency to be independent of place, this relation may be determined, and consequently the place dependency of the hair cell stimulation for a given noise to the ear. The analysis leads to the attractive result that hair cells corresponding to characteristic frequencies near 5000 Hz will have the strongest stimulation, almost irrespectively of the noise spectrum.
LOUDNESS OF SUB- AND SUPER- CRITICAL BAND STIMULI IN
OBSERVERS WITH NORMAL HEARING AND COCHLEAR HEARING LOSSES

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INTRODUCTION

The purpose of this study was to examine the loudness of stimuli that were narrower and wider than a critical band at a center frequency of 1000 Hz in normal-hearing observers (Os) and Os with cochlear hearing loss. Several studies have employed the method of loudness summation to measure the critical band in patients with cochlear hearing loss (1-2-3-4). While some studies report no loudness summation beyond a number of critical bands, other studies report some loudness summation. This conflict may be a function of the stimuli used to measure the critical band and/or large individual differences. The present study examines these factors.

EXPERIMENTAL DESIGN AND METHOD

The loudness of 5 stimuli was investigated in 20 Os. Ten Os had normal hearing and 10 Os had cochlear hearing losses. The stimuli consisted of a pure tone, two-tone complexes and noise bands. The complex stimuli had frequency separations of 1/3 octave and 1600 Hz. The components of the two-tone complexes were set equally loud. Stimuli were 1 sec. in duration with a rise-fall time of 35 msec. Each of the five stimuli was matched in loudness to itself and to each of the other stimuli, yielding a total of 25 judgements per O. Test conditions were identical for all Os and the loudness level was approx. 65 phons.

RESULTS AND CONCLUSIONS

Less loudness summation was obtained for Os with cochlear hearing losses than for normal-hearing Os. This reconfirms earlier studies. All Os showed greater loudness summation for the 1600 Hz noise band. While normal-hearing Os have approx. the same amount of loudness summation independent of the comparison stimulus, loudness summation for Os with cochlear hearing losses depended on the comparison used. The conflict between earlier studies can be explained by this fact. Furthermore, inter- and intra-subject variability varied depending on the stimulus pair tested for both groups.

ACKNOWLEDGMENTS

The provision of excellent working conditions by Dir. O. Juhl Pedersen and Dr. G. Salomon is gratefully acknowledged. The measurements were carried out at the Technical University of Denmark. This work was supported by The Research Medical Council and The Tuborg Foundation.

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JUST AUDIBLE PITCH AND LOUDNESS MODULATIONS WITH COCHLEAR IMPAIRMENT

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Experiment

Just audible modulation depths (jnm's) of a pure tone which were simultaneously modulated in frequency and amplitude were determined for an observer with a bowl-shaped sensorineural hearing loss of about 40 dB between 1 and 2 kHz. The jnm's are plotted as a contour in the frequency-amplitude plane. The orientation of the significance bars indicate the way in which the AM and FM modulations were varied for the measurement of the jnm. The modulation frequency was 5 Hz.

Results

Figure (a) presents the jnm-contour for f = 3000 Hz and L = 15 dB SL (on the high frequency tail of the hearing dip). The curve reveals clear differences between in-phase jnm's (upper half) and 180° out-of-phase jnm's (lower half). This result is not found for normal hearing observers in similar stimulus conditions (1, 2); it points to an interaction between the perception of AM and FM. For this particular observer this interaction is ascribed to the effect of frequency on loudness as a result of the frequency dependent hearing loss.

When the jnm's are transformed from the physical frequency-intensity coordinates into the perceptual pitch-loudness coordinates, Fig. (b) is obtained. The symmetrical shape of the transformed curve does not any more suggest interactions between the perception of pitch and loudness changes. The transformation is given by the slopes of the equal-loudness and equal-pitch contour of this observer (2).

Similar results were obtained for a stimulus condition on the low frequency tail of the hearing loss. The conclusion of independent detection of pitch and loudness changes confirms the conclusion from similar experiments on normal hearing observers (2). Moreover, the jnm's in the frequency-intensity domain suggest interactions only if there is an effect of frequency on loudness and/or intensity on pitch, either as a result of a hearing loss (this experiment) or of the presence of a partly masking noise (2).

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THE EFFECT OF ACETYL CHOLINE ON THE ACTION POTENTIAL IN
THE AUDITORY NERVE, AND BRAIN STEM

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London, SE1, England

INTRODUCTION
In the junctions (synapses) in the nervous pathways from the ear to the
brain (afferents) and from brain to ear (efferents) chemical transmitters bridge the gaps. This paper is concerned with the efferent trans-
mitter.

THEORETICAL BASIS
The electrical changes in the nerves (in the millivolt range) have a
latency from the sound stimulus and an amplitude which are varied by
the intensity and character of the sound and by variations in the
internal milieu of the listener. Drugs which have known effects on the
transmission of nerves and on the quantities and availabilities of
transmitter substances, in particular on the transmitter acetyl choline
have been used to elucidate the nature of the process.

EXPERIMENTAL TECHNIQUES
In this work the technique of Sohmer (1) was used on guinea pigs. 2048
clicks of 50 dB were presented at a rate of 10 per second through a
magnetically shielded (TDH 39) click earphone to both ears. Suitable
filtration, common mode rejection and signal averaging are used to
obtain the electrocochleogram and the brain stem responses. The anim-
als which were fully conscious and unsedated were lightly restrained
in an accoustic chamber. The apparatus used was the Medelec mark II
Electrocochleograph.

The animals were given physostigmine which potentiates the effect-
s of acetyl choline by impeding its breakdown by cholinesterase (anti-
cholinesterase effect) and which passes through the blood brain barrie-
r, and prostigmine which has the same effect but which does not pass
the barrier. They were given atropine sulphate which antagonises the
effects of acetyl choline by competition and which passes the blood
brain barrier, and atropine methyl nitrate which has the same peripher-
al effects but which does not cross the barrier.

RESULTS AND CONCLUSIONS
This work using a non invasive technique on conscious animals confirms
the work of Gannon (2) who showed that atropine increases the latency
of the electrocochleogram. We have shown that this is a central
effect. When the effects of acetyl choline within the brain are
enhanced, there is a halving in amplitude and little effect on latency,
supporting its claim to be the inhibitory transmitter (3).

REFERENCES
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Introduction

It has long been known that the ratio of correct responses in pitch discrimination tasks depends on the intensity of the stimuli embedded in noise and their duration. It also depends on the number of alternatives in the n-AFC tasks [1,2], time separation of the stimuli, frequency separation and sensitivity of the observers. It should be also emphasized that the comparisons of pitch should be carried out outside the region of post-stimulatory masking [3]. The present report aims at extraction of these factors which seem to deteriorate pitch discrimination in pulses separated by 700 ms intervals at low sensation levels.

Experimental procedures

Three musicians of the group of 25, highest scoring in the DL for frequency screening test were used as subjects. Experimental procedures were n-AFC and a-2AFC. Time spans from the onset of the stimuli to the post-decision signals from the subjects were measured in the detection and pitch discrimination tasks.

Results

The experimental data, containing over 5.10⁴ individual judgements show that time spans necessary for the discrimination of pitch are significantly larger than those in the pure detection tasks and also for smaller frequency increments Δf. The nature of decision to be undertaken by the subject requires in both cases different amount of information income and processing time. This is thought to be a partial explanation for flattening of the frequency discrimination psychometric curves relative to the detection curves when the duration of the stimuli and Δf decrease.

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STEepness of (PSYchophYSical) Tuning CURves

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Introduction
There is a discrepancy between the steepness of the basilar membrane motion patterns and the physiological tuning curves for single frequency stimuli. A second filter was proposed to account for the discrepancy. For various reasons a non-linearity was placed between the first (mechanical) filter and the second filter. We are interested in the effects of the non-linearity on tuning curve determinations and the consequences for the second filter.

Method and results
We determined pulsation threshold patterns in four different methods (Verschuure et al, 1974). The presentation conditions are described by Verschuure et al (1976). The measured patterns show a strong non-linear behaviour with level: the steep slope steepens for higher levels while the slight slope gets less steep.

We analysed the patterns by assuming that continuity is perceived if the absence of the probe signal cannot be detected by the central auditory system; we also assumed that both probe and test signal are extended along the frequency axis. These assumptions mean that the probe pattern may not exceed the test pattern by more than a just noticeable difference at any frequency.

The observed non-linear behaviour of the pulsation patterns allow for and point at off-frequency detection of pulsation at the steep slope of the excitation pattern. An implication of these non-linearities is a pulsation pattern that is much steeper than the excitation pattern on which it is based; the slope certainly does not reflect the tuning properties alone, but also the non-linearity. This statement also holds for single cell-recordings from the auditory nerve in cats.

The absence of the non-linearity in the basilar membrane motion and its presence in the auditory nerve results suggests that it is generated at haircell-level, either by the kind of motion detected by the haircell, or by inner-outer haircell interactions.

The relation of the described non-linearity with other non-linear phenomena will be discussed.

References
INTRODUCTION

Blind persons are able "to hear obstacles" and to use auditory information to avoid collisions. This ability is known as "facial vision" (1). Two different modes of perception play a role. Obstacles at a large distance of the sound source and the blind produce distinct echos, especially when the sound has a discontinuous impulsive character (e.g. tap with the long cane).

For nearby obstacles, and particularly with continuous wide-band sounds, the reflections are not perceived as echos but they interfere with the original sound. In spectral terms: the original sound is comb-filtered with maxima at \( n/\tau \), where \( \tau \) is the time delay between the original and the delayed sound (\( n = 1, 2, 3, \ldots \)). This is heard as a coloration of the original sound accompanied by a pitch sensation (2).

In a simple geometric situation where a person is positioned between a sound source and a reflecting wall, this pitch corresponds to \( 1/\tau = c/2d \), where \( d \) is the distance between the person and the wall, and \( c \) is the velocity of sound.

The early-blind has spontaneously learned to use these perceptual skills. The late-blind, however, has to be made familiar with the subjective effect involved and should be trained in using it.

TRAINING PROCEDURE

A training procedure as it is applied at the centre for revalidation of the blind "Het Looerf" (the Netherlands) can be subdivided in three stages. Stage 1 deals with basic familiarization, stage 2 provides training under idealized sound conditions with real obstacles in a special training hall. In stage 3 the blind person encounters daily-life situations.

This paper deals with stage 1 of the training, its methodology and the development of a training device. Basically, the blind person listens via a loudspeaker or a headphone to an idealized sound (a short click or white noise) together with the same sound delayed (the addition is done electronically). The blind is seated during stage 1. Moving towards or from an obstacle is simulated by linearly sweeping the delay time \( \tau \) in the appropriate direction. The strength of the pitch sensation can be changed by control of the relative intensity of the delayed sound.

A psychophysical method is applied during the control of such parameters as delay time, intensity, "walking velocity", to assist the blind person in obtaining a threshold of perceptibility of pitch as low as possible. This is done "manually" by the revalidation therapist, but could also be done automatically.

TRAINING DEVICE (EOS 1)

The training device (EOS: electroacoustic obstacle simulator) basically contains two digital shift registers with modulo-2 feedback. They generate two identical maximum-length-series, pseudo gaussian noise signals, which can easily be delayed with respect to each other. This is done by applying clock pulses differently to either of the shift registers. Several modes of manual or automatic time shift are incorporated: constant \( \tau \) and changing \( \tau \) in a linear way with adjustable (walking) velocity.

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INTRODUCTION

It is desirable that an audioengineer should possess good hearing ability, because he must finally decide by his ears whether a tone attains his object or not. For this purpose his judgment on tones must be accurate and reliable. In the department of acoustic design of Kyushu Institute of Design, the first and second year students receive training of hearing every week. This training has been designed so as to systematize what the audioengineer must acquire.

METHOD

The contents of this training of hearing can be divided into three main classes: the training for tone sensation, the training for art of tones and training for speech sounds. The first one is for relative and absolute judgment on pitch, loudness, tone colors, length of tones, rhythm, tonal memory, reverberation time, sense of direction of tones, etc.. The second one is for relative and absolute judgment on musical interval, harmony, tone colors of musical instruments, tone colors of played music, tones of radio drama, etc.. The last one is for relative and absolute judgment on tone colors of vowels and consonants of Japanese and foreign languages.

For example, the contents of training of hearing for relative judgment on pitch is like this. 50 pairs of tones are made and the first tone is 440 Hz pure tone and the frequency of the second tone is different from that of the first tone. The differences in frequency are 17, 12, 8, 5, 4, 3, 2 and 1. Students listen to the 50 pairs of tones and are asked to tell whether the frequency of the second tone is higher or lower than the first one. At the beginning, they listen to these tones, record their answers and then are given the correct answers. In the next step, they once again take the test using the same type of 50 pairs of tones and are again given the correct answers. Thus each student can know his degree of hearing ability. After repeating this process, the learning curve of all students is observed. When a teacher judges from the shapes of these curves that this training has satisfactorily advanced towards its goal, the next period of training is started.

RESULTS AND CONCLUSION

In Fig.1, are shown the results of training for relative judgment of pitch in 3 students. Student S1 is shown to have good hearing ability for pitch discrimination at the beginning. The curves of students S2 and S3 show their progress through the training.

Through this program students will learn to judge the various properties of tones accurately and develop in listening to sounds carefully. In conclusion, students who have successfully completed this program, will come to acquire the judgment on tone properties that are highly accurate and reliable.

Fig.1 The results of training of hearing for pitch discrimination in students S1, S2, S3.
INTRODUCTION

There has been increasing concern (1,2) over the past ten years or so about possible damage to hearing from regular exposure to amplified pop music. A pilot study was undertaken over a 7-month period in Southampton University to measure the noise levels and spectra of amplified recorded and 'live' pop music and its effect upon the hearing of young people regularly exposed to it.

PROCEDURE

Equivalent continuous noise levels, $L_{eq}$, and octave-band frequency analyses of the music were obtained at various positions in an auditorium on each of 10 occasions. The thresholds of hearing of 17 screened subjects who regularly attended such functions were also measured at the beginning and end of each evening on 10 occasions.

RESULTS AND DISCUSSION

Measured noise spectra showed maximum sound levels in the octave band centred on 125 Hz, which decreased at about 5 dB/octave above this frequency. The overall average $L_{eq}$ of 'live' pop concerns was 105 dB(A), whilst that for discotheque music was about 97 dB(A).

At the start of the study, subjects exhibited a mean pre-exposure hearing loss of about 6 dB (averaged over 3-6 kHz) due to pop music, but no apparent change in this hearing level could be detected over the period of the study. However, subjects acquired a mean temporary threshold shift of about 10 dB (averaged over 3-6 kHz) after each exposure.

CONCLUSIONS

It would appear that pop music represents a 'borderline' hazard to hearing and that some limitation of sound levels in public places of entertainment is probably necessary. A tentative limit is suggested.

REFERENCES

INTRODUCTION

As part of a large study on the screening of middle ear disorders and hearing in 3 to 6 year old children, values of static acoustic susceptance (otoadmittance) at 220 Hz have been examined. These values have been computed in an attempt to improve their predictive utility in comparison with the otolaryngological findings.

EXPERIMENTAL TECHNIQUES

The audiometrist who collected tympanometric data and the otolaryngologist recorded findings independently. Static acoustic susceptance \( B_A \) was calculated in the conventional manner using maximum peak compliance minus the average of positive and negative (200 mS) stiffened conditions of the tympanic membrane. The otolaryngological findings were then noted. Subsequently \( B_A \) values were compared to normative values of .45 to 2.25 acoustic millisiemens. Some investigators have written that to generalize to middle-ear function, zero-cross on the tympanogram is a more realistic comparison with the stiffened condition than maximum peak. Therefore, susceptance \( B_p \) was calculated at the zero-cross. In addition, the difference between maximum peak value and that at zero-cross was taken. The average derived value from a small group of subjects, was .15 millisiemens, and the results were compared with otolaryngological findings.

RESULTS AND CONCLUSIONS

Conventional static acoustic susceptance values on the preschool group are in close agreement with those reported by others. The data on those children with serous otitis media and \( B_A \) within the normative range (.45 to 2.25 mS) were examined separately from those whose susceptance fell below .45 mS and will be compared with those of other studies.

The results are based on an expanding sample; however, at the present time, the characteristics for predicting positive or negative otolaryngological findings are better for the method which incorporates the compliant factor at both maximum peak and zero-cross readings than for the method using one reading alone. The method is not necessarily selective for an ear but rather a condition that requires follow-up care. The method is not selective for subjects who have middle-ear pressure peaks of -200 mmHg or more negative pressures, nor for those who have zero or positive peaks.

REFERENCES


INTRODUCTION

Pour l'éducation orthophonique des jeunes sourds profonds, il faut établir la boucle de contrôle par un autre sens que celui de l'ouie.

BASES THÉORIQUES

Différentes techniques d'analyse de la parole, analyse spectrale, codage prédictif, filtrage inverse, ... extrayaient des caractères simples de la parole : intensité, mélodie, position et amplitude des formants, fonction d'aire du conduit vocal ...

Le système établit le dialogue entre trois partenaires : l'orthophoniste qui est le maître des commandes et juge la qualité de la voix, l'enfant qui suit les indications de l'orthophoniste et contrôle son élocution par examen permanent d'un écran, le système lui-même qui traite la parole et en fournit des représentations visuelles simples et instantanées, la plupart du temps sous forme de jeux.

Le système est également prévu pour que l'enfant puisse s'exercer seul et travailler sa parole pour l'amener à l'état réflexe. Dans ce mode de travail, il est nécessaire d'évaluer automatiquement la qualité de la parole et d'arrêter les exercices en cas d'erreurs répétées.

TECHNIQUES EXPERIMENTALES

Le système réalise plusieurs fonctions : fonctions de commande, d'analyses de la parole, de visualisation, de mémorisation et d'apprentissage. Cette structure modulaire facilitera la réalisation ultérieure par microprocesseurs. Des ensembles câblés effectuent les analyses, tandis que les autres fonctions se présentent sous forme de sous-programmes traités sur minicalculateur.

Le module essentiel est la visualisation des formes. Par souci de clarté, on ne doit visualiser qu'un seul paramètre à la fois : intensité, mélodie, voixement, ... La visualisation doit se faire rigoureusement en temps réel afin d'autoriser un contrôle immédiat de l'expression vocale. Dans ce sens, le couplage d'un écran peu rémanent au calculateur permet le tracé instantané de dessins animés. Enfin, il est primordial que l'indication visuelle apparaîsse sous une forme très simple, facilement interprétable et adaptée au stade d'évolution de l'enfant.

RESULTATS ET CONCLUSIONS

Les différents aspects de la parole que nous visualisons sont : l'intensité et la mélodie en fonction du temps, le voixement, le signal vocal, un mot ou groupe de mots sous forme de sonogrammes, l'amplitude en fonction de la fréquence, l'écriture ou le dessin d'un mot reconnu automatiquement.

La relative simplicité du système doit lui permettre de devenir une aide efficace pour l'éducation des enfants sourds.
FIRING PATTERNS IN THE COCHLEAR NUCLEUS AT ASYMPTOTIC THRESHOLD SHIFT

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Introduction

Noise-induced hearing loss typically presents a constellation of symptoms that include a pattern of hearing loss corresponding to the energy distribution of the noise, recruitment, tinnitus, decreased speech discrimination and compressed temporal integration function. These changes in auditory performance must be the consequence of alterations in the neural code underlying the auditory perception. Our approach to understanding the phenomena of noise-induced hearing loss is to create a well-defined asymptotic threshold shift, measure the response characteristics of neurons in the auditory system and, finally, relate both the hearing loss and the single neuron changes to morphological changes in the cochlea.

Experimental Techniques

Chinchilla were exposed to an 86 dB SPL octave band of noise centered at 4 kHz for 3-5 days. At the cessation of the exposure, the firing patterns of units in the dorsal and ventral cochlear nucleus were sampled for 1-12 hours. The animals were sacrificed and cochleagrams were measured.

Results and Conclusions

Figure 1 presents typical tuning curves and the presumed hearing loss as reported by Mills (1973). We have applied a correction for the external meatus of the chinchilla (von Bismark, 1967), therefore, thresholds and tuning curves are comparable as sound pressure at the tympanic membrane. Clearly, there is good agreement between the shape of the audiogram and the distribution of the tuning curves. At frequencies above 2 kHz not only is there a loss of sensitivity, but the low frequency side of the tuning curve is also very shallow leading to a pronounced broadening of the tuning curve. In some units only inhibitory areas were present, while in others, the threshold for the low frequency inhibitory areas was considerably lower than the threshold for the excitatory area. Spontaneous activity is systematically decreased in the frequency range where there is a hearing loss; especially in the range of 4-8 kHz. Finally, anatomical correlations showed small outer hair cell lesions in the basal coil of the cochlea.

References

INTRODUCTION

This study was intended to achieve the following aims:

a) To test the possibility of generating recovery curves to convert TTS to TTS₂, similar to those obtained by Kryter (1).

b) To test the validity of the equal energy theory when applied to TTS produced by impulsive noises.

c) To test the TTS produced by continuous and impulsive noises of the same energy.

EXPERIMENTAL PROCEDURE

Six normal hearing subjects were exposed to three different impulsive noises of different shape and duration, but of the same spectral content (white noise) and of the same energy. Also they were exposed to continuous white noise of the same energy as before. Each signal was presented one at a time, in three different occasions. The experiment was performed in an anechoic chamber, and the noise was monitored permanently from the outside by osciloscopic means.

Immediately when the noise was stopped, the hearing threshold of the subject's right ear was measured at 4,000 Hz, at different time intervals, using a Bekesy audiometer. Each audiogram was averaged by a computer method designed for this experiment.

EXPERIMENTAL RESULTS AND CONCLUSIONS

They can be summarized as follows:

a) It is not possible to generate recovery curves for impulsive noises, due to the large spread among the results, and also due to the dependance of the signal characteristics upon the TTS.

b) This dependance makes impossible the application of the equal energy theory to impulsive noises.

c) The TTS produced by impulsive and continuous noises of equal energy are quite different, both on the amount of the TTS, and also in their recovery rates.

If TTS is accepted as a valid estimator for the PTS, then the obvious conclusion is that the application of the equal energy theory to predict the effects of impulsive noises and to help to establish damage risk criteria can lead to erroneous conclusions.

REFERENCES

INTRODUCTION

It is generally accepted that the form of absolute threshold recovery curve after stimulation approaches to attenuating exponent. Nevertheless, there are some literature indications (1,2,3,4) that after the first decrease to normal level a new threshold shift follows, and only afterwards the temporary threshold shift disappears (diphasic curve).

The characteristics of temporary threshold shift under adaptation is great interest because it permits to understand the possible mechanisms of auditory sensitivity regulation.

EXPERIMENTAL TECHNIQUES

The work was carried out on 4 normal listeners. The threshold of hearing was tested by 1dB steps for tone 1000Hz before and after 2-minutes stimulation by the same tone at intensity of 80dB over sensitivity level.

RESULTS AND CONCLUSIONS

It was found that under adaptation the manner of auditory threshold recovery, after it first had reached the initial threshold level (first "minimum"), looked like attenuating periodic (50-60sec.) undulatory process. The characteristics of the process are similar to transitional characteristics of self-regulating automatic systems. This mechanism of regulation is probably connected with central nervous system.

REFERENCES

Introduction

An important question in studies of the effects of noise on hearing is whether or not repeated exposures over long periods of time result in a cumulative effect, or is there an upper bound to the hearing loss one can expect from a given noise. Asymptotic threshold shift experiments, coupled with exposure paradigms designed to mimic working conditions may answer such questions. While continuous noise paradigms have been used quite extensively in these experiments, there is a complete lack of comparable data on another important class of noise - impulse noise. This paper will report on an experiment designed to study the TTS from a work-week exposure to impulse noise, and how such a TTS relates to the ATS level of that same impulse.

Experimental Techniques

Five chinchilla were used as subjects; their hearing loss was estimated using auditory evoked response techniques or behavioral avoidance conditioning. All animals were exposed to 113 dB SPL impulses having a reverberant duration of 160 msec. The repetition rate was 1/sec. The animals received these impulses for five days, 8 hrs/day. Thresholds were obtained (.5, 1, 2, 4, 8 kHz) before, and at the end of each daily exposure. Recovery was followed daily for 5 days after the exposure then once again at 40 days post-exposure. After this last test, all animals were sacrificed for surface preparation histology.

Results

The accompanying figure depicts the median threshold shift (8 kHz) at the beginning and the end of the 8-hr. exposure for the 5 days. The results show a clear oscillation in the animals thresholds upon repeated exposures. The high frequencies were shifted 30-40 dB following each daily exposure, while recovery during the 16-hr. rest period before the next day's exposure amounted to only 20 dB. Thus, each animal began the next day's exposure with significant residual hearing loss. On the third day of recovery following the weeks exposure the animals still had a 5-10 dB hearing loss. This would coincide with the beginning of a new work week. There did not appear to be any accumulation of TTS over the 5 day period. This data is interesting to compare with the results of an ATS impulse noise experiment using the same impulse paradigm. For the work-week exposure, each animal had essentially reached his ATS level by the end of the first day, and the ATS level does, in fact, represent an upper bound of loss for this limited exposure paradigm.

References

TTS FROM TIME-VARYING NOISE: A FUNCTION OF AVERAGE LEVEL, NOT EQUIVALENT LEVEL

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In their quest for a single simple index to be used for characterization of noise exposures, many people have become entranced with the concept of equivalent level $L_{eq}(t)$--a fictitious steady A-weighted SPL that, had it been acting constantly over time $t$, would have resulted in the same "A-weighted energy" being delivered to the ear as that actually delivered by an intermittent or time-varying sound. However, there is scant evidence that any of the effects of noise depend only on $L_{eq}$. Indeed, earlier research has implied that temporary threshold shift (TTS) is determined better by average SPL than by equivalent level. This experiment was designed to extend this generalization.

PROCEDURE

Ten young normal-hearing listeners were exposed for 6 h to magenta noise (200-5000 Hz, OBL slope -5 dB/oct), using 5 different schedules on successive sessions. Two of the schedules were continuous 90 and 95 dBA noise, respectively. The other three consisted of successive 30-sec periods of 90-X, 90, and 90+X, repeated continuously, with X=5, 10, and 15, resp.; thus the average level was 90 dBA for all three, although the equivalent levels were 91.7, 95.7, and 100.4 dBA. Pulse-tone Bekesy audiometry (250 msec on, 250 msec off) was used to determine thresholds at 500, 1400 and 4000 Hz, before and at various times after each exposure.

RESULTS

The values of TTS, either 2 or 32 min after exposure, at 500 and 1400 Hz confirmed the average-level principle. That is, the TTS even after the 75-90-105-dBA exposure was not different from that after continuous 90 dBA. At 4000 Hz, although both the 80-90-100 and 75-90-105 conditions resulted in slightly higher values of TTS$2$ and TTS$32$, these TTSs corresponded only to values that would be produced by a 92- or 93-dBA continuous exposure. In no case did TTS exceed 10 dB.

DISCUSSION

It is obvious that the fatiguing capability of a noise that fluctuates over a range of even 30 dB is most adequately represented by its average level, not by its equivalent level. If other effects of noise prove to be as poorly related to equivalent level as is TTS, perhaps it will not be necessary to replace all sound-level meters with intensity meters after all, or at least to try to mentally weight the peaks observed on a standard meter face any more heavily than the valleys. In this regard, it will be noted that even the most ardent of $L_{eq}$ supporters realize that $L_{eq}$ performs rather badly in predicting annoyance. Noise Pollution Level, for example, consists of the sum of the $L_{eq}$ and 2.56 times the standard deviation of the noise level--of the level, let me emphasize, not of the intensity. Uncritical acceptance of "equivalent level" as an adequate measure of noise represents a giant step backward.
INTRODUCTION

In several animal and human experiments it has been shown that temporary threshold shifts produced by exposure to noise increase for the first few hours of exposure and then reach a plateau or asymptote. It has been hypothesized (Mills et al., 1970) that threshold shifts at asymptote produced by a given noise are an upper bound on any permanent threshold shifts that can be produced by that noise regardless of the duration of the exposure or the number of exposures. In the present study, the purpose was to determine the relation between the threshold shift at asymptote and the octave-band sound pressure level of the noise for bands of noise spaced across the audible range of frequencies.

EXPERIMENTAL TECHNIQUES

Groups of human observers were placed in a diffuse sound field for periods as long as 26 hours and exposed to different levels of an octave-band noise centered at 0.5, 1.0, 2.0, or 4.0 kHz. Measurements of auditory sensitivity were made during quiet periods interspersed within an otherwise continuous exposure as well as before and after the noise exposure.

RESULTS AND CONCLUSIONS

For all exposure conditions threshold shifts increased for the first 4-12 hours of exposure and then reached a plateau or asymptote. Threshold shifts at asymptote in the frequency region of greatest shift increase about 1.7 dB for every 1 dB increase in the level of the noise above a critical level, C. The value of C is frequency dependent, ranging from a value of about 78 to 80 dB at 0.5 kHz and 1.0 kHz to a value of 75 dB at 2.0 kHz and 70 dB at 4.0 kHz. Individual differences in the magnitude of the threshold shift were greatest for the 0.5 and 1.0 kHz exposures (std. dev.=6.0 dB), and least for the 2.0 and 4.0 kHz exposures (std. dev.=3.5 dB). In the temporal domain some subjects reached asymptote in 2 hours whereas other subjects required about 16 hours. Complete recovery of threshold shifts occurred for every subject and usually required about 16 hours. In some instances, however, subjects required about 48 hours for complete recovery. The present data, in many respects, are consistent with that reported for humans by others (Ward, 1975; Melnick, 1976). Where differences do occur they can be accounted for by ± 1.5 dB in terms of noise level or by different criteria for subject selection.

REFERENCES

W. Melnick, Effects of Noise on Hearing (1976) 277.
INTRODUCTION

The nonlinear amplitude response of the auditory system may represent a basic parameter of hearing. Although the existence of this nonlinearity and some of its effects are well documented, the precise nature of the characteristic has been subject to dispute over the past decade. This paper examines the nature of subjective pitches produced by excitation of the auditory nonlinearity by sound.

THEORETICAL BASES

Subjective pitches of the form \(nf_1+mf_2\), where \(n,m=0,1,2,...,k\) may be produced by a nonlinearity of the form

\[X = a_0 + a_1 p + a_2 p^2 + \ldots + a_n p^n\]  

(1)

where \(p\) is the sound pressure of the stimulus. The distortion products thus produced should have the following characteristics: 1) Their amplitude should increase exponentially with stimulus level at a rate determined by the order of the term of the nonlinearity from which they derived (e.g. 2dB/dB for \(f_1+f_2\); 3dB/dB for \(2f_1+f_2\) or \(3f_1\)), 2) If one primary level is fixed, the amplitude (in dB) of the subjective tone should vary in proportion to the multiplier ("n" or "m") of the other primary (for \(2f_1+f_2\), a 2dB/dB growth with \(f_1\) variable, a 1dB/dB growth with \(f_2\) varied), 3) Equal order sum and difference tones should be of equal intensity, 4) The subjective tone levels should be frequency independent.

EXPERIMENTAL TECHNIQUES

Subjective tones were measured with the tone-on-tone masking paradigm (1) in normal hearing (+10dB re ISO 1964) listeners. Results from fifteen listeners in four experiments will be reported. In all cases \(f_1=1000\) Hz.

RESULTS AND CONCLUSIONS

Subjective tones once they are generated are similar to acoustic stimuli. Those higher in frequency than the primaries are masked by the tones producing them. This masking increases at 2dB/dB with stimulus level regardless of distortion product order. Masking of distortion products lower in frequency than the primaries is little influenced by stimulus level. With few exceptions (most cases of \(f_1-f_2\) and \(2f_1-f_2\)) distortion products grow at an accelerated rate predictable from eqn. 1 and in agreement with theoretical bases 1, 2, and 3. Furthermore, sum and difference tone levels are predictable from harmonic measurements. Variation of the sum tone \(2f_1+f_2\) at \(f_1/f_2=1.2\) and 1.6 does not support a frequency sensitive nonlinear mechanism. Absolute levels of distortion products of equal order (2f_1+f_2) decrease with their frequency at rates which are explained on the basis of the spread of masking.

REFERENCE

DIFFLUENCE PHENOMENON TAKEN INTO CONSIDERATION FOR DETERMINING THE "MAGNITUDE" OF A GIVEN SOUND SENSATION FEATURE

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INTRODUCTION

The analysis (1) of difference limens (DL) occurrence leads to the conclusion that they are the effect of diffuence in sound sensation (but their magnitude is affected by nonlinear processes of signal energy transformation). It is assumed that DL of perception and discrimination of a given sound (S) parameter increments is the basis for any evaluations of the "magnitude" of a given feature of sound sensation (FSS) or subfeature of sound sensation in case of timbre (SFSS). However, using the magnitude of FSS or SFSS (and particularly the determination of infinitesimal increments of FSS eg. in case of loudness curves) have no justification in view of the phenomenon of diffuence. The occurrence of this phenomenon permits only a partial ordering of S in respect to the given FSS or SFSS.

THE NUMBER OF PERCEIVABLE AND THE NUMBER OF DISCRIMINABLE PARAMETER INCREMENTS INSTEAD OF THE "MAGNITUDE" OF A GIVVEN FSS OR SFSS

We take into account S differing by a given parameter designated by y. A partial ordering is designated (2) by the relation \( P(A_x) - P(B_y) > 0.5 \) where \( A_x \) for \( y = y_1, y_2 \) and for \( y = (y_3, y_4) \) are events consisting in the following answer: 1) the second S, where \( y = y_2 \), in respect of a given FSS or SFSS (eg. it is louder) follows the first S, where \( y = y_1 \) 2) the two last S, where \( y = y_3 \) and \( y = y_4 \) are more remote in respect to the given FSS or SFSS than the two standard S, where \( y = y_1 \) and \( y = y_2 \), and \( B_y \) are an events independent of \( A_x \) consisting in the answer: 1) the second S precedes the first one, 2) the two last S are less remote from each other than the first two ones. The difference \( P(A_x) - P(B_y) \) is designated by \( R_y \). By \( \Delta y_1, \Delta y_2, \Delta y_3, \Delta y_4 \) such y-parameter changes are designated where there are respectively \( R((y_1, y_2) - y_1, - y_2) = 0.5 \), \( R[(y_3, y_4) - y_3, - y_4] = 0.5 \), \( R[(y_1, y_2) - (y_3, y_4)] = 0.5 \). Thus it is assumed that the psychoacoustic conditions can be designated by the inequality \( P(A_x) \leq P(B_y) < P(A_x) = 0.5 \), \( P(B_y) = 0.5 \), \( P(A_x) = 0 \) when in the given conditions \( P(A_x) = 0.5 \) and \( P(B_y) = 0 \) or \( P(B_y) = 0.5 \) and \( P(A_x) = 0 \), then \( \Delta y_1 \) and \( \Delta y_2 \) are the traditionally determined DL of parameter change perception and \( \Delta y_3 \) and \( \Delta y_4 \) are analogically the determined DL of parameter change discrimination. We plot curves \( y_2 = y_1 + \Delta y_2(y_1) \), \( y_3 = y_1 - \Delta y_3(y_1) \) and curves \( y_4 = y_1 + \Delta y_4(y_2) \), \( y_4 = y_1 - \Delta y_4(y_2) \). It is assumed that for S between those curves the perception and discrimination of \( y \)-parameter changes is diffucent. From the particular curves we read the number of percepted and the number of discriminated \( y \)-parameter increments. The above outlined partial ordering can be carried over on more than one parameter.

CONCLUSION

In view of diffuence phenomenon occurrence the FSS and SFSS magnitudes are idealized notions. It seems that it would be practically less exaggerated to determine the above defined numbers than to read the magnitudes of the given feature from the curves of that magnitude.

REFERENCES

NONLINEAR TRANSFORMATION OF ENERGY OF THE ACOUSTIC STIMULUS AND PERCEPTION AND DISCRIMINATION OF ENERGY CHANGES IN CASE OF PARTIAL MASKING

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INTRODUCTION

It is a known fact (6) that the loudness of a partially masked sounds (PMS) increase more rapidly with the increase of SPL (over determined SPL) than the loudness of an unmasked sound (UMS). Therefore it should be expected that difference limens (DL) are smaller in the first case than in the latter. Such dependence of DL magnitude on loudness increment was observed (3) in case of perceptive deafness. Research results in case of masking can suggest (1,2) that such dependence does not exist. However, DL's are compared at the same SPL or SL. The problem to be solved is whether in case of masking the magnitude of DL of perception and discrimination of SPL increments is mainly the effect of difference (5) in the decision process, or whether it is the result of nonlinear transformation (4) of signal energy.

ANALYSIS OF DIFFERENCE LIMENS IN CASE OF PARTIAL MASKING

With each peripheral excitation evoked by a given signal with SPL equalizing y we connected a definite function (4) uniquely connected with the formation of loudness of that signal. The space of those functions is designated by Z_y. The space for the same PMS is designated by Z'y. If we have such y corresponding to a given y that Z'y and Z_y would be equal (Z'y=Z_y) then in the range of the faster increase of loudness for the PMS should be Z'y+Ay=Z_y+Ay for Ay < Ay. Thus masking would shift DL in the direction of lower SPL which would involve the increase of DL for the PMS in relation to DL of UMS with the same SPL. From the inequality Ay < Ay it follows that DL for PMS should be smaller than DL of equally loud UMS (Z'y=Z_y). This phenomenon would be connected with the decrease of the noneffectivity of energy transformation determining the formation of signal loudness (in a complex sound sensation) of PMS with the increase of its SPL. This phenomenon can be superposed by the increase of difference of loudness comparison corresponding to the excitation evoked by PMS. Then Z'y+Z_y. If the difference had a deciding effect on the DL magnitude of PMS, it should be reflected in the "cross DL" magnitude determined for the PMS in relation to UMS. The accessible experimental data show that "cross DL" should be smaller than DL of equally loud UMS. The above outlined considerations can be carried over on DL of SPL increment discrimination.

CONCLUSION

Theoretical argumentation based on accessible experimental data leads to the conclusion that the DL magnitude in case of masking is decidedly affected by nonlinear signal energy transformation processes. For supporting this hypothesis further experimental researches are needed.

REFERENCES

A MODEL OF THE ACOUSTIC REFLEX

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INTRODUCTION

It is well known that acoustic reflex contraction of the stapedius muscle changes the acoustic admittance of the tympanic membrane at low frequencies. However, there are no theories currently available which are able to predict these changes quantitatively. The aim of this paper is to develop an electroacoustic analogue model of the middle ear, including the stapedius muscle explicitly. The model is based upon measurements of the acoustic admittance of the eardrum in human subjects at two frequencies before and during various degrees of stapedius muscle contraction.

EXPERIMENTAL TECHNIQUES

Measurement of changes in the acoustic admittance at the eardrum at 220 and 660 Hz were obtained from 16 normally-hearing subjects using a Grason-Stadler 1720 otoadmittance meter and associated recording apparatus. Admittance changes were elicited by pulses of wide-band noise at sound pressure levels of between 80 and 120 dB.

RESULTS AND CONCLUSIONS

It was found that the change in eardrum admittance involved changes in both conductance and susceptance but could be approximated at both measurement frequencies by an increase in reactance, with resistance remaining constant. An electroacoustic analogue of the middle ear was evolved, at first from anatomical considerations similar to those of Zwislocki (1), and subsequently using the admittance-change data. The final model accurately simulated the measured admittance changes at both frequencies throughout the range of responses. The model has been used to predict the effects of hypothetical lesions of the middle ear on the eardrum admittance and to demonstrate the cause of particular tympanometric and acoustic reflex response patterns, such as the so-called W-notch which is often obtained during tympanometry using a 660 Hz probe tone.

REFERENCES

NEW DATA ON HARMONIC INTERVAL HEARING

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Lack of accuracy in tuning up musical intervals meets general acquiescence as result of loudness, spectral content or simply poor human hearing. For instance, octaves tuned up by skilled performers mostly exceed the 2:1 ratio. (1)

In order to draw up the scattering patterns of deviation from pure physical intervals, seemingly required to meet neat perception of (subjective) musical intervals, a test series among 160 conservatoire students has been carried out.

TESTING

Subjects had to match unisono, octave, fifth and fourth by (a) simultaneous; (b) sequential listening to pure (sine) tones; and finally (c) preset, more or less matched, intervals were randomly performed.

Both tones performed on separate audio chains had outputs set to assure equally 60 or 78 dB acoustic pressure level.

DISCUSSION

Although data processing has not been completed so far, preliminary results seem to point to some assumptions:
1.- skilled subjects are able to tune up very neatly (see dotted line on figure) if allowed to watch audible beat frequencies;
2.- if impeded, e.g. listening sequential (melodical) intervals or randomly performed presettings, subjects mostly avoid the near (beat) zone, thus fitting in one of the lobes of the solid curve;
3.- deviation dispersal fits in typical distribution pattern, as shown in fig. 1 for quarts; distribution differs slightly depending on interval and subjects' speciality (strings, piano, brass, voice);
4.- in higher and lower register other relationships may prevail;
5.- hearing mechanism variants (2) acting in different ranges may not be the same;
6.- extending tests on wider range may contribute to selecting actual hearing mechanism within the ranges.

REFERENCES

INTRODUCTION

Several auditory mechanisms are evoked when the intensity of a pure tone is manipulated in one ear of a listener with normal hearing. Loudness changes are obvious while the production of subjective harmonics usually goes unnoticed because of their relatively low amplitudes and the spread of masking (1). These phenomena, however, are probably related since they are only different aspects of the same integrated process.

THEORETICAL BASES

The behavior of aural harmonics is describable with the simple power series often used to characterize amplitude distortion (2). With added assumptions and simplifications, equations for the sensation of loudness can be derived from this supposed initial conversion of a pure tone input, e.g.,

\[ L = \frac{1}{2} a_2 \beta^2 + [a_1 \beta + \frac{3}{6} a_3 \beta^3] C, \]

where \( a_1, a_2 \) and \( a_3 \) are the first three coefficients of the power series, \( \beta \) is the difference in amplitude between the tone and its threshold, and \( C \) is a weighting factor.

EXPERIMENTAL TECHNIQUES

To evaluate the postulated loudness definition, the relevant parameters can be inferred from aural harmonic estimates and from fitting to a limited set of loudness judgments. The tone-on-tone masking procedure and loudness equissections have been used. The accuracy in predicting: (1) other equissections over different ranges, and (2) results of ratio productions are then examined (3).

RESULTS AND CONCLUSIONS

When the proposed loudness equation is linked to responses via an equal-loudness-interval assumption, observations from the equissections can be described and extrapolations matched with reasonable accuracy. Loudness ratio productions for each of the same seven subjects also can be described and predicted with sometimes surprising accuracy by relaxing the usual assumption that listeners will necessarily adjust the instructed proportions. These findings offer a new resolution for disparities in results obtained with traditional interpretations of the observed behaviors. This theoretical approach, furthermore, proposes absolute loudness measurements for individuals, an explanation for the growth of loudness including the principle underlying Steven's Power Law, and predicts changes in loudness growth patterns with shifts in sensitivity. Finally, this nonlinear-algebraic theory defines a specific association between loudness and products of harmonic distortion.

REFERENCES

In masking-period patterns, the threshold level of triggered sequences of short high-frequency test tone bursts is determined as a function of their temporal spacing throughout the period of a masker. These patterns can be measured, using either low-frequency maskers or medium-to-high-frequency maskers which are modulated in amplitude by low frequencies. Both kinds of patterns show a distinct nonlinearity: an increment of the masker level of 10 dB results in an increment of the test tone level up to 20 dB. Using 30 Hz as masker frequency [1] or as modulation frequency of a 1-kHz masker [2], respectively, leads to results which are redrawn together in Figs. 1a and 1b. The masker-test signal configuration is indicated on top of both figures. Masking-period patterns produced by low frequency maskers show two maxima, a higher one near the rarefaction maximum and a lower one near the condensation maximum of the masker. Those patterns produced by AM-maskers show only one maximum near the maximum of the masker's envelope.

Fig. 1c suggests quite similar characteristics for all three types of maxima from Figs. 1a and 1b as well as for continuous maskers. Thus, the same basic mechanism may be relevant, regardless the time function of the masker, even though the low-frequency masker acts as a slowly changing DC-masker while the 1-kHz masker obviously acts as an AC-masker.

Zwicker has demonstrated that the threshold for a high-frequency test-tone burst in the presence of a continuous low-frequency masking tone is a complicated function of the phase of the masker at which the high-frequency burst is presented (Zwicker, 1976). "Masking period patterns" measured in this way appear to show nonlinear effects in that at high masker levels the threshold of test-tone bursts reaches local maxima at two different phases of the masker.

We have attempted to model these psychophysical data in a model for motion of the basilar membrane (Hall, 1977). The model consists of a nonlinear mechanical system followed by an additional stage of frequency selectivity ("second filter"). The output of the model is applied as input to a threshold-level detector.

With this model it is possible to reproduce Zwicker's major effects. Masking period patterns are interpreted as a manifestation of two-tone suppression (Abbas and Sachs, 1976). In addition, the phase information present in Zwicker's masking period patterns is shown to impose non-trivial constraints on the form of the nonlinearity and on the nature of the neural excitatory signal. Some electrophysiological studies are suggested.

REFERENCES

INTRODUCTION

Hearing sensations with static character like loudness of steady state sounds can be successfully described and calculated on the basis of their masking patterns (cf. Zwicker [1]). Whether or not this principle can be extended to hearing sensations with dynamic character like subjective duration and roughness is discussed below.

EXPERIMENTAL RESULTS

The fact that the subjective duration of short impulses is relatively overestimated (Zwicker [1]) can be described by means of correlated temporal masking patterns (Fastl [2]). Fig. 1 illustrates the model proposed: from the temporal masking pattern of an impulse of duration \( T_i \), the duration \( T_i^* \) is derived, corresponding to the subjective duration of the impulse. Applying the model, subjective duration of broad band noise impulses can be calculated with an accuracy of typically 10% (cf. Fastl [3]).

The roughness \( R \) of sinusoidally amplitude modulated broad band noise can be estimated on the basis of correlated temporal masking patterns as follows:

\[
R \sim \Delta L_T \cdot f_{mod}
\]

where \( \Delta L_T \) represents the level difference between maximum and minimum of the temporal masking pattern and \( f_{mod} \) the modulation frequency. Estimated roughness values are in good agreement with data of control experiments (cf. Fastl [4]).

CONCLUSION

Three-dimensional transient masking patterns (Fastl [5]), i.e. masked thresholds of test tone impulses as a function of both frequency and time, obviously represent a common starting-point for the description and calculation of hearing sensations with static as well as dynamic character.

REFERENCES

A PSYCHOPHYSICAL DEMONSTRATION OF DUPLEX PROCESSING IN THE HUMAN AUDITORY SYSTEM

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INTRODUCTION

The masking of tone bursts and phase modulated signals by pure tones was studied in human observers. Careful selection of signal parameters disclosed two classes of "internal" auditory spectra having different maxima and "integration" times. An analogy is drawn between the characteristics of these "internal" spectra and various aspects of cochlear anatomy and physiology.

THEORETICAL BASES

Von Békésy (1960) found two different vibrational modes occurring above and below the traveling wave maximum in the cochlea. Furthermore, the microphonic output was shown to be differentially sensitive to these two modes of vibration at the inner and outer boundaries of the tectorial membrane; i.e. above the inner and outer hair cells.

EXPERIMENTAL TECHNIQUES

Continuous sinusoids were adjusted to mask a repetitive signal burst presented at fixed sensation levels. Masking contours were established for signals which varied in level, frequency, duration and spectral composition.

RESULTS AND CONCLUSIONS

A brief high frequency signal (e.g. 4 periods at 6.0 kHz) presented at a sensation level (S.L.) of 20 dB appears to have its energy maximum well below that of the nominal center frequency (C.F.). In addition, the energy in the signal is not fully integrated. Increasing the duration of the signal moves the apparent energy maximum progressively in the direction of the nominal C.F. At approximately 128 periods, the signal appears to be fully integrated and its apparent energy maximum coincides with the nominal C.F.

By decreasing the sensation level of this signal (4 periods at 6.0 kHz), the apparent energy maximum also moves toward the nominal C.F. At a S.L. of 3 dB, the signal has its energy maximum at or above the nominal C.F. and the signal appears to be fully integrated after only 4 periods.

The quantitative differences between these two classes of auditory spectra are greatest at high frequencies and are less obvious at lower frequencies, becoming undetectable below 250 Hz.

It is speculated that the inner hair cells are most sensitive to longitudinal motions of the basilar membrane and have their maximum output at a frequency below that of the outer hair cells which are most sensitive to radial motions. It is further hypothesized that at high sensation levels the signal is being processed by the inner hair cells, which show a relatively long (e.g. 20-40 ms) integration time. At low sensation levels, however, the signal is processed by the outer hair cells which, acting as a delay line, process the signal energy much more rapidly (e.g. 0.5-2 ms).

REFERENCES

MASKING RELEASE AND REMOTE AMPLITUDE CHANGES IN RESONANT FREQUENCY DISCRIMINATION

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Comparisons of diotic and dichotically split sounds suggest variables that normally influence discriminability of speech spectra. We know that upward spread of masking can be released dichotically DMR (2, 3). But what is the temporal extent of the masking? We delayed the excitation of Fp with respect to Fy by half the fundamental period (3.3 msec) and measured the discrimination of small changes in vowel quality, looking for an asynchrony masking release, (AMR). F2 changes showed slight AMR (1). The present experiment (Table 1) shows that for F3 changes, there is no general AMR. The large AMR of F2 here could be due to greater (5 msec) temporal separation of the intense portions of the waveforms for the two formants. Here the output of F3 was also LP filtered and that of F2 HP filtered at 36 db/octave to limit acoustical interaction. This improved performance absolutely and did not remove the AMR, showing that for F2, AMR is indeed psychoacoustic, not a stimulus artifact. Hence in the normal ear, masking of upper formants by lower is shortlived, and when two formants are close in frequency, separating them in time or space improves the discrimination of the upper one.

But is formant interaction ever beneficial? Expt.2 investigated discrimination of F2 and F3 changes in /i/ vowels as a function of ear-formant arrangements. Comparison of the underlined figures shows insignificant DMR of F2 from F3, but massive DMR for F3 from F2. The + figures show that the lower formant benefits from diotic serial presentation, confirming the effect for F1 in Table 1. This suggests that correlated amplitude changes at higher frequencies can signal low formant frequency changes but not vice versa. The + figures suggest that grouping one formant with a formant of remote lower frequency leaves a poor performance for the isolated formant, which may be heard in non-speech mode. A single upper formant is adequate for /i/. Consonant identification experiments (4) with normal F2 levels have confirmed both this "naturalness" effect and the frequency-dependence of DMR. The effect agrees with results showing a preliminary analysis of the incomplete spectra in each ear/hemisphere system (5, 6). While a common fundamental is sufficient for subjective fusion it may not be sufficient for speech-mode analysis.

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INVESTIGATION OF MONOAURAL PHASE EFFECTS IN TWO-TONE COMPLEXES USING THE PULSATION-THRESHOLD METHOD

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One part of our investigation of monaural phase effects /1/ consists of the examination of octave complexes. We measured the phase-dependent pulsation-threshold of the octave tone (400 Hz) of a phase-locked two-tone complex 200 Hz/400 Hz. Fig. 1 shows a typical result for one of our subjects. The level $L_T$ of the test tone (400 Hz) at the pulsation-threshold is plotted as a function of the zero-phase angle $\phi$ of the octave tone in the octave complex which is used as the masker. The parameter at the curves is the level difference $\Delta L = L_o - L_i$, where $L_i, L_o$ are the SPLs of the fundamental (200 Hz) and the first harmonic (400 Hz).

The experimental results can be summarized as follows:
- The fundamental in the octave complex has a phase-dependent effect on the octave tone, which can be described as a phase-dependent two-tone suppression. This yields a maximal and a minimal pulsation-threshold level $L_{\text{MAX}}$ and $L_{\text{MIN}}$, if $\phi$ is altered by $2\pi$.
- The variation width $L_{\text{MAX}} - L_{\text{MIN}}$ of a curve depends on $\Delta L$ and it reaches a maximum for $\Delta L = -10$ dB. It is remarkable, that the 'best beats' of a mistuned octave complex are found at similar level conditions /2/.
- The phase angle $\phi_{\text{MIN}}$, belonging to the minimal test tone level $L_{\text{MIN}}$, depends on the levels $L_i, L_o$ of the two-component masker. It can be seen from Fig. 2, that $\phi_{\text{MIN}}$ increases, if the level difference $\Delta L$ decreases.

We propose a model, that reproduces a phase-dependent pulsation-threshold level of the octave tone in an octave complex. This model takes into account the non-linear behaviour of the ear and the half-way-rectifying property of the mechanical-neural transduction.

The supposition, that nonlinear damping losses are involved in the movement of the basilar membrane, leads to an explanation of the level-dependent $\phi_{\text{MIN}}$-shift.

INTRODUCTION
The discrimination ability of the auditory organ may be characterised by means of the standard deviation \( \sigma = F(f, T) \) of frequency values of the comparison tone in estimating the pitch of short tone pulses by matching.

Doughty and Garner (1) experimentally investigated \( \sigma \) as a function of frequency \( f \) and duration \( T \). From results of these experiments it may be deduced the relation
\[
\sigma(t_n, t_m(f)) = 4.4 \times 10^{-2} f
\]
where \( t_n \) is another characteristic of the auditory organ, the duration threshold for tone-pitch, defined as the minimal duration of tone pulse for which the pitch character of perception dominates over the noise one. An attempt was made to verify eq. (1) experimentally.

EXPERIMENTAL TECHNIQUES
The experiments were carried out by the method of matching on two normal hearing subjects. Two sine-wave generators (Brüel & Kjaer) were used, one producing the sinusoidal signal for test pulses and the other one for comparing pulses.

Both pulses passing a matching network were conducted to an earphone (Beyer DT 96). The duration of the test signal was just equal to the duration threshold for tone-pitch \( t_m/f \) at the given frequency. Then interpulse interval of 500 ms followed, after that the comparison tone of duration 250 ms came on. This sequence of the pair pulses repeated itself every three seconds.

When the subject had matched the standard and comparison tones in pitch, gave an optic signal to the operator, who recorded the frequency of the comparison tone pulse as the point of subjective equality. One value of \( \sigma \) was statistically determined from 36-40 measurements.

RESULTS AND CONCLUSIONS
The results of our experiments are shown in Fig. 1. We may note the following points:

1. \( \sigma \) increases with increasing \( f \) of the tone pulses, the duration of which is equal to \( t_m \).
2. We can see the trend toward the \( \sigma/f = \) constant. However, this constant is not equal in the case of each subject.

If the ratio \( \sigma/f \) is independent on frequency, it may be determined the number of discriminable tonic signals and so computed their specific information content (2).

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INTRODUCTION

Si nous analysons les surpressions et dépressions originées par la composition de deux ondes sonores, nous pouvons faire une relation avec les sensations auditives et formuler un postulat simple.

Tout ça nous origine un modèle d’audition, qu’en outre d’expliquer les sensations sonores produites par les doubles sons, les sons différentiels et résiduels n’ont pas un caractère purement psychique. Il y a, alors, une base physique capable d’interagir avec l’appareil auditif aux niveaux mécaniques et physiologiques du tympan et de la membrane basilaire.

BASES TEORIQUES

Quand \( f_1(x-vt) \) et \( f_2(x-vt) \) sont deux mouvements ondulatoires sonores, de fréquence respective à \( V_1 \) et \( V_2 \),

\[
P(t) = \int (f_1(x-vt)+f_2(x-vt)) \, dt
\]

nous montre que, principalement, il y a des surpressions-dépressions de fréquence \( V_1 \), de fréquence \( V_2 \) et de fréquence \( V_2-V_1 \).

OBSERVATION AUDITIVE

Nous observons que les sensations auditives relatives aux doubles sons de caractère stationnaire, ont une relation de logique sensitive avec les calculs que nous exprime le paragraphe antérieure.

Si nous écoutons soigneusement un double son de fréquences \( V_1 \) et \( V_2 \), autant la sensation de fréquence \( V_1 \), comme la de \( V_2 \) et aussi la sensation de fréquence \( V_2-V_1 \) qui très souvent apparaît, ont une relation formelle avec la formule exprimée dans le paragraphe "BASES TEORIQUES".

POSTULATS ET CONCLUSIONS

Alors, nous postulons que dans la composition de deux ondes sonores de fréquences \( V_1 \) et \( V_2 \), plutôt que de faire un analyse de Fourier, ce qui fait l’oreille, c’est percevoir à la fois des surpressions-dépressions de fréquences \( V_1 \), \( V_1 \) et \( V_2-V_1 \). Tout cela d’accords avec les calculs conséquents des pressions et dépressions originées par la composition de deux ondes sonores, d’après le Principe de Superposition d’Ones.

Parce que, il y a une base physique de surpressions-dépressions de fréquences \( V_1 \), \( V_1 \) et \( V_2-V_1 \), nous admettons son interaction mécanique et physiologique au niveau du tympan et membrane basilaire. En conséquence, les sons différentiels et les sons résiduels n’ont pas un caractère subjectif et sont apperçu par l’organe auditif d’après le procès ordinaire.

- Si nous analysons un phénomène avec un schéma fixe, tout qui n’est pas adapté au schéma court le péril d’être un pur produit d’imagination.
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INTRODUCTION

L'effet de Doppler lié à l'observation du son d'une source mobile est intéressant aussi du point de vue psychoacoustique. La première étape de l'analyse fait penser au problème de la formation de l'impression de hauteur du signal monochromatique émis par une source mobile se propageant avec une vitesse fixe par rapport à l'observateur.

PERCEPTION DE LA HAUTEUR ET DE L'INTENSITÉ DU SON VARIABLES DANS LE TEMPS ET SE PRODUISANT ENSEMBLE D'UNE SOURCE MOBILE

On peut distinguer deux aspects différents dans les recherches sur la perception par l'observateur du signal acoustique provenant d'une source mobile, c'est-à-dire: le changement de fréquence et le changement d'intensité (Fig.1). La complexité du problème se révèle au moment où l'on étudie ces grandeurs ensemble ainsi que leur corrélation réciproque. Ceci a une signification essentielle pour évaluer la formation de l'impression acoustique ainsi que d'une éventuelle possibilité de distinguer le rôle décisif d'une des grandeurs mentionnées.

RÉSULTATS - CONCLUSIONS

Concentrés sur l'évaluation des changements de la hauteur nous pouvons distinguer certains intervalles de temps, particulièrement "la zone de croisement" de l'observateur par la source mobile. Suivant la vitesse du mouvement de la source et l'éloignement de trajectoire du mouvement de l'observateur - l'effet de changement de fréquence se fait par sauts ou bien d'une manière continue par rapport aux possibilités potentielles de la perception et de la formation de l'impression sonore par l'organe de l'ouïe. Il est essentiel alors d'étudier "des seuils différenciels dynamiques" de la hauteur, c'est-à-dire des changements de cette grandeur simultanément au changement continu de l'intensité prenant en considération la constante de temps de l'oreille.

RÉFÉRENCE

APERCEPTIVITE DU CHANGEMENT DE LA STRUCTURE SPECTRALE DU SON SE PROPAGEANT DANS L'INTERIEUR

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INTRODUCTION

L'influence des propriétés acoustiques de l'intérieur sur le changement de la structure spectrale s'y propagant - fut étudiée du point de vue physique (1). Généralement observée une certaine divergence de l'évaluation objective et "subjective" dans l'acoustique architecturale nécessite d'introduire dans cette évaluation des éléments psychoacoustiques. Ceci a déjà permis de déterminer des changements de son provoqués par la déformation des signaux acoustiques se propagant dans l'intérieur (2).

APERCEPTIVITÉ DES CHANGEMENTS DE RELATIONS RÉCIPROQUES DES AMPLITUDES COMPOSANTES

Les recherches présentes ont pour but d'entrer plus profondément dans la structure spectrale des signaux acoustiques étudiés et de déterminer la possibilité d'aperceptivité des changements survenus par rapport aux composantes isolées.

Des essais entrepris pour évaluer l'aperceptivité des changements de relation réciproques des amplitudes des composantes spectrales du son étudié (comme première étape de recherches) ont exigé avant tout de la transformation du spectre physique en spectres du genre "des spectres perceptuels" (131,141) prenant en considération des éléments de mécanisme du fonctionnement de l'organe de l'ouie.

Des seuils différenciels employés pour des changements d'amplitude rapportés à la fréquence et au niveau de perception, ont permis de distinguer des changements perceptibles d'amplitude pour chaque composante du son physiologiquement évaluée.

RÉSULTATS - CONCLUSIONS

D'obtenus résultats théoriques permettent d'affirmer que les changements d'amplitudes de certaines composantes se font dans les limites d'aperceptivité (Fig.1) ce qui mène à percevoir par l'observateur des changements de relations réciproques des amplitudes et d'influer la perception générale d'un des éléments du timbre c.-à-d. du changement qualitatif du son se propagant dans l'intérieur.

RÉFÉRENCES


Fig.1. Les différences aperceptibles du niveau des intensités des composantes du son complexe relativement aux critères admis.
AUDITORY SENSITIVITY TO TRANSIENT FREQUENCY CHANGE

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INTRODUCTION
The thresholds of transient frequency change in FM sinusoidal tone with some typical patterns of the change were measured through the perception of dynamic pitch change. A functional model of detection mechanism of frequency change was presented.

EXPERIMENTS.
The thresholds of frequency change (Δf) for each initial frequency (fi; 250, 1k, 4kHz) of FM tone were determined by means of 2IFC method in which FM tone and pure tone in pairs were successively delivered to the subjects through the binaural head phones. The pitch of pure tone was matched with the over all pitch of FM tone, so that the subjects could pay attention exclusively to the dynamic pitch change. The patterns of frequency change were as follows: rising step at a middle portion of tone burst, convex triangle and rising ramp over the whole extent of the burst (20 - 300 msec.), and rising ramp at a portion of the burst (500 msec.) having various combinations of the onset delays of frequency glide from tone onset (T1) with the durations of frequency glide (T2).

RESULTS AND CONCLUSIONS
It was found that one might be harder to detect the change according to the order of the pattern of step, ramp, and triangle and further found that in spite of the patterns used the thresholds reached to the constant value above the duration of 100 msec. when fi was 4kHz. Fig.1 shows the experimental results for rising ramp at a portion of the burst. Nearly constant thresholds were also obtained here as far as W (=T1+T2/3) held constant as reported in the previous paper (1). Applying the simple weighting, deduced from the shape of the thresholds of roughness by Terhardt (2), to the model shown in Fig.2, the correspondence between the theoretical results and the experimental ones was good.

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(2) Terhardt, E., Acoustica, 30 (1974) 201
INTRODUCTION

At the 8th ICA we outlined a new research program whose objectives include quantifying the effects of hearing loss on soldiers' performance of combat-relevant skills (1). It was shown that an important type of soldier performance involves detecting and identifying the sounds resulting from personnel movement and personnel activity: such sounds, if heard, may betray the presence of enemy personnel. Moreover, typical patterns of noise-induced hearing loss seem to coincide with the frequency bands where personnel sounds have their maximum energy. It therefore appeared likely that hearing losses could significantly affect soldiers' ability to detect and/or identify personnel sounds.

DEVELOPMENT AND VALIDATION OF A DETECTION MODEL

A detection model has been developed (2) which takes into account two basic properties of the auditory system, viz., its frequency sensitivity by critical bands, and its ability to integrate energy for periods of up to 200 msec. To validate this model, we developed a computer system to analyze test sounds; this system performs a Fourier analysis on 20-msec segments of the sounds, combines the results into critical bands, integrates for 200 msec, and then searches for the 200-msec period containing the most energy. Subjects' hearing thresholds were measured using 200-msec tones at the center of each critical band. The audiograms are then compared with the sound spectra to obtain a prediction of relative detectability.

24 test sounds were selected, recorded and analyzed: walking on various types of terrain, movement through foliage, use of simple tools, handling of weapons, etc. Their spectral shapes included rising, falling and peaked types. The test sounds were presented to 20 ears representing a wide range of hearing sensitivities. Correlation coefficients computed between the predicted and actual detection thresholds ranged from .89 to .98 with a mean of .94 for the 24 sounds. Thus, we concluded that the model predicts detectability quite well.

EXTENSION OF DETECTION MODEL TO MASKED LISTENING

The above tests were conducted in quiet. However, even under the quietest of outdoor listening conditions there is some background noise present (3). So, tests are presently in progress to extend the detection model to masked listening conditions, using white noise and realistic background noises (wind, rain, running water, etc.). These tests will also consider the fact that background noises are rarely steady-state; rather, they vary in level and possibly in spectrum.

FUTURE TESTING: SOUND IDENTIFICATION

Once the model has been extended to cover detection under typical listening conditions, the program will be broadened to include sound identification. A different testing paradigm will be required for this aspect, and various alternatives are being considered.

REFERENCES

A NEW METHOD FOR DETERMINATION OF FREQUENCY SELECTIVITY
IN POST-STIMULATORY MASKING

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INTRODUCTION

The new method of measurements used in determining the slopes of upper flanks of the psychoacoustical tuning curves is described.

EXPERIMENTAL PROCEDURE

Contrary to the common practice subjects task was to detect the pitch interval between the masker and the maskee. Three very experienced musicians served as subjects in the experiment. An adaptive-two-alternative-forced-choice procedure was used. Time paradigm was 1000 ms masker, 1 ms break, 25 ms maskee-envelopes rectangular.

RESULTS

The slopes of upper flanks thus obtained reach from $2 \cdot 10^5$ to $2.5 \cdot 10^5$ dB/oct in the frequency range 1 to 3 kHz i.e. more than in Vogten's /1974/ and authors previous /1976/ reports. The experimental data contain $1.4 \cdot 10^4$ individual judgements. Control experiments show that substantial pitch shift occurs also at 15 dB masking so that $L_w - L_m$, resulting in transformation of relatively small frequency intervals into much larger pitch intervals. Probably these enlarged pitch intervals were used by subjects in the detection tasks. The transformation ratio from frequency to pitch intervals as found from control experiments data amounts from about 1:3 to 1:5 in the frequency range investigated and at $L_m=15 \text{ dB}$.

REFERENCES

Binaural release from masking can improve the audibility of tones in noise by as much as 15 dB. Binaural gains in speech intelligibility correspond to increases in signal-to-noise ratio of 13 dB. Binaural unmasking can also reduce apparent reverberation.

These binaural effects of suppression of unwanted signals or noise can be explained by an "equalization and cancellation" (EC) theory (Durlach, 1972). The EC theory says, in effect, that at each frequency - relative amplitudes and phase angles are adjusted by man's binaural processor in such a way that, upon subtraction, the signal-to-noise ratio (SNR) is maximized. (Small errors in the EC process limit the SNR improvement to about 15 dB - even if the noises at the two ears are identical.) While the EC theory accounts for a large body of experimental data, the postulated phase shifters or delay lines have never been found in the brain. In this paper it is shown that the necessary differential gains and delays could easily be supplied by what is already available in the auditory periphery, namely the basilar membrane in the inner ear.

Sound waves striking the eardrums set up travelling waves on the basilar membrane (BM) in the inner ear. A given frequency component will peak at its "characteristic place" and then decay again. The lower the frequency, the further down the BM the peaking will occur. And, of course, there is increasing delay with travel along the BM. The delay increases first exponentially and then linearly up to several tens of milliseconds at the far end of the BM. The total phase shift for most frequencies is of the order of 30 radians, corresponding to a phase delay of 10 msec at 500 Hz - most of it occurring near the characteristic place.

Thus, right on the BM, near the characteristic place for each frequency - i.e., near its place of detection by the hair cells - the ear has a range of delays and gains to "play with". The available delays within the passband of each place are about $5/f_r$, where $f_r$ is the resonance frequency of the place. If the brain forms binaural differences not only from places with the same characteristic frequency, but neighboring places as well, then many binaural effects are easily accounted for. In fact, one can make the argument even stronger: it would be quite unreasonable to assume that binaural differences are formed exclusively from places with exactly the same characteristic frequency. Thus, binaural gain and delay differences are almost unavoidable at the point of binaural interaction.

If the gain and delay differences needed for binaural signal enhancement come predominantly from the basilar membrane, then there is a predictable limitation to what the ear can do: gain and delay differences cannot be chosen arbitrarily. While for equal gain a large range of delay differences can be realized (up to about 10 msec below 1 kHz) by selecting points of equal gain on the BM filter curve (one above and one below resonance), the converse is not true: For zero delay difference, one must go to the same point on the filter curves and there is thus no gain difference. To summarize: equal gains - a range of delays possible, equal delays - no gain differences possible. Thus, if the noises at the two ears are in phase, but have different amplitudes, cancellation should not work as well as when amplitudes can be adjusted independently of phase. This is precisely what was found by Durlach when he tried to apply his EC theory to in-phase noises of different amplitudes and could not explain the experimental results. ("In general, the model has very little to contribute to an understanding of these results." op. cit., p. 411.)

What is needed is an EC theory in which amplitude and phase differences cannot be chosen completely independently but are constrained. I submit that the basilar membrane is a good place to look for these constraints. New binaural experiments, designed and interpreted according to the hypothesis presented here, could yield important information, otherwise not available on the transfer functions of the living human ear.

Presenting alternately pulses of a masker tone to one ear and pulses of a test tone to the other ear, the fainter tone-pulse train (of the test tone) will be perceived as a continuous tone for appropriate signal parameter [1] (similar to monaural pulsation threshold measurements [2]). Fig. 1 shows the level $L_t$ of a 1000 Hz test tone pulse (arrow) at the transition from continuous to pulsating sensation, called the dichotic pulsation threshold (DPT). The DPT is represented as a function of the masker tone frequency $f_m$ for 5 fixed levels $L_m$ of the masker, so as to allow a comparison with response areas of single auditory neurons [3]. For levels below the DPT, the test tone pulse train is perceived as a continuous tone. The course of DPT as shown in Fig. 1 is always very broadly tuned and reveals a strong nonlinearity for increasing masker level. In the case of $f_m = f_t$, there is some indication of a saturation similar to the saturation of firing rate of single auditory neurons.

Further results of experiments with broad-band noise superimposed on the test tone pulse [2] can most parsimoniously be accounted for by the rate characteristic of the activity of single auditory neurons under comparable stimulus conditions; there is no relation of DPT to the loudness of the signals, in contrast to the case of monaural pulsation thresholds ([1], [4]).

It is inferred, that the DPT may be a direct measure of the firing rate of some single representative neuron at the lower level of the auditory pathway.


VERSUCHE ZUR VERBESSERUNG DER LOKALISATION DER SCHALLQUELLEN DIE SICH VOR DEM KUNSTKOPF BEFINDEN

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Der erste Versuch war die Lokalisation des Sprechers (Feststellung der Hörereignisrichtung und Entfernung) beim unmittelbaren Zuhören. Das Experiment fand in einem großen Laboratorium mit der Nachhallzeit von etwa 1,2 s statt. Die Augen der Versuchspersonen waren gebunden und die Bewegung des Sprechers in dem Saal war durch lautes Rauschen verdeckt. Die Versuchspersonen wussten, dass nur acht, sich um 45° unterscheidende Richtungen, möglich waren, sowie nur drei verschiedene Entfernungen: "nahe", "unweit" und "weit", die etwa 1m, 3m und 5m bedeuteten. Zehn Versuchspersonen haben 99% richtige Antworten gegeben.

Der selbe Versuch wurde mit einer Kunstkopfaufnahme und nachherigem Anhören über Kopfhörer wiederholt. In diesem Falle wurden Aufnahmen in drei Sälen durchgeführt, und zwar in dem erwähnten Laboratorium, in einem kleineren aber halligen Saal und in einem grossen, ebenfalls halligen Hörsaal. Die Resultate von zehn Versuchspersonen waren nicht für die drei Säle viel unterschieden: in der vorderen Raumhälfte (3 Richtungen) hat man 15% bis 30% genaue Antworten bekommen, weil das Prozent für seitliche und hintere Richtungen auf 50% bis 75% stieg. Mehr als eine Hälfte von Versuchspersonen hat den sich vorne befindenden Sprecher als "hinten" gehört.

Die Hörereignisentfernung wurde mit etwa 70% genau geschätzt und keine besondere Abhängigkeit von der Richtung wurde bemerkt.


Der selbe Versuch mit etwas verbesserten Resultaten für die Richtung "vorn" wurde nachher mit dem Kunstkopf vorgenommen, aber anstatt von Mikrophonen in den Ohren hat man Hörschläuche und Stethoskop verwendet. Die Versuchspersonen (9) waren in einem Nebenraum isoliert.

In allen vorliegenden Versuchen hat man einen Kunstkopf aus steifem Gipsmaterial verwendet. Nachdem man die Ohrmuscheln aus elastischem, vibrierendem Material gemacht hat, wurde die Hörereignisrichtung "vorn" sehr viel besser wahrgenommen. Die Versuche sind jetzt im Laufe, und an dem Kongress wird man genaue Resultate mitteilen und Tonbandaufnahmen vorführen. Als Beispiel kann man erwähnen, dass schon mit den Ohrmuscheln aus Karton in der Form eines Halbkreises drei Versuchspersonen in keinem Fall die Richtungen "vorn" und "hinten" vertauscht und keine IKL bemerkt haben.

Schrifttum:
LOCALIZATION OF THE SOUND SOURCE INDOORS UNDER DIFFERENT ACOUSTICAL CONDITIONS

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INTRODUCTION

It is possible to define accurately, for each room, the characteristic fields where the direct and the reverberant sounds are in a definite ratio. The final subjective impression, at direct listening as well as at listening to the recorded sound through the artificial head, depends much on the field wherein the sound receptor is located.

THEORETICAL BASES - DETERMINATION OF THE CHARACTERISTIC FIELDS

Indoors, the localization of the sound source depends on the distance between the listener and the sound source. In this paper we shall analyse only the influence of characteristic fields on the localization. Therefore we must calculate the distance \( r \) when the direct and the reverberant sounds are equal:

\[
r = \frac{0.141 \sqrt{A}}{\sqrt{1 - \bar{\alpha}}}
\]

where \( A \) is the entire absorption of the room and \( \bar{\alpha} \) - the mean absorption coefficient of the material sheathing the room.

EXPERIMENT

The artificial head was located at three selected distances from the sound source, representing three characteristic fields. These distances were 80 cm, 2.25 m and 8 m, because the value \( r \) varied between 2.18 and 2.42 m on frequencies from 125 Hz to 4000 Hz, in the Belgrade Philharmonic Hall where experiments have been carried out. The angles, with respect to the sound source - listener axis, have changed by 30°, in total 12 times. The sound source was emitting groups of nonsense words and octave band white noise.

RESULTS AND CONCLUSIONS

At direct hearing there was no doubt about the sound source location, under the above described conditions. But when listening through the stereophonic headphones to the recorded material, the positions of the sound source were sometimes wrong. When the distance from the artificial head to the sound source was 2.5 m or greater, the location was more difficult.

The shortest conclusion is: if we want to record a signal indoors through the artificial head and have, at the same time enough information on the sound source position, the distance between them must be less than the radius \( r \).

REFERENCES

An analytical interaural time difference (ITD) model is derived on the basis of diffraction by a rigid sphere. This model is confirmed by ITD data measured on a manikin's head with and without a torso.

The analytical ITD, derived from scattering theory for a rigid sphere, when \((ka)^2<<1\) is shown to be:

\[
ITD = 3 \left( \frac{a}{c_0} \right) \sin \theta_{inc} \quad (1)
\]

where \(c_0\) is the ambient speed of sound in the medium, \(\theta_{inc}\) is the angle of incidence, \(a\) is the radius of the sphere, and \(k\) is the acoustic wave number. From "creeping wave theory," the ITD for \(ka>>1\) is shown to be:

\[
ITD = 2 \left( \frac{a}{c_0} \right) \sin \theta_{inc} \quad (2)
\]

for \(0^\circ \leq \theta_{inc} \leq 60^\circ\). Normalizing Eqs. (1) and (2) yields a nondimensional parameter:

\[
\Pi = \frac{ITD}{(a/c_0) \sin \theta_{inc}} = \text{constant} \quad (3)
\]

This constant is equal to 3.0 and 2.0 in the low- and high-frequency limit, respectively.

EXPERIMENTAL TECHNIQUES

The ITD was measured using a manikin's head and torso under plane-wave incidence in an anechoic room.

RESULTS AND CONCLUSIONS

Measured and predicted values of \(\Pi\) are shown in Fig. 1 for \(\theta_{inc} \leq 60^\circ\).

Fig. 2 shows the average (measured) ITD between 80 Hz and 500 Hz versus \(\theta_{inc}\). The low-to-high-frequency ITD ratio is 3/2. The measured ITD is a minimum around 1500 Hz for \(\theta_{inc} \leq 60^\circ\). Below 500 Hz the ITD is relatively independent of frequency but larger than reported elsewhere (1,2).

REFERENCES

A CONSIDERATION OF DISTANT PERCEPTION IN BINAURAL HEARING

Sakamoto, N.
Gotoh, T.
Kimura, Y.
Kurahashi, A.

Acoustic Research Laboratory,
Matsushita Elec. Ind. Co., Ltd.
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INTRODUCTION

The factors in reflected sound affecting a distant perception of sound in binaural hearing were examined with simulated reflections by using loudspeakers in an anechoic chamber. As a result, the effective reflected sound structures to get a maximum feeling of distance while a spatial impression set minimum have been found.

THEORETICAL BASES

An acoustic energy density ratio of reflected sound to direct sound was found to be one of the important factors which caused "out-of-head" localization (i.e. distance) in binaural headphone listening (1). However, the parameters in reflected sound structure such as the delay time of initial reflected sound, reverberation (decay) time, and directions of reflected sound paths against a listener seem to be greatly contributing to a distant perception of sound. In this case we conducted psychological experiments by controlling the parameters within the range where a spatial impression remains minimum.

EXPERIMENTAL TECHNIQUES

The block diagram of the experiments is shown in Fig.1. A listener with his head fixed and his eyes blindfolded sits at the center of loudspeakers circularly and horizontally situated with a radius of 1.5m in an anechoic room. A direct sound is radiated from the front (0°) loudspeaker and single or complex delayed sound paths simulating reflections are given from a pair of speakers placed symmetrically in the directions of +30°, +60°, +90°, +135°. The time intervals between direct and delayed sound were chosen as 5, 10, 15 and 20 ms. A listener was asked to judge which sound is farther in paired comparison.

RESULTS AND CONCLUSIONS

(1) In case of a single delayed sound presented the direction +60° gives the maximum distance. (2) The longer the delay time of the first reflection becomes, the larger the distance grows. (3) Complex delayed sound paths gives more distant feeling than a single.

REFERENCES

(1) N. Sakamoto, et. al., AES Convention Preprint(1975)52nd,1065(E-2)
LOCALIZATION OF SOUND SOURCE SIMULATED BY A DIGITAL COMPUTER

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Ando, Y.

Two systems have been reported as the 2ch. recording-reproduction system for three-dimensional sound localization. One is the dummy head - headphone system(1), and the other is the dummy head - 2ch. loudspeaker system(2). In this paper, the 2ch. loudspeaker system using a digital computer without a dummy head recording is discussed.

Let head-related transfer functions to the entrance of the auditory canal be defined as Fig.1. Then, the signals $P_L(\omega)$ and $P_R(\omega)$ at the two entrances from a source $S$ at any position $(r, \theta, \psi)$ in the three-dimensional space are expressed by

$$
P_L(\omega) = S(\omega) \cdot H_L(\omega; r, \theta, \psi)
$$

$$
P_R(\omega) = S(\omega) \cdot H_R(\omega; r, \theta, \psi)
$$

where $\omega$ is frequency, $r$ is distance, $\theta$ is elevation angle and $\psi$ is azimuth angle. On the other hand, the signals $P_L(\omega)$ and $P_R(\omega)$ at the two entrances from two loudspeakers $L_1$ and $L_2$ located at the positions $(r_1, \theta_1, \psi_1)$ and $(r_2, \theta_2, \psi_2)$ are expressed by

$$
P_L(\omega) = X_1(\omega) \cdot H_{11}(r_1, \theta_1, \psi_1, \omega) + X_2(\omega) \cdot H_{12}(r_2, \theta_2, \psi_2)
$$

$$
P_R(\omega) = X_1(\omega) \cdot H_{1r}(r_1, \theta_1, \psi_1, \omega) + X_2(\omega) \cdot H_{2r}(r_2, \theta_2, \psi_2)
$$

In order to simulate $P_L(\omega)$ and $P_R(\omega)$, according to Eq.(1) and (2), the input signals $X_1(\omega)$ and $X_2(\omega)$ to $L_1$ and $L_2$ are expressed by

$$
\begin{bmatrix}
X_1(\omega) \\
X_2(\omega)
\end{bmatrix} = H^{-1} \begin{bmatrix}
H_{11}(r_1, \theta_1, \psi_1, \omega) & -H_{12}(r_2, \theta_2, \psi_2) \\
-H_{1r}(r_1, \theta_1, \psi_1, \omega) & H_{2r}(r_2, \theta_2, \psi_2)
\end{bmatrix} \begin{bmatrix}
S(\omega) \\
H_{1r}(r_1, \theta_1, \psi_1, \omega)
\end{bmatrix}
$$

where $H = H_{11}(r_1, \theta_1, \psi_1, \omega) \cdot H_{2r}(r_2, \theta_2, \psi_2) - H_{1r}(r_1, \theta_1, \psi_1, \omega) \cdot H_{2r}(r_2, \theta_2, \psi_2)$

Experimental conditions are as follows;
(1) $S(\omega)$ is white noise(300-13600 Hz).
(2) Positions of $L_1$ and $L_2$ are (1.5m, 60°, 90°) and (1.5m, 60°, 270°), where the head-related transfer functions have no remarkable peak and dip.
(3) Simulated directions are every 15° on the horizontal plane and in the median plane.
(4) All head-related transfer functions are measured by a dummy head which is a spheroid with a replica of a personal pinna.
(5) Computation of Eq.(3) is carried out using a digital computer with a AD-DA converter.

REFERENCES
(1) P.Laws & H.-J.Platte:DAGA '75
(2) P.Damaske:JASA 50(1971)

Fig. 1. Definitions of head-related transfer functions.
AN UNSUCCESSFUL ATTEMPT TO REPRODUCE VON BEKESY'S AUDITORY BACKWARD INHIBITION EXPERIMENT

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INTRODUCTION

In a 1971 article (1), G. von Békésy reported on auditory backward inhibition phenomena that he achieved with an experimental set up as shown in fig (1). In a typical experiment loudspeaker A radiates a 35 ms-tone impulse, followed by a 70 ms delayed 35 ms-tone impulse via the circle of loudspeakers B. Both impulses have the same SPL. v. Békésy's subjects recognized a significant masking effect on the first impulse. A backward inhibition effect of this kind would be important in room acoustics.

EXPERIMENTAL TECHNIQUE

We duplicated v. Békésy's set up and experimental technique almost exactly with the exception that our subjects were not trained for their specific task within these experiments, though some of them were quite familiar with psychoacoustic tests in general. The experiments were performed not only under normal "office-conditions" but also in an anechoic chamber. At least 5 subjects took part in each of the tests.

RESULTS AND CONCLUSION

None of our subjects recognized any significant backward inhibition effect, neither under "office conditions" nor in the anechoic chamber. Two sound sensations appeared subsequently, both of about equal loudness. Furthermore, the sound sensations, caused by the loudspeakers circle were not spread out in space as described by v. Békésy (see Fig. 1) but as concentrated as the ones caused by the center speaker. The position of all sound sensations was somewhere in the median plane, depending on the center frequency of the tone impulses.

Keeping in mind that the average users of concert halls have no specific training in backward inhibition experiments as von Békésy's subjects have had, we now have some doubt that backward inhibition really plays the important role in room acoustics that v. Békésy assumed it to do.

REFERENCE

(1) BEKESY, G. von: Auditory Backward Inhibition in Concert Halls Science 171, 529-536
INTRODUCTION:
Recently, an explanation for different dichotic pitch phenomena like Huggins Pitch, Fourcin Pitch and Dichotic Repetition Pitch was proposed by Bilsen (1). He used a model which is based on Jeffress' model of sound localization (2) and is meant to deal with the auditory processing of interaural time (or phase) differences. Anatomically, the model incorporates internal delay lines and coincidence detectors between ipsilateral and contralateral neurons with corresponding characteristic frequencies. Essential assumptions are: sharp peripheral frequency resolution and the existence of coincidence detectors of the EE- and the EI-type. According to this model, the auditory system generates a neural activity pattern with the dimensions: power (neural activity e.g. spike rate), frequency and interaural delay.

GENERAL VALIDITY OF THE MODEL:
In general such a three-dimensional activity pattern can be scanned by the central auditory system and should provide all the information necessary for directional hearing (lateralization), signal to noise ratio improvement (BMLD), dichotic pitch sensations etc.. As an example, the expected activity pattern is given in fig.1 for broad band noise, dichotically delayed over 0.5 ms in which case the noise is localized in a lateral position (lateralization). For an internal delay of 0.5 ms the frequency spectrum is most equal to the original spectrum of the noise, though weighted in frequency. There is a maximal modulation around 500 Hz (dominant region)(3,4), probably due to such imperfections in binaural interaction as the loss in synchrony in the spikes for higher frequencies.

EXPERIMENTS AND CONCLUSIONS:
In order to quantify and refine the assumptions mentioned above, binaural masking experiments have been undertaken using sinusoidal probe signals under different dichotic stimulus conditions to scan the central activity patterns. One series of measurements was carried out using interaurally delayed white noise with various interaural delays. During other experiments dichotic noise was used with interaural phase shifts at harmonically related frequencies. Besides, different extra interaural delays were introduced. Qualitatively, the results under these stimulus conditions turned out to be in good agreement with the expected three-dimensional internal power spectra according to the model proposed.

REFERENCES:
Binaural beats are perceived if a tone (below 1000 Hz) is presented to one ear and a tone of a slightly different frequency to the other ear. The beats result from neural interactions of the two auditory pathways. With added noise, the beats are best perceived if each of the two tones is just above its monaural masking threshold [1].

If the binaural beats are compared with "physical" beats obtained by simply adding the two signals, a phase shift is found between the binaural beats and the physically produced beats. The phase shift depends on the level of the noise and on the level of the signal relative to the binaural masking level (BML). For in-phase noises, the phase shift is π for binaural beats at the BML. A decrease of the noise level shows a splitting of the beats in two components (Fig. 1), corresponding to a lateralization of the perception of the beats. For low S/N-ratios the phase shift of the two components of the binaural beats amounts to 3/4 π and 5/4 π, respectively (Fig. 1). For high S/N-ratios the perception of the two components fuses and measurements of the phase of binaural beats become difficult at all.

In addition to measurements with correlated noise, experiments with uncorrelated and anticorrelated noise are performed. The results are discussed in the light of the equalization and cancellation theory.

REFERENCE

Auto- and cross-correlation are, since 30 years ago, key tools in the study of auditory perception of sound signals and fields, and room-acoustic criteria are due to continue developing on models based on them (1). Contrary to experiments with signals of fixed cross-correlation, those performed with signals of time evolving cross-correlation are scarce (2°). Accordingly, it is presented here a generator that affords two noise signals with close modulated short-term cross-correlation coefficient, along with listening results obtained with such signals.

The generator is based on the particular summation procedure expressed by the following formulation:

\[ S_1(t) = N_1(t) \]
\[ S_2(t) = N_1(t) \cdot \cos \omega t + N_2(t) \cdot \sin \omega t \]

where \( N_1(t) \) and \( N_2(t) \) being two uncorrelated white noise signals of equal RMS value and bandwidth. Consequently \( \text{RMS}[S_1(t)] = \text{RMS}[S_2(t)] \) and \( \phi_{S_1S_2}(0) = \cos \omega t \).

The constructed device allows for a great variation of \( \omega \), as no moving mechanical parts are involved but solid state integrated technology. Only for a stepwise output \( (\omega t = \theta_1) \) a sin-cos step pot is used.

On feeding \( S_1(t) \) and \( S_2(t) \) to the respective deflection plates of an oscilloscope, a movie-like continuous sequence of the IICKLIDER and DZENDOLEFT's (3) scatterplots is visualized, period being equal to \( 2\pi/\omega \).

\( S_1(t) \) and \( S_2(t) \) can be formally considered as a generalization for the signals that give rise to the classical binaural beats involving \( \Delta \omega \) increments equal to \( \omega \), and to the classical binaural localization phenomenon involving time delays or phase differences if we use for \( N_1(t) \) and \( N_2(t) \) orthogonal pure tones and \( \omega t = \theta_1 \). Both phenomena are therefore assignable to a central auditory short-term cross-correlator.

The brain's cross-correlation ability is once more experimentally supported by the dichotic listening test results, that will be presented to complete this summary. Particularly they explain, on the correlation basis, the data reported by ASCHOFF (4) when listening to a circular array of sequentially radiating loudspeakers. GRUBER's "Korrelationschwebung" (2") is similarly present with these new signals, IIX, and the limits for its occurrence and subjective features, as a function of \( \omega \), shall also be given.

References:
(1) W. de V. KEET,- VI ICA, Tokyo 1968, paper E-2-4.
(4) V. ASCHOFF,- 132 Sitzung Arbeitsgemeinschaft für Forschung NFW, Düsseldorf 1964, p 23.
INTRODUCTION

Les résultats des recherches sur la latéralisation en présence de signaux binauraux à grande différence de niveau sonore publiés jusqu'à présent peuvent être présentés ainsi: l'augmentation de l'IAD entraîne un agrandissement des dimensions de l'image sonore subjective et une perte de précision des contours, liée au déplacement [1,2]. Si l'on augmente encore l'IAD l'image subjective se latéralise entièrement et ne se distingue plus d'une écoute monaurale. Parfois cependant l'impression se divise entre monaurale et binaurale [1]. L'IAD nécessaire pour une latéralisation complète est de 15-20 dB pour certaines auteurs [3], de 9-10 dB pour d'autres /Pinheiro et Tobin, 1969/.

MÉTHODE

Au cours de nos expériences les auditeurs /qui n'en connaissaient pas le but/ devaient exprimer oralement leurs jugements sur les effets du changement d'intensité dans l'un des écouteurs quand le niveau dans l'autre écouteur reste constant. Ces changements d'intensité ont été effectués pour les niveaux d'écoute commençant même au dessous du seuil d'audibilité. Tous les auditeurs ont constaté qu'il existe un niveau pour lequel apparaissent simultanément une impression monaurale, et, plus près du milieu de la tête, une impression binaurale. Si on augmente encore le niveau, toujours dans la même oreille, les auditeurs, concentrés, peuvent percevoir ces deux impressions séparément. Quand ils sont déconcentrés, ou quand l'IAD est petite, ces impression se joignent en une seule qui est la même que celle décrite par Bekesy et Sayers. Dans la deuxième partie de l'expérience on a émis dans l'écouteur où le niveau était variable un son interrompu, de sorte que les auditeurs aient la possibilité de comparer l'impression binaurale avec une impression réellement monaurale.

RESULTATS ET CONCLUSIONS

Les résultats quantitatifs montrent que l'impression binaurale apparaît quand le niveau d'intensité acoustique dans l'oreille la moins excitée, franchit le seuil de 1-6 dB. On a établi statistiquement que cette dispersion n'est pas significative et que le changement de fréquence et d'intensité du son pur ainsi que l'intensité du bruit dans la deuxième oreille n'influent pas sur les résultats. Par suite, il semble qu'on ne puisse pas parler d'une valeur maximale de l'IAD pour la latéralisation puisque le seuil d'apparition de l'impression binaurale ne dépend pas de l'intensité dans l'oreille la plus fortement excitée. Cette expérience peut être considérée comme une mesure de déplacement du seuil du à un effet de masque central. Dans cette interprétation on peut être surpris par l'indépendance de l'effet de masque contra-latéral.

BIBLIOGRAPHIE

Data on Temporal Threshold Shifts 2 minutes after exposure to impulse noise (TTS)<sub>2</sub> were collected from the existing literature. A hearing loss criterion that best covered the data appeared to be 15 dB TTS averaged across 1, 2, 3 kHz and exceeded by only 10% of the subjects exposed. Peak levels that satisfy this criterion are shown in Fig. 1 as a function of the duration N·τ<sub>10</sub> (N = number of impulses, τ<sub>10</sub> = time interval in which the level of a single impulse declines to 10 dB below its peak level). Linear regression analysis produces a slope of -8 dB per factor of 10 in duration and a regression coefficient of -0.97. The variance remaining with respect to the regression line is 9 dB<sup>2</sup> and could be due to intersubject differences only.

In comparing TTS<sub>2</sub> and Noise Induced Permanent Threshold Shift (NIPTS) the general view is that TTS<sub>2</sub> equals NIPTS (ASA 1951 audiometric zero) after 10 years of exposure to the same noise. Fluctuating noise at L<sub>eq</sub> = 90 dB (A) induces PTS after 10 years of 20 dB (ISO audiometric zero, 11 dB re ASA 1951) averaged across 1, 2, 3 kHz and exceeded by 10% of the subjects exposed. Peak levels for 5, 10 and 25% unprotected at 11 dB TTS<sub>2</sub> were calculated on the basis of the collected data. Figure 2 shows that a smooth transition is found between the damage risk levels for impulse noise and for fluctuating noise on the basis of the above relation between TTS<sub>2</sub> and NIPTS. The inter-subject differences for impulsive noise appear to be greater. Damage risk levels for impulsive noise corresponding to L<sub>eq</sub> = 85 dB(A) are 7.5 dB lower (corresponding criteria are 6 dB TTS<sub>2</sub> or 15 dB NIPTS).

---

**Fig. 1**

![Graph showing TTS<sub>2</sub> levels for 10% unprotected, 15 dB TTS<sub>2</sub> average 1.23 kHz normal incidence.]

---

**Fig. 2**

![Graph showing PTS levels at 25%, 10%, and 5% unprotected at 11 dB TTS<sub>2</sub>, 20 dB NIPTS.]

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E: M.A. Elwood, in:
H1: D.C. Hodge et al., Tech. Memo 15-1964,
P: F. Pfander, Das Knalltrauma (1975), Springer Verlag (Berlin)
INTRODUCTION

The mechanisms responsible for hearing loss to impulsive stimulation are not presently known and until they are, prediction of the hazard from such stimulation will remain uncertain. In order to begin filling this gap the present series of investigations was begun.

It has long been established that the ear differs in its sensitivity to sound as a function of its spectral content. This same characteristic is undoubtedly important in the case of impulsive stimulation as well (1). In addition, there are indications that the mechanisms of hearing loss change as a function of the level of the stimulation (2-5). Experiments were therefore designed to examine losses in hearing as they relate to the spectral content of the impulses and the level at which they are presented with the aim of revealing the basic mechanisms producing hearing loss.

EXPERIMENTS

Adult cat ears were exposed via a closed-tube stimulating system to tone pips whose spectrum was approximately 1/3 octave wide, whose spectral location varied (1.0, 5.0, and 10.0 kHz) and whose level was varied systematically. Pre- and post-exposure measures of hearing included nerve responses to tone pips between 1.0 and 20 kHz and cochlear microphonics, both recorded from the round window in anesthetized acute preparations.

ANALYSIS OF THE DATA

In order to relate the changes in cochlear function measured to the cochlear inputs, the transfer function for the cat middle ear (6) was applied to the input spectra measured at the ear drum. Data will be presented showing changes in sensitivity across frequencies as a function of the spectral content and level of the exposure impulses and these findings will be interpreted with respect to whether the basic mechanisms of hearing loss to impulsive stimulation are displacement or power related.

REFERENCES

Untersucht wurden interindividuelle Unterschiede der Gehörbeanspruchung bei 3 verschiedenartigen Gehörbelastungen und intraindividuelle Schwankungen bei Belastungswiederholungen. Als relevantes Maß für die Gehörbeanspruchung in Hinblick auf die gehörschädigende Wirkung des Lärms wurde das Integral über den zeitlichen Verlauf der TTS bestimmt (1):

\[ S = \int_{t_E}^{t_R} TTS(t) \, dt \]

\[ t_E + t_R = \text{Einwirkungszeit} + \text{Erholungszeit} \]

Es wurden 12 normalhörende Versuchspersonen (VP) je 10 mal folgenden Belastungen ausgesetzt:
1. 30 min, breitbandiger Lärm, \( L_A = 100 \, \text{dB} \)
2. 2 min, breitbandiger Lärm, \( L_A = 100 \, \text{dB} \)
3. 4 Schlagimpulse, Spitzenwert \( L_I = 145 \, \text{dB}, \ t_B = 200 \, \text{ms} \)

Ergebnisse (2):

<table>
<thead>
<tr>
<th>Lärm</th>
<th>( \overline{S} )</th>
<th>( \overline{v}_{\text{intra}} )</th>
<th>( \overline{v}_{\text{inter}} )</th>
<th>( \overline{TTS}_2 )</th>
<th>( \overline{v}_{\text{intra}} )</th>
<th>( \overline{v}_{\text{inter}} )</th>
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</thead>
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<td>1</td>
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<tr>
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<td>33</td>
<td>8,9</td>
<td>32</td>
<td>30</td>
</tr>
<tr>
<td>3</td>
<td>40</td>
<td>62</td>
<td>28</td>
<td>6,4</td>
<td>40</td>
<td>25</td>
</tr>
</tbody>
</table>

\( \overline{S}, \overline{TTS}_2 \) Mittelwerte über 10 Wiederholungen und 12 VP
\( \overline{v}_{\text{intra}} \) Mittelwert der individuellen Variabilitäten
\( \overline{v}_{\text{inter}} \) Variabilität ermittelt an den individuellen \( S_i \) bzw. \( TT S_{2i} \)

Die intraindividuellen Schwankungen der Gehörbeanspruchung \( S \) und der \( TT S \) sind also größer als die interindividuellen Unterschiede.

Es konnte gezeigt werden, daß für \( v_{\text{intra}} \) und \( v_{\text{inter}} \) die Meßunsicherheiten bei den Hörschwellenmessungen maßgeblich verantwortlich sind.

Untersuchungen von RICHARTZ (3) haben gezeigt, daß bei mehrjähriger Lärmbelastung die individuelle PTS stark mit der Gehörbeanspruchung bei Beginn der Tätigkeit korreliert. Die großen intraindividuellen Schwankungen der Gehörbeanspruchung erfordern jedoch, daß die individuelle Beanspruchungsgröße aus mehreren Beanspruchungswerten ermittelt wird.


(2) Kracht, L. et al., Intraindividuelles und interindividuelles Beanspruchungsverhalten des Gehörs infolge Lärmeinwirkungen. Techn. Univ. Dresden 09-26-75

INTRODUCTION

The AP on-effect, in response to tone-bursts, received convincing explanation but no interpretation is available for the so-called off-effect and, apart from the paper by Kupperman (1), the off-AP has never found an extensive experimental description.

Some typical off-APs, in response to a 4kHz tone burst, are reported in fig. 1.

THEORETICAL BASES

Due to the low-pass filtering of the signal which contains a Summating Potential, the upward (positive) deflections seen in the 85 and 65 dB responses most likely have a presynaptic origin. Therefore, they will be ignored in this context. The whole-nerve Action Potential can be conceived as the convolution of many unit potentials; in analytical terms:

\[ AP(t) = \int P_j(t) \cdot u(t-\tau) \, d\tau \]

where the index \( j \) ranges all over the entire set of the cochlear nerve fibres, \( P_j(t) \) is the Post Stimulus Time (PST) Histogram of the \( j \)-th fibre, \( u(t) \) is the single unit action potential and \( \tau \) is a dummy variable. Since no exhaustive description is available for PST Histograms in response to the offset of tone bursts, it is of some interest to consider, as a valid alternative, the PST Histograms obtained in response to a trapezoidal driving of the cochlear partition (2).

In fig. 2 the PST Histograms of three units with low, middle and high Characteristic Frequencies (CF) are shown. The top trace is the Cochlear Microphonic (CM) recorded at the Oval Window.

It is easily seen that the Histograms show well pronounced peaks in correspondence to any polarity changes of the CM.

By making the convolution between the summed firing probabilities (or the Compound PST Histogram) and the single unit potential one could obtain an off-effect with almost the same height on the on-AP (see the 45 dB responses of fig. 1).

If, on the other hand, the human APs are deconvoluted with the single unit action potential, the Compound PST Histograms derived by this means show a well noticeable peak at the offset of the tone-burst, especially at low and middle stimulus amplitudes.

REFERENCES

The Army needs to protect the hearing of its personnel in the same way that a civilian firm needs to protect the hearing of its employees. The Army does, however, present two noise hazards not often seen in civil employment—the very high levels of noise in tracked armoured vehicles, and the impulsive noise from gunfire. Both these hazards arise during peacetime training.

Reduction of noise at source is either impossible or possible only by radical redesign, and some form of hearing protection—ear plugs, ear muffs or noiseexcluding helmet—becomes necessary. The hearing protection may form part of a communication system, which helps to overcome the user's objections to it.

Unfortunately the noise attenuation of the hearing protection is not always adequate to reduce the noise entering the ear to a reasonably low level. This is particularly the case with tracked vehicle noise which has very strong low-frequency components. Attenuation measured in the laboratory, particularly by real-ear attenuation-at-threshold methods, is likely to over-estimate the protection afforded under normal conditions of use. In addition the noise dose may be increased by speech and vehicle noise transmitted through the intercommunications system.

Noise levels at the ear canal entrance, underneath a noiseexcluding helmet, have been measured using miniature microphones. Since these measurements can be carried out in moving vehicles they are much more realistic than laboratory tests, and show noise levels which frequently exceed 100 dB(A). Although they will not replace laboratory measurements, they give a useful check on the overall performance of a system.

The attenuation provided by hearing protection against impulsive (gunfire) noise is also important, and has been measured using both temporary hearing threshold shift measurements and miniature microphone techniques. In the latter case, measurements based on total (A-weighted) energy were preferred on theoretical and practical grounds to measurements based on peak pressures and durations. Effective attenuation for a variety of ear muff types was found to be in the range 20-25 dB, considerably less than some previous estimates.

The effectiveness of hearing protection is seriously degraded by careless or unsupervised use, and by equipment incompatibilities, eg from spectacles, or fabric hoods worn underneath noiseexcluding helmets. On a more hopeful note, there does appear to be a prospect of improved attenuation from helmets for vehicle use; and there is some prospect of increased attention being paid to noise problems at the equipment design stage.

Hearing levels in British Army personnel vary very considerably, depending on noise exposure and use or non-use of hearing protection as well as on individual susceptibility to noise. With increasing emphasis on hearing conservation measures and the wider availability of improved hearing protectors, some progress is being made, but some individual soldiers show a reluctance to protect themselves from hearing loss.
To investigate the conditions for the appearance of the continuity effect at the pulsation threshold, loudness-matching experiments were performed.

First the loudness of a tonal masker is reduced by a broad-band noise. The loudness of the masker tone is measured as a function of the level of the noise [1]. Then the loudness of the noise in the critical band of the masker tone is matched by narrow-band noise. It turns out, that the pulsation threshold \( f_{\text{masker}} = f_{\text{test tone}} \) of the tone-in-noise masker can be described by summing the loudness of the masker tone and the masker noise in the critical band of the masker tone. That means that the pulsation threshold curve seems to be an iso-loudness curve of masker and test tone related to the critical band of the test tone.

From this point of view, the influence of a further effect of non-simultaneous interaction has to be taken into account: Loudness-matching experiments of the test tone show, that the test tone is suppressed by the non-simultaneous masker. Especially broad-band noise (Fig. 1) and masker tones of higher frequencies than the test tone reduce the loudness of the test tone at pulsation threshold up to one half.

Fig. 1:
Loudness reduction of the test tone by a broad-band noise as a non-simultaneous masker.

OBSERVATIONS ON BRITISH AND INTERNATIONAL STANDARDS FOR RISK OF HEARING HANDICAP DUE TO OCCUPATIONAL NOISE

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BACKGROUND TO THE COMPARISON

ISO Standard 1999 gives the percentages of noise-exposed populations who, at various ages, will be expected to show hearing threshold levels (av. 1/2-1-2 kHz) worse than 25 dB ISO. The data derive from a survey of some 7000 workers in a US automobile plant by Baughn (1). British Standard 5330 published in 1976 gives analogous figures based on work by Burns and Robinson (BR) (2); its similarities and differences compared to ISO 1999 are outlined here.

The Standards agree in using A-weighting to characterize the damaging effect of noise and on the energy principle (3 dB trade off for double duration) such that equivalent continuous sound level over a working day of 8 h is given by:

\[ L_{eq} = 10 \log \left( \frac{1}{8} \int_0^T L_A/10 \, dt \right) = 10 \log \left( \frac{1}{8} \int_0^T (p_A/p_o)^2 \, dt \right), \]

where \( T \) is measured in hours, \( L_A \) is the "running value" of A-weighted SPL, \( p_A \) is the instantaneous A-weighted sound pressure, and \( p_o = 20 \mu Pa \). Based on recent research data (3), BS 5330 handles impulse noise strictly according to the equation in \( p_A \) above, whereas ISO 1999 offers only a rough guide.

The Standards differ on the audiometric criterion defining onset of hearing handicap, 25 dB HL (av. 1/2-1-2 kHz) in ISO 1999 and 30 dB (av. 1-2-3 kHz) in BS 5330. The change reflects later otological opinion and implies a numerical change greater than 5 dB (see below), so that BS 5330 is actually more strict.

The biggest difference is that Baughn used unselected subjects whereas BR studied only otological normals, this being a more reproducible baseline than a population with unknown content of auditory pathology. This difference originally appeared to be equivalent to a 12 dB audiometric discrepancy, but Baughn (4) later stated that instrumental artefacts probably made his thresholds at least 5 dB too high. This disclosure, whilst explaining half the discrepancy, casts serious doubt on the "risk tables" in ISO 1999.

NEW EXPERIMENTAL DATA

Recent data on 723 long-service steel-workers have thrown new light on the factors described. In particular, the whole group (unselected, like Baughn's) was compared with a subset of 291 obtained by eliminating auditory pathology using criteria similar to BR. Mean values differed by 6 dB (av. 1/2-1-2 kHz). This, together with the adjustments conceded by Baughn, fully explains the difference between ISO 1999 and BS 5330, after allowing for the frequency change. According to theoretical prediction (5) the respective "low fence" levels for equal risk should be 25 and 34 dB. This difference of 9 was exactly confirmed by the steelworkers' study. The data agreed closely in all material respects (mean HL, dispersion, frequency dependence) with the prediction for a noise exposure of 87 dB(A) for 8-hour working days over a period of some 45 years.

REFERENCES

HEARING RESPONSE TO COMBINED WHOLE BODY VIBRATION AND NOISE EXPOSURE IN MAN

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INTRODUCTION
Noise and whole body vibration have direct physiological, psychological and social consequences. They can occur at the same time on worksites in industry. The total biological effect usually attributed to noise only may be modified or enlarged by coexisting vibration.

This problem has important practical consequences connected with the evaluation of hearing impairments of workers exposed to these both factors.

THEORETICAL BASES
Apart from the way by which the acoustic wave reaches the receptor we may assume, basing on the same excitation mode, the magnitude of the stimulus accepted by the hair cells of the Corti organ being the sum of the airborne and material component.

In combined exposure, both to noise and vibration, the effect attributed to noise only can be modified by coexisting vibration.

EXPERIMENTAL TECHNIQUES
The following experiments were performed to elucidate the above-mentioned problem: /1/ estimation of the vibration transmission from the vibrostand along the body to the head; /2/ estimation of the sound pressure level in the outer ear canal using a probe microphone, /3/ estimation of the TTS after selective and combined exposures /whole body vibration and noise/. /4/ The experimental results have been confirmed by hearing analysis of workers exposed to both factors under defined conditions on their worksites.

30 human subjects have taken part in the experiments.
Exposure time - 1 hour;
Vibration: - frequency: 100 Hz and 50 Hz;
- acceleration: 0,5 g and 1,0 g;
Broadbent noise: 77-88 dB SPL.

Few subjects were exposed to increasing vibration intensity from 0,5 g to 3,0 g.

RESULTS AND CONCLUSIONS
The vibration transmitted along the human body reached the head with an intensity sufficient to stimulate the auditory receptor. Increments of the sound pressure levels in the outer ear canal during combined exposure and increased sound pressure levels proportional to changes of the vibration intensity have been found. This confirmed the emission of vibration energy into the ear canal as well as the increase of the airway stimulation. The highest TTS values and the longest recovery time after combined exposures as well as greater hearing loss among workers exposed to both factors than under selective exposure have been found confirming the hypothesis of the cumulative action of these two agents. From the presented results the necessity of corrections for the permissible vibration and noise levels has been concluded.
EFFETS DES BRUITS IMPULSIFS SUR LE SYSTEME NERVEUX CENTRAL

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INTRODUCTION

Le bruit représente l'exemple type du stress neurotrope. L'action de ce type d'agression, au niveau du système nerveux central, a été mise en évidence par l'étude des taux de renouvellement de trois neuro-transmetteurs cérébraux (Nor-Adrénaline, Dopamine, Sérotonine) chez diverses souches de rats.

TECHNIQUES EXPERIMENTALES

Des séries de rats mâles de diverses souches (Wistar, Long Evans et Auguste) sont exposés à des bruits impulsifs, du type onde en N, caractéristiques du bang sonique. Après exposition pendant deux heures à des stimuli de divers paramètres physiques, les rats sont sacrifiés et les médiateurs cérébraux sont dosés.

Les calculs des taux de renouvellement des neuro-transmetteurs sont effectués selon les principes de IVERSEN (1), TOZER (2) et NEFF (3).

RESULTATS ET CONCLUSION

A surpression de créte constante 3,5 mbar, quels que soient les autres paramètres physiques du bruit impulsif, on observe toujours une augmentation des taux de renouvellement de Nor-Adrénaline tandis que ceux de la Dopamine présentent des variations moins significatives. L'étude de la Sérotonine ne permet de dégager aucun effet du bruit.

La durée d'exposition au bruit n'influence pas sur les pourcentages de variation des taux de renouvellement des médiateurs cérébraux.

Par contre les pourcentages de variation des taux de renouvellement de Nor-Adrénaline semblent fonction de la fréquence de présentation des stimuli.

L'étude des modifications des nucléotides (AMP cyclique et GMP cyclique) ainsi que du GABA cérébral a été effectuée. Seule une modification du GMP cyclique a pu être mise en évidence.

L'action d'un anti-épileptique (n-DPA) qui agit par une augmentation de GABA au niveau du système nerveux central (GODIN et coll.)(4), semble indiquer que le contrôle de la réponse au bruit ne passe pas par l'intermédiaire de ce médiateur.

La sensibilité de la réaction au bruit semble dépendre également de facteurs génétiques.

REFERENCES

INTRODUCTION

Loudness of brief sound-stimuli grows with increasing duration. This phenomenon (temporal integration) has been studied for many years in terms of "the time-constant of the ear" but great disagreement exists about the numerical value of the time-constant.

It is shown here that a single time-constant does not yield an adequate description of temporal integration, (1).

THEORETICAL BASIS

A time-constant description of the loudness growth is given by

\[ S(t) = k I (1 - \exp(-t/\tau)) \]

where \( S \) = loudness, \( I \) = sound intensity, \( t \) = time, and \( \tau \) = time-constant. According to this model, loudness can be maintained - for very short sound stimuli (\( t \ll \tau \)) - if a doubling of the duration is compensated by a 3 dB decrease in intensity.

EXPERIMENTAL TECHNIQUES

Loudness balances between tone pulses of duration \( T \) and 2\( T \) have been performed, (\( T = 5,10,20,40,80,160,320 \) ms). Pulses were filtered (1/3-octave). Frequencies: 500, 1000 and 4000 Hz. Levels: threshold, 35 and 55 dB SPL.

RESULTS AND CONCLUSION

Fig. 1 shows typical experimental results (crosses) with 95% confidence intervals. Results are in good agreement with results from ISO-Round Robin on loudness level of impulsive sounds (filled circles). The theoretical curve for a 100 ms time-constant model is shown for comparison. It is seen that the level difference for short durations exceeds 3 dB. A better model consists of a combination of 2 time-constants (typical 10 ms and 200 ms) as illustrated in Fig. 2.

The deviation from the energy time-constant model for small pulse durations is also found in much of the literature, (1). As different investigators take different parts of the curve into account when estimating the time-constant, different values of the time-constant will result.

REFERENCES


The Acoustics Laboratory, Technical University of Denmark.
MEASUREMENT AND ASSESSMENT OF FLUCTUATING NOISE

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INTRODUCTION

Several methods of measurement are today used to assess the annoyance from fluctuating noise, e.g. road traffic noise. One of the most widely used methods of measurement is determination of the equivalent, continuous sound level, $L_{eq}$. $L_{eq}$ measures the energy of the acoustic signal but no one knows explicitly whether the annoyance is a direct function of the energy. However, several experiments show that $L_{eq}$ is a good estimate for the annoyance (1, 2). Today we also know that in some cases $L_{eq}$ works badly (3) and some adjustments to $L_{eq}$ have, therefore, been suggested, e.g. Noise Pollution Level, $L_{np}$ (4). Lately, other methods such as the derivative methods (methods where the adjustment to $L_{eq}$ is a function of the time derivative of the SPL) have been suggested (5).

Because legislation for noise in our environment is more and more required it is very important to have objective methods of measurement which are good for estimating the annoyance. Therefore, we have started up a laboratory experiment which will give us more information about the correlation between subjective assessment of annoyance and objective measurements. The practical part of the experiment will be carried out during the first part of 1977.

METHOD

Test subjects are exposed to different types of fluctuating noise in a specially established room which looks like an ordinary living room. After the exposure test subjects are asked how annoying the noise was. The noises used are different road traffic noises and artificial noises with various fluctuations. The subjects will give their assessment on a questionnaire. The noise is also measured with different objective methods. These measurements are carried out by means of a mini computer and different units such as $L_{eq}$, $L_{np}$, $L_{10}$, derivative methods etc. are computed.

RESULTS

After the subjective and objective measurements the data will be statistically analyzed and the correlation between the different objective data and the subjective data will be determined.

REFERENCES

(2) P. Voigt et al., Sozial- und Präventivmedizin (1974), 19, 197.
INTRODUCTION

Noise measurements can be divided into two classes: objective measurements in which only instruments are used, and subjective measurements in which test subjects are involved. It is, generally, desirable to replace subjective measurements by objective measurements provided a comparable outcome is achieved.

This is particularly difficult for quasi-steady and impulsive noises where many suggestions for measurement of loudness or loudness level have been made.

EXPERIMENTS

Within the framework of ISO a scientific cooperation between 22 laboratories in 12 countries has been established. All laboratories made objective and subjective measurements of loudness level of a set of 14 quasi-steady and impulsive noises.

The results were submitted to the organizers for further analysis and comparison. Due to the structure of the about 50,000 data, great difficulties were encountered in the statistical analysis and unconventional methods had to be applied.

RESULTS

The general trend is that the more complicated objective methods, according to Zwicker and Stevens, yield results in close agreement with subjective measurements. Sound level measurements on "fast" and "slow" give results about 10 dB lower but can nevertheless be used for rank ordering of noises if the A-weighting is used. For this purpose the "impulse" setting and the D-network are particularly poor.

REFERENCES


INVESTIGATION OF BINAURAL SIGNAL PROCESSING THEORIES REGARDING BINAURAL INTELLIGIBILITY LEVEL DIFFERENCE IN FREEFIELD CONDITIONS

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INTRODUCTION

Some theories of binaural signal processing regard as a criterion for the perceptibility of an information signal within noise the signal to noise power ratio of that signal which results from the addition of the two earsignals after special processing of each signal. Supposing a definite signal processing mechanism one can calculate the Binaural Masking Level Differences (BMLD) for sinewaves and playback with headphones quite simple and can compare these values with the BMLD's measured in listening tests. Contrary to this proceeding the calculation of Binaural Intelligibility Level Differences (BILD) in freefield conditions is very difficult among other reasons because of the complicated structure of speech signals.

EXPERIMENTAL BASES

Therefore we tried to test some theories of binaural signal processing while simulating the supposed processing mechanism of the sound signals we measured in the ears of a median test person in freefield conditions. Instead of the dichotic playback of the earsignals we find in a free sound field we used in our listening experiments a monotic playback of the one sound signal we get when summing up the externally in a special way processed earsignals. Neglecting an internal hearing noise one could suppose to get Monaural Intelligibility Level Differences which are in case of a correct external signal processing identical with the BILD's in freefield conditions.

EXPERIMENTAL TECHNIQUES

For performing an external processing of the earsignals we use a special device we call 'Direction Mixer'. Feeding its input with any signal \( s(t) \) one will get at its outputs the same signals \( g_r(t) \) and \( g_l(t) \) one would measure in both earcanals of a median test person in an anechoic room with a loudspeaker in a definite direction radiating the signal \( s(t) \). In a second operating mode one will get the signal \( s(t) \) when feeding the 'Direction Mixer' with the earsignals \( g_r(t) \) and \( g_l(t) \).

RESULTS AND CONCLUSIONS

In this way an external processing of earsignals measured in a free sound field with one speech and one noise source only can be performed so that the information or noise parts of the left and right earsignal will be equalized in phase or in magnitude and phase. Addition or subtraction can give a complete elimination of the noise signal or a doubled magnitude of the information signal. Results of listening experiments performed in the way described before will be given in details in our paper.
THE RELATIONSHIP BETWEEN THE TRANSMISSION PROPERTIES OF THE OUTER EAR AND THE BINAURAL INTELLIGIBILITY LEVEL DIFFERENCE

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INTRODUCTION

It is known that the Binaural Intelligibility Level Difference (BILD) is depending on the acoustic transmission characteristics of the human outer ear. In case of normal operation of the middle and inner ear and normal signal processing in the brains a changing of the normal earsignals - for instance when closing up one ear or using hearing aids - will cause smaller BILD's mostly.

EXPERIMENTAL TECHNIQUES

The relationship between the BILD in freefield conditions and the properties of the sound signals in both earcanals one can investigate in the best way if one use in listening tests head related stereophonic transmission technique with its great advantage of playback the test signals with headphones. To avoid complicated arrangements of sound sources around a dummy head we developed in the Institute of Electrical Communication of the Technische Hochschule Aachen a special electronic device we call 'Direction Mixer'. This device will simulate the acoustic transmission characteristics of the outer ears of a median test person for different directions of a sound source to the head in an anechoic room. For 12 angles of incidence electronic filter circuits approximate first the absolute value of the freefield outer ear transmission functions and second the group delays of the interaural outer ear transmission functions.

When fed with any signal s(t) at its input the device will supply the same signals at its outputs one would measure in both earcanals of a median test person who will be exposed to a sound signal s(t) coming from a selected direction in the anechoic room.

RESULTS AND CONCLUSIONS

Experiments with one source of information signal and one source of broadband noise only showed that the BILD's in both cases - first using our 'Direction Mixer' and headphones and second using loudspeakers in an anechoic room - gave identical results.

With the 'Direction Mixer' one can test in a simple way the influence of absolute value and interaural group delay of the outer ear transmission functions separated one from the other over the BILD in freefield conditions.

On the other hand one can test in the same way the relations between BILD's in freefield conditions and fundamental transmission characteristics of some kinds of hearing aids (monaural, binaural, CROS-Technique etc.).

The results of such experiments will be reportet in details in our paper.
ANALYSE EINIGER EINFLÜSSE AUF DIE ERKENNUNGSZEIT VON SPRACHLAUTEN

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EINFÜHRUNG

Geprüft wurde der Einfluss des weissen Rauschens auf die Verhältnisse zwischen Sprachlautenverständlichkeit und Zeitdauer der Sprachlaute.

MESS UND PRÜFTECHNIK


SCHLUSSFOLGERUNG

Die Ergebnisse sind graphisch mit unterbrochener Linie dargestellt und sie zeigen dass sich die Verständlichkeit der Vokale im Geräusch mit Verlängerung der Zeitdauer, ohne Steigerung des Schalldruckes verbessert.

LITERATUR:
(1) K. Schubert, Sprachhörprüfmethoden
Cette recherche traite du problème de la perception de l'intensité des voyelles produites à différents niveaux d'effort physiologique.

On sait depuis les travaux de Lehiste et Peterson (1959) puis ceux de Rossi (1971) que les voyelles produites à une même intensité objective ne sont pas perçues comme isophoniques. Une table de facteurs de pondération a été établie pour le niveau de la conversation normale. Nos travaux sur la pression sous-glottique et l'intensité Marchal (1975) laissent penser que ces valeurs ne sont pas constantes et doivent être établies pour différents niveaux de production.

Nous avons soumis les productions de voyelles françaises de 10 locuteurs répétées plusieurs fois à des intervalles de temps régulier à un groupe de 50 auditeurs: Ajustement par la méthode croissante et décroissante.

Les résultats des tests de perception confirment l'hypothèse de facteurs variables de correction et permettent d'apporter une contribution originale à la critique de la théorie motrice de la perception. Les auditeurs sont sensibles à l'effort fourni lors de la production: une preuve supplémentaire nous est apportée par la comparaison de la même voyelle à différents niveaux. Il ressort également de cette étude que la pression sous-glottique ne peut être retenue comme un corrélat systématique de l'effort physiologique.


INTRODUCTION
A study was performed to generate information about the use of hearing aids with the telephone. The modes of coupling, frequency of use and success are investigated.

METHODOLOGY
The survey sample was drawn in approximately equal proportion from three major cities - Toronto, Montreal and Ottawa - to result in a total of 301 respondents. Source contacts for the sample were hearing impaired associations and hearing aid dealers. The sample was specifically designed to provide substantially equal representation from all age groups - the young, the middle aged and the older. The sample in Montreal was divided equally between French and English speaking people. Because of this type of emphasis, the sample was not generally representative of the total hearing impaired population, but it did allow analysis of telephone use over a variety of conditions. The survey was conducted on a face-to-face personal interview basis.

RESULTS AND CONCLUSIONS
The respondents in the survey indicated the quality of their hearing impairment on a four point scale: good hearing, a little trouble, a lot of trouble and no hearing. Results show that the sample contains a majority of people who consider their impairments to be severe. The hearing aid was judged to provide improved hearing ability. The rating improved from an average of "a lot of trouble" without an aid to a rating between "no trouble" and "a little trouble" with an aid. A rank order in terms of hearing difficulty for different situations was established in order of easiest to most difficult: 1. face-to-face, 2. home, 3. T.V., 4. automobile, 5. work, 6. telephone, 7. school, 8. small groups, 9. public places and 10. large groups.

The hard of hearing person normally hears signals by using the microphone on the hearing aid. Sometimes the hearing aid is equipped with a magnetic pickup telecoil which allows the user to pick up information radiated in the form of magnetic energy. Both the microphone and the telecoil are used in conjunction with the telephone. Some respondents do not use the aid at all but listen to the telephone with an unaided ear. The hard of hearing have therefore three different methods of using the telephone: 1. Telecoil, 2. Microphone and 3. Unaided ear.

Of the 301 respondents, 42% use the telecoil, 23% use the microphone, 26% do not use their hearing aids with the telephone and 9% do not use the telephone at all. A subsequent survey (1) provided national figures for these categories which are thought to be more representative of the hearing impaired population. These are 14% ± 9.9, 21% ± 11.7, 51% ± 14.3, and 14% ± 9.9 in these categories. Telecoil and microphone users do not report significantly different hearing impairments. The group that uses the telephone unaided reports a significantly less severe impairment than the other groups. The group that does not use the telephone at all reports a more severe impairment.

The use of telephones in different locations by the telecoil, microphones and unaided groups is not significantly different nor are these groups significantly different in the number of sets they use in the average day.

Telecoil and microphone users report degrees of satisfaction with their performances on the telephone that show no statistical difference.

REFERENCE
During face-to-face speech communication, both the hard of hearing and the profoundly deaf rely to a great extent on the visual information they receive via speech reading to understand what is being said. Speech reading is the method of reading speech by watching the lips and face of the speaker.

Firstly, the audiologist can be guided by means of a simple test in recommending a hearing aid to a patient as well as checking its effectiveness after a period of use. Secondly, the information can be utilised in guiding the research worker who is developing speech aids for the profoundly deaf.

A pilot study was conducted to investigate the possibility of determining, quantitatively, the contribution offered,
(a) by speech reading and,
(b) by hearing aids,
to the total amount of speech understood by hard-of-hearing people
Seventy-five pupils of a school for the hard-of-hearing were used for the study. The age and type of hearing loss, derived from the slope of an audiogram, was known of each pupil.

Four lists of words, spoken by a male person, were recorded on a video-tape such that, when played back via a television monitor, a life size, frontal image of the head and shoulders of the speaker was visible. The sound was reproduced by a loudspeaker at normal volume.

Each pupil's ability to understand speech by,
1) speech reading while using his hearing aid,
2) using his hearing aid alone,
3) speech reading without using his hearing aid, and,
4) neither speech reading nor using his hearing aid,
was recorded in the form of a score of words correctly repeated by him.

Some of the results arising out of the analysis are:
1) While the majority of pupils received great benefit in using a hearing aid, for a significant number the use of a hearing aid appeared to degrade face-to-face communication.
2) More speech information is received via speech reading alone than by use of a hearing aid alone.
3) The profoundly deaf are capable of normal face-to-face communication, provided they receive a minimal amount of aural information.

The last of these results is of special significance in research being conducted in developing tactile aids for the deaf.
MODERNE MESSVERFAHREN FÜR HÖRGERÄTE

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STANDARDVERFAHREN UND "WIRKSAME AKUSTISCHE VERSTÄRKUNG"


Die "wirksame akustische Verstärkung" des Hörgerätes ("etymotic gain" (nach Burkhard), "insertion gain" (2)) gibt an, um welchen Faktor der Schalldruck am Trommelfell gegenüber dem unversorgten Fall erhöht wird. Sie entspricht dem subjektiven Hörvergleich zu bestimmten "akustischen Freifeldverstärkung".

ERGEBNISSE

Beim Hinter-Ohr-Gerät übersteigt die "wirksame akustische Verstärkung" unterhalb von 500 Hz die nach Standardbedingungen ermittelte "akustische Verstärkung" um etwa 3 dB, während sie zwischen 2 kHz und 4 kHz unter der "akustischen Verstärkung" liegt. Oberhalb von 4 kHz ist die "wirksame akustische Verstärkung" in der Regel größer (3).

Wird die "wirksame akustische Verstärkung" für verschiedene Schall-einfallsrichtungen bestimmt, so läßt sich auch die Richtwirkung der Geräte beschreiben und u.a. der Einfluß unterschiedlicher Schalleintrittsöffnungen am Gehäuse untersuchen. Fig. 1 zeigt die Erhöhung $\Delta L$ der Mikrofonempfindlichkeit eines Hinter-Ohr-Gerätes im Vergleich zum freien Schallfeld (0 dB) für zwei verschiedene Stellen des Schalleintritts bei gleichartigem Schallkanal für Schalleinfall aus verschiedenen Richtungen (0° - von vorne). Die Vorteile der frontalen Schallaufnahme sind zu erkennen.

LITERATUR

(3) Helle, R., Messung der wirksamen akustischen Verstärkung von Hörgeräten, "Fortschritte der Akustik" DAGA 76, VDI-Verlag, Düsseldorf
THE INFLUENCE OF DIFFRACTION ON THE PERFORMANCE OF HEARING AIDS

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There is a great difference between the performance of a hearing aid when measured under standardized conditions (1-2) and the performance when the hearing aid is worn on a person (3).

The difference is due to several causes:
1) Diffraction of the sound field by the body of the wearer.
2) Variations in earmould dimensions.
3) Effects of leakage between the earmould and the meatal wall.
4) The acoustical properties of the actual ear are different from those of the coupler.

In order to further analyze these differences a research project has been undertaken, regarding the influence of the body (or the head) of the wearer on the frequency response of hearing aids.

THE PROJECT

The project comprises 3 hearing aids: 1 pocket aid, 1 ear level aid with frontal microphone and 1 ear level aid with bottom microphone.

It is also intended to measure the influence of the diffraction on the response of the untreated ear.

The test crew comprises 10 male and 10 female subjects.

MEASURING TECHNIQUE

Contrary to the standard method of measuring hearing aid performance we have chosen to use 1/3 octave bands of pink noise instead of pure tones.

This has been done for several reasons:
1) Diffraction effects very often produce sharp interference dips in the response curve. These dips are not likely to be perceived by the user, and they make the interpretation of the results difficult. By using bands of noise, approximately corresponding to critical bands, a better prediction of the subjectively noted effect is estimated.
2) The use of noise bands facilitates the mathematical treatment of the results.
3) The use of discrete noise bands is more suitable for loudness balance measurements than pure tones.

As the amount of measurements are considerable it has been attempted to automatise the measurements, using a computer (PDP8) to control the measurements and to store the measuring results for a later statistical treatment.

MEASURING RESULTS

At present no results are available, but they will be presented at the congress.

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ESTIMATION OF REAL-EAR GAIN OF A HEARING AID

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INTRODUCTION

A previous study has demonstrated how an acoustic reflex threshold procedure can be employed to measure the real-ear gain of a hearing aid in normal ears (1). Measures of real-ear gain reported here have been obtained from the aided and unaided reflex growth functions of ears exhibiting sensori-neural hearing loss. The growth curves for the two ear conditions should be parallel but separated in sound pressure level by the real-ear gain of the aid.

EXPERIMENT

Eighteen subjects (29 ears) were tested, their ages ranging from 37 to 74 years (mean age, 57.3 years). All had a degree of hearing loss between 20 and 60 dB, exhibited normal tympanograms and produced reliable reflex responses during preliminary trials. A single aid (Oticon, BE 11) and type of earmould (Universal Hearing Aids, Uniduct) were employed throughout the study. Reflex responses were recorded for pure tone stimuli in the range 250 Hz to 4 kHz. A minimum of three points on the growth function were obtained for both aided and unaided ear conditions, the sound pressure level at the exposed ear being measured by probe microphone. Coupler measurements of the gain of the aid were also carried out at the same frequencies.

RESULTS

With the system employed, the growth of reflex response with sound pressure level was found to be linear. Least squares regression lines were fitted to the individual data. Unfortunately, only 44% of the 227 paired functions were parallel with negligible deviation. However, considering all paired functions for which there exists at maximum a difference in gain of 5 dB between that estimated at reflex threshold (zero response on the function) and that estimated at the lower of the two maximum reflex responses obtained for the individual, 82% of the data qualifies for analysis. The group mean gains (calculated from the 5 average of the two estimates) are shown in fig (1) together with the coupler measurements of gain. The two sets of data are clearly different. The differences are highly significant (p < 0.001) except at 750 Hz, 1.5 and 4 kHz were the appropriate p-values are 0.01, 0.02 and 0.25 respectively.

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Three separate sets of statistical distributions determine the relative effectiveness of personal hearing-protective devices such as earplugs, ear caps, and earmuffs. These distributions describe 1) the variability of noise spectra in which the devices are to be used, 2) the variability of attenuation characteristics each device is capable of producing for a variety of potential wearers in a given noise, and 3) the variability of average attenuation characteristics for a variety of hearing protectors being used by a given population of wearers in a given noise.

Consider the measurement of noise attenuation, not according to a standard method of testing attenuation at each frequency, but simply by determining the overall attenuation given by a particular hearing protector to a particular listener in a particular noise. Clearly the result will change when the noise spectrum is changed. Therefore, in order to develop a valuable rating system, one must take into account the fact that, normally, the specific noise in which the protector will be used is unknown (indeed, for many kinds of working environments where one might possibly expect to know the exact noise spectrum, a given employee is exposed to several different spectra in the course of a usual working day). Thus it has been necessary to develop hypothetical noises that can be used in attenuation calculations to estimate the effects of noises with unknown spectra. Of the several such hypothetical noises so far offered, one (Jerry Tobias and Daniel Johnson, unpublished) gives accurate estimates for 99% of industrial and military noises. If hearing protectors have their effectiveness computed as if they were to be used in the spectrum of this "Typical Noise," the resulting attenuation figures will be very close to correct in most cases.

However, the results of that computation will describe the effectiveness of a hearing protector only for the average member of the test population, since the values come from standard, frequency-by-frequency tests of ten or more listeners, whose data are averaged. Thus the resulting attenuation value indicates the worst performance the device can offer to the best 50% of the user population. One really ought to be concerned about the performance of the device for larger proportions of users. Luckily, standard measurements permit the calculations of standard-deviation scores for the group of listeners who serve as subjects. If, instead of computing the effective attenuation for the mean subject, one computes the score of the subject at the mean-minus-one (or the mean-minus-two) standard deviations, the result indicates the effectiveness of the device under test for 84% (or 98%) of people in 99% of noises.

Finally, because many prospective buyers of hearing protectors are not trained to understand the decibel scale, and because many of those who do understand decibel notation tend to treat fractional decibel differences as if they were meaningful, a truly valuable rating system for personal hearing protectors needs to be simplified—preferably simplified enough to classify effectiveness without naming decibels of attenuation. Therefore, I propose a Protector-Attenuation Rating (P-AR) number derived from the distribution of previously calculated attenuations for a large series of hearing protectors. Those protectors that are at least 2 standard deviations better than the mean protector (for a preselected proportion of people) form Class 1; those that are at least 1 standard deviation better than the mean protector (but are less than 2 standard deviations better) form Class 2; those that are at least at the mean (but are less than 1 standard deviation better) form Class 3; and so on through Classes 4, 5, and 6. All the prospective user or buyer needs to know is that Class 1 is better than Class 6. He may also want to know that, depending on the measured noise level in dBA in which they'll be used, hearing protectors should be chosen from the top 1 or 2 (or, if the noise is quiet enough, 3 or 4) Classes if they are to be expected to protect the majority of employees from permanent, noise-induced hearing loss.
APPLICATIONS AND RELIABILITY OF THE HEARING PROTECTOR RATING

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INTRODUCTION

Many scientific papers are concerned with the classification of hearing protector devices, its attenuation properties and the comparison between their attenuation curves. This comparison is not very simple and to know if a certain hearing protector is adequate to solve a given problem are necessary cumbersome calculations to know if the new level reaching the occluded ears is below the maximum acceptable limits.

BEHAR, TAIBO and RAITZIN (1), describe a new method to classify internal and external hearing protectors, based in the use of shifting standard profiles, obtaining thus a rating number for the device.

The present paper consists of two parts: the first one is concerned with the description of a pool of large samples of noise spectra measured in several laboratories, including our Acoustical Division.

In the second part the rating numbers of several external and internal hearing protectors are obtained and related with the noise spectra.

DESCRIPTION OF THE METHOD

The first part of the study is concerned with the description of samples of noise spectra. These data are later treated by non-parametric statistical methods in order to find out tendencies in the spectra configurations. It must be pointed out that these spectra configurations are closely related to those found out by WAUGH (2).

In the second part rating numbers of attenuation curves of external and internal protectors were obtained. These rating numbers are related to the spectra configurations described in the first part using both the conventional and the rated procedure to calculate the resultant attenuation values. Regression and correlation techniques including a specially developed procedure of fitting a polygon by least squares are used to reach some practical conclusions about the reliability of the new technique and its advantages when compared with the classical one. Among these advantages it must be emphasized the practical and straightforward way of choosing the right protector concerning its attenuation characteristics.

It must be remarked that the previous considerations can be directly extended to a broader class of spectra configurations usually found in industry, using developments running on the same lines.

REFERENCES

(3) J. Tobias, PH,D, "Earplug rankings based on the protector attenuation rating (P-AR)", FAA Civil Aeromedical Institute, Oklahoma City, october 1975.
Einführung

Experimentelle Untersuchungen

Ergebnisse und Gehörschützerauswahl
Die Kenntnis der Frequenzbereiche, in denen die Geräuschveränderungen beim Lockern des Gesenkwerkzeuges auftraten, erlaubte eine angepaßte Gehörschützerauswahl, die die unvermeidbare Anpassung des Hörens an ein verändertes Geräuschspektrum auf ein Minimum beschränkte. Durch das ständige Tragen des nach diesen Kriterien ausgewählten Gehörschutzes wurden sonst zwangsläufig während der Arbeitsschicht eintretende zeitliche Hörschwellenverschiebungen (TTS) vermieden und damit die volle Leistungsfähigkeit des Gehörs für die selektive Geräuscherkennung aufrecht erhalten.

Die deutsche VDI-Richtlinie 2560 (Entwurf) "Persönlicher Schallschutz" erleichtert durch die Kennzeichnung der Schalldämmung von Gehörschützern in drei Gruppen (Z-Werte für normierte hoch-, mittel- und tiefrequente Arbeitsgeräusche) die Berücksichtigung der Sonderanforderungen der akustischen Informationsvermittlung. Eine Liste aller Z-Werte der in Deutschland benutzten Gehörschützer enthält das vom Institut herausgegebene Lärmschutz-Informationsblatt LSI 01-830 "Persönlicher Schallschutz".
INTRODUCTION

A reason often given by workers for not using hearing protection is that they feel unsafe or restricted whilst wearing them, because they are less able to hear sounds which convey important information. As part of a research programme aimed at assessing the basis of this attitude, an experiment was conducted to determine the masked thresholds of two typical warning signals with and without hearing protection.

THE EXPERIMENT

The two warning signals, a siren and a bell, were presented to subjects in random noise at one of two sound pressure levels, and masked thresholds determined using a manual audiometry technique. The following four ear conditions were considered: unoccluded, plugs, muffs, and a filter simulating the mean attenuation of the muffs.

RESULTS

It was found that the hearing protection did not cause any large change in the masked thresholds of the warning signals. There was no significant difference between the protected ear conditions, so that the results can be summarised as in Table 1. At the higher noise level there was a small but significant advantage i.e. lower masked threshold, when wearing protectors, but there was no significant difference at the lower level. This protection effect was predominantly due to the attenuation provided, but did not appear to be sensitive to its exact value or spectral characteristics. It is postulated that prevention of a widening of the auditory filter at high noise levels explains the advantage provided by hearing protectors.

CONCLUSIONS

In terms of masked thresholds it appears that there is no basis for the attitude that the wearing of hearing protectors creates a hearing handicap for people with normal hearing. The next stage is to simulate more realistically the industrial situation, where a person may not be deliberately listening for a warning sound, and to look for any effects on the alerting characteristics of the sounds.
EFFECTS OF SHIPS FIREROOM NOISE ON PERFORMANCE

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INTRODUCTION

The noise exposure for men standing the normal one-in-three watch schedule in propulsion machinery spaces on large naval ships is 94±3 dB A for 4-hours-on and 8-hours-off for as many days as the ship is underway. The purpose of this investigation was to measure Temporary Threshold Shifts (TTS) and noise dosemeter results (to be reported in Seville) and performance changes accruing from 12 days continuous steaming.

THEORETICAL BASES

Almost no data exist for specifying the effects of long time high level noise exposures on serial 4-choice reaction time nor on perceptual interference as measured by a modified form of the Stroop color word test. Much of what work is reported is contradictory.

EXPERIMENTAL TECHNIQUES

Three groups of men were selected from the Engineering Department as follows: 11 who stood watches in "quiet" places (69±5 dB A); 28 who stood watches in continuous noises of 94±3 dB A, 11 of whom wore some sort of hearing protection (18±7 dB) and 17 who did not. All of these men were given audiometer tests at the beginning and end of certain of their 4-hour watches and some of these wore noise dosemeters during their noise exposed watch periods. More than half were given 4-choice serial reaction time tests and a version of the Stroop color word test toward the beginning and end of one 4-hour watch period.

RESULTS AND CONCLUSIONS

Dosimeter results showed that the typical noise exposure in terms of equivalent steady state noise level in the space, where most measurements were made, averaged 89.9 dB A with a standard deviation of 4.5, whereas physical measures averaged 91.6 dB A with a standard deviation of 7.9. The unprotected noise-exposed group did show small but consistent TTS at 4, 6, and 8 kHz and 7 of the 8 who showed consistent losses were from the unprotected group. On the 4-choice reaction time test, all groups deteriorated on minutes 4 and 5 of a 5-minute test, although the quiet group deteriorated less. End-of-watch tests showed less deterioration (apparently some learning took place) especially for the unprotected noise group.

All groups improved their performance on the Stroop retest (at the end of the watch). The unprotected group had the least, and the quiet group the most, perceptual interference within the first half-hour of their watches. The control groups (quiet and unprotected) showed reduced interference at the end of the watch; the unprotected noise-exposed group did not.

However, the major observation of this shipboard (field) study may be the insurmountable difficulty of running a controlled experiment on a not-to-interfere basis on subjects who are not highly motivated. It appears that if extra duty for which the subject sees no gain to himself is added, it must be exacted in a hard-nosed, chain-of-command, continually supervised effort. This reaction in itself may say something about the fatiguing, irritating aspects of standing four-hour watches in hot, humid, noisy engine rooms.
SPEECH AUDIOMETRY VIA BONE CONDUCTION

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INTRODUCTION

Speech audiometry via bone conduction (bc) has important applications in clinical audiometry (1). In the present study, a new method of calibrating the bc branch of a speech audiometer was developed and tested. It allows direct comparisons between bc and air conduction (ac) results.

EXPERIMENTAL TECHNIQUES

Intelligibility measurements were carried out using a record of German monosyllabic words according to DIN 45626. The words were presented binaurally, at first via headphones (Beyer DT 48 with flat cushions) and then via a bone vibrator (Präcitrionic KH 70, applied to the forehead, static pressure 5N), the test person sitting in an audiometric chamber without any ambient noise. Discrimination of the test words was determined as a function of the sound pressure level with a group of 12 persons of normal hearing.

Starting from the free field sensitivity levels of the headphones (2) and the vibrator (3) determined previously by loudness comparisons with a free sound field, a nearly flat frequency response of both branches of the speech audiometer was achieved by adding suitable equalizing networks. The final response in the frequency range of 125 Hz to 8 kHz was flat within ± 2 dB (for ac) and ± 3.5 dB (for bc). Using a speech simulating noise of the same level as the mean of the test words, both branches were calibrated in terms of sound pressure level of speech referred to a free sound field.

RESULTS AND CONCLUSIONS

The result is given in fig. 1. The mean ac and bc discrimination curves agree fairly well (only 2.5 dB difference at 50 % discrimination). The standard deviations of the individual discrimination results near 50 % were ± 13 % for bc and ± 20 % for ac. The small shift of the bc curve to higher levels may be caused by remaining differences of frequency response and by small uncertainties in the determination of the free field sensitivity levels of the headphones and the vibrator.

The result demonstrates that it is possible to calibrate bc speech audiometers with sufficient accuracy only by using known sensitivity data of the bone vibrator.

REFERENCES

6. INTERNATIONAL CONGRESS ON ACOUSTICS

M. E. Allard

Introduction

The purpose of this paper is to present the results of an experimental investigation of the propagation of sound through a medium containing a large number of small particles. The medium is a suspension of fine particles in a liquid, and the particles are assumed to be permanently fixed in their respective positions.

EXPERIMENTAL RESULTS

The experimental setup consisted of a sound source placed at one end of a long tube, and a microphone at the other end. The sound source was a loudspeaker, and the microphone was a high-sensitivity condenser type. The tube was made of glass and had a length of 10 meters. The particles used were of 1 micron diameter and were suspended in water.

The results obtained show a significant decrease in the sound intensity as the particles are introduced into the medium. The decrease is more pronounced at higher frequencies. The pattern of the sound wave is also altered by the presence of the particles.

Spectral Analysis

A spectral analysis of the sound wave was performed to determine the frequency components present in the sound. The analysis was carried out using a computer program that could handle large data sets efficiently. The results showed that the sound wave contained a significant amount of energy at frequencies below 1 kHz.

Conclusions

The results of this study indicate that the propagation of sound through a medium containing small particles can be significantly affected. The decrease in sound intensity and the alteration of the sound pattern provide new insights into the behavior of sound in such environments. Further research is needed to understand the underlying mechanisms and to develop more accurate models for predicting sound propagation in such media.

Reference

III. Psycho and physiological acoustics

I. SPEECH AND COMMUNICATION
EFFECT OF HYPOXIA ON SHORT-TIME SPECTRA OF SPEECH

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INTRODUCTION

A condition of hypoxia or oxygen deficiency imposes on the human articulatory system certain difficulties and limitations. The present experiments attempt to clarify what are the basic effects of hypoxia on short-time spectra of vowels and consonants in connected speech.

THEORETICAL BASES

The present knowledge of the mechanism and properties of speech production in a condition of hypoxia is poor. The main effects of hypoxia on articulatory movements are expected to be noticeable in transients between stable configurations of vocal tract.

Fig.1. Sound spectrograms of the word /zdravka/: (a) normal; (b) anoxia

EXPERIMENTAL TECHNIQUES

A high altitude environment was simulated in a low-pressure chamber. After 4 minutes exposure to 17,000 feet altitude Subject uttered standard sentences. Sound spectrograms of the recorded anoxia speech was compared with the normal one within the same Subject.

RESULTS AND CONCLUSIONS

Representative anoxia and normal spectrograms of the Serbo-Croatian word /zdravka/ are presented in Fig.1. Apparently, in situation of hypoxia general articulation is soft and formants glides are less pronounced. For example, the formants and pitch inflections in semi-vowel /r/ are badly damaged. In the first and second formant regions there is a lot of anoxia noise added. Voiceless stop consonants and fricatives are particularly noisy.

The spectral analysis clearly demonstrates that the articulatory movements in hypoxia are not properly coordinated.
Speech disorders are either of organic or functional origin. However, it is frequently difficult to differentiate with reliability these two causes. The approach to and outcome of treatment depend upon accurate differentiation. It is known that only physical examination and psychological testing do not always discover the real causes of disorder.

The authors consider that a combined analysis of electromyographic activity of the orbicularis oris muscle and psychological testing under normal and hypoxic conditions can contribute to more accurate differentiation in the establishment of voice and speech disorder origins. Thus, with the aid of this combined method it is possible to make a timely diagnosis of disorders and this ensures faster and more qualitative treatment.
INTRODUCTION

The present state-of-the-art for solving the problem of helium-speech, that is, degradation of speech intelligibility under pressure is unsatisfactory[1]. A perceptual study on the synthetic speech by dynamically controlling the acoustical conditions of breathing gas is carried out in order to prepare the data for designing the helium-speech unscrambler and to clarify the underlying nature of helium-speech perception.

According to the acoustical analysis, the relation between the formant frequency in a helium mixture $F_{He}$ and that in normal air $F_{air}$ is well explained by

$$F_{He}^2 = k^2 F_{air}^2 + (r-1) F_{wo}^2,$$

(1)

where $k$ and $r$ are the sound velocity ratio ($c_{He}/c_{air}$) and the density ratio ($He/air$) of the breathing gas respectively[2]. The perceptual analysis has indicated that vowels and consonants of helium-speech show unusual confusion, and that these singularities are supposed to be due to the non-linear formant elevation[3].

EXPERIMENTAL PROCEDURE

5 synthetic vowels of Japanese for the 15 combinations of $k$ and $r$ shown by dots in the Figure were prepared for the listening test of vowel identification. $F_{He}$s are determined by Eq. (1), where typical values of male adults are adopted for $F_{airs}$ and $F_{wo}$=200 Hz. Total of 750 synthetic vowels are presented to 5 male adults. Mean error rate and confusion matrices are taken for the 15 synthetic conditions.

RESULT AND DISCUSSION

A systematic relation between the articulation score of CV-syllables and that of vowels has been observed under several conditions of helium-speech[3]. Speech intelligibility for 15 combinations of $k$ and $r$ has been predicted on the basis of this relation. Necessary process for improving helium-speech intelligibility with respect to $k$ and $r$ is summarized in the Figure.

Unusual (uni-directional) confusion of helium vowels can be explained by the non-linear formant elevation, that is, effect of $k$ (/u/ - /e/ and /o/ - /a/) and effect of $r$ (/i/ - /e/) are superposed in the helium mixture.

REFERENCES


Figure - Necessary process for unscrambling helium-speech with respect to $k$ and $r$
THE CORRELATION BETWEEN CHANGES OF SPEECH CHARACTERISTICS AND PHYSIOLOGICAL PARAMETERS AT MODERATE ALTITUDES

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The higher the altitude and the greater the exposure, the more significant are the psychophysiological changes, among them speech. In this regard, of special psychophysiological importance are low and moderate altitudes, for the following reasons: first, a common feature of altitudes is the presence of hypoxia /decrease of pO₂/. It is precisely this hypoxia which is the principal element of most, if not all, pathophysiological states; second, the degree of hypoxia varies in dependence upon the given altitudes and this makes possible the simulation of different developmental phases in any pathophysiological process; third, low and moderate altitudes can be tolerated longer than high altitudes without injurious consequences. Due to these three above-cited characteristics, altitude can serve both as a suitable model for the study of psychophysiological changes which are encountered in various pathophysiological states as well as for the in-depth study of the nature of hypoxia as a phenomenon.

In this connection, preliminary studies have shown a high correlation between the acoustical features of speech and the prevalence of normotony, sympatheticotony and vagotony. In principle, sympatheticotony dominates in the initial phase of hypoxia while it is also a characteristic of defense reactions in stressogenic stimulus. When appropriate adjustment has been made, normotony sets in. As a rule, vagotony occurs in the terminal phase of hypoxia, which is accompanied by cardiovascular alterations. At the same time, in the terminal phase there is a deterioration of the more recent phylogenetic voice suprastructures while only the biologically rudimentary features of voice are preserved. Therefore, the degree of voice change is highly related to the degree of alteration in vital physiological functions.
A detailed investigation has been made of speech structure in the postoperative period.

Samples of standard speech and certain test-words were taken down on a tape recorder at various time intervals following the patient's coming out of general anesthesia. The speech signals were analyzed with the aid of a computer. On the basis of computer processing, it was found that anesthesia and abdominal surgery significantly change the structure of speech. The amplitude-frequency characteristics of speech were considerably altered. They stand in high correlation with the changes in physiological parameters and biochemical indicators. Namely, as is well-known, for a certain period of time - usually 12 to 24 hours after coming out of anesthesia - insufficient oxygenation or slight arterial hypoxemia perseveres. Arterial oxygen saturation dropped from a mean of 97 per cent to 93.8 per cent in the postoperative patients. This decreased oxygen content of blood is caused by several factors, such as pain, variations in ventilation-perfusion relationship etc. All these and others tend, in different ways, to change the physiological basis of the patient and thus his voice as a psychophysiological function. Therefore, speech changes can serve for the rough evaluation of the psychophysiological status of the patient at various intervals in the postoperative period. It is obvious that paralelly with the patient's general recovery there is an improvement in speech function.
THE HWIM SPEECH UNDERSTANDING SYSTEM

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OVERVIEW

HWIM (for Hear What I Mean), the speech understanding system developed at BBN as part of the recent five-year ARPA Speech Understanding Research Project (1), is designed to "understand" naturally spoken utterances relevant to a task domain of travel budget management such as:

"What is the round-trip air fare to Madrid?"
"Schedule a trip by train to New York."

Its vocabulary is presently 10897 words, and its grammar permits a habitable subset of natural English (2). The system is characterized as follows:

A digitized speech signal is first analyzed using linear prediction into a 13-pole spectral model, energy, 3 bandpass energies, F0, 3 formant frequencies, and a spectral shape measure. An Acoustic-Phonetic Recognizer uses context-sensitive acoustic-phonetic rules to produce segmental (phoneme-sized) hypotheses in a segment lattice representation that permits ambiguity in both segmentation and labeling. A Lexical Retrieval component matches word pronunciations against the segment lattice using a tree-structured dictionary that takes into account both within-word and across-word phonological rules. A Word Verifier uses a synthesis-by-rule program to produce a spectral word template from a word pronunciation plus context and matches it against the parametric representation of the speech signal using the Itakura spectral distance measure and a dynamic programming time registration algorithm. A Parser uses an augmented transition network grammar that contains both syntactic and semantic knowledge and uses a semantic network to provide factual and discourse constraints. It parses sentences or sentence fragments in either direction, and it can enumerate the words and syntactic/semantic classes permissible at each end of any sentence fragment. A Control component uses the other components to formulate, evaluate, and extend hypotheses about the interpretation of the utterance. It is responsible for guiding the system to the most likely interpretation as efficiently as possible. The result of understanding an utterance is a formal semantic representation that can be executed by the Travel Budget Executive to produce the desired response (such as printing the cost of a plane ticket to Madrid).

RESULTS

At the end of the project, on 124 new sentence length utterances by 3 adult male speakers, HWIM correctly understood 43% of the utterances, incorrectly understood 38%, and found no interpretation for 19%, requiring about 1350 times real time on a PDP-10 computer. For the same sentences in a smaller version of HWIM (409 word vocabulary), 52% were correctly understood, 33% incorrect, and 15% failed.

REFERENCES


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An insufficient supply of oxygen to cerebral tissue also alters, to a smaller or greater extent, the function of speech. Among others, the distinctive characteristics of phonemes are especially changed in hypoxia. Hypoxia erases the differences which exist among phonemes as meaning units. In this way, hypoxia diminishes speech intelligibility.

Of the inherent features of phonemes, voiced and voiceless ones and vocality and nasality are especially subject to the impact of hypoxia. Hypoxia simultaneously influences the duration of phonemes and tone movement. One of the most important effects of hypoxia is the loss of individual features of speech. Therefore, the degree to which individual speech elements are preserved reliably reflects the degree to which nerve tissue is supplied with oxygen and nutritional matter as well as resistance to hypoxia, in other words, the general physiological range of the individual. Thus, in light of all these elements, hypoxia is of high diagnostic value.
As it is known from the anatomy of the acoustic system there are many parallel paths which transmit the signal from periphery to central nervous system. Moreover the acoustic system as the possibility (at the level of nuclei) of modifying the number of parallel paths transmitting the signal. According to the hypothesis of decision theory, it can be assumed that acoustic message recognition is done by utilizing likelihood function in decision task. The problem of message interpretation is reduced to that of positioning appropriate separation lines among the receivable signals in the decoder.

Learning can be defined as the process of positioning the above mentioned separation lines.

By considering the existence of parallel paths a new hypothesis can be introduced on the behaviour of the system during the learning phase. The hypothesis is that the system varies during all phases of the learning the number of parallel working paths. In such a way the signal to noise ratio of the message reaching the control nervous system is varied.

By assuming the displacement of the above mentioned separation lines proportional to the percentage of correct message interpretation, the following relation holds for the displacement velocity of each separation line:

\[
\dot{x}(t) = -K \frac{\delta}{\delta x} P_R(x, \sigma)
\]

where

\( \dot{x}(t) \) represents the position coordinate of the separation line

\( K \) is an appropriate constant

\( P_R(x, \sigma) \) is the probability of a correct answer given by

\[
P_R(x, \sigma) = \frac{1}{\sqrt{2\pi\sigma^2}} \left[ e^{-\frac{(x+1)^2}{2\sigma^2}} - e^{-\frac{(x-1)^2}{2\sigma^2}} \right]
\]

where

\( \sigma \) is the variance of likelihood function of the signals to be recognized.

Experiments have been carried on utilizing sound subjects with the aim of verifying what above stated.

The same experiments on subjects with acoustic deficiencies can give useful information about the selection criteria of the appropriate prostheses for every patient.

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Does noise improve learning in a biological net?
TIME TO TIME RELATIONS IN THE ANALYSIS OF BABIES' CRIES

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INTRODUCTION

Babies' cries are studied as the first stage of human communication from various acoustical aspects, such as fundamental frequency and spectral analysis, as well as from medical and linguistic aspects. The duration of single cries within cry sequences and the pauses between them are discussed here.

THEORETICAL BASES

It has been shown that the duration of an average baby's cry is very similar to that of a syllable in adult language (1). It has also been claimed that human speech is controlled by fixed rhythmic units (2). Babies' cries have also been described as being made up of cycles of crying (3). It is, therefore, assumed that studying these cycles from the aspect of cry/pause durations can be instructive as to their meaningfulness in the context of babies' communication.

EXPERIMENTAL TECHNIQUES

Different techniques are used for analysing babies' cries, including e.g., spectral analysis, cepstral analysis and laryngographic analysis. For the study of cry durations, however, a simple method of writing out the tape recorded cries on to a level recorder was adopted. By halving the tape speed for cries and by suitably adjusting the level recorder, a good display of the cries could be obtained for measurements of both cries and pauses. The data was then tabulated and analysed.

RESULTS

The study was concerned with differences between hunger, pain, illness and alarm cries of normal babies, where differences of practical importance were observed on the cry/pause duration time scales. It seems justified, then, to analyse durations of babies' cries as significant not only for distinguishing between normal and abnormal babies (4), but also as meaningful utterances for different stimuli (psychological, physical, pathological, for example) of a certain individual baby. It also seems important to analyse not only the period of "regular crying" within the cry-sequences, but also the structure of the whole cycle and cry-sequence in order to find differences in the introductory, medial and final stages of the cries.

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EFFECT OF ACCENT AND SPEAKING RATE ON CONTROL OF THE VELAR ARTICULATION

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INTRODUCTION

The authors published previously some results about dynamic control of the velum during speech, based on fibrescopic observation of the velum as well as on electromyographic data obtained from the levator veli palatini muscle (LP) (1,2). In this paper we investigated the effect of suprasegmental features such as the accent pattern and speaking rate on the control of the velar articulation in Japanese.

EXPERIMENTAL PROCEDURE

A male Japanese subject repeated various speech samples at two speaking rates, slow and fast. The speaking rate was controlled by timing sound signals of 3 and 4 Hz given through an ear-phone to the subject.

EMG signals were recorded using bipolar hooked-wire electrodes inserted to LP perorally. The data combined with acoustic signals were subsequently computer-processed to obtain activity patterns as a function of time. Simultaneously with EMG recording, the velar movements were filmed at 50 fps by means of a fibrescope inserted through the nose.

RESULTS AND REMARKS

In Japanese nonsense syllable sequences there was no difference in both EMG peak value and velar height for high-velum consonant /t/, regardless of presence of an accent kernel. For low-velum sounds, nasals, there was no difference in both suppression of muscle activity and velar lowering, either. Thus, it was considered that in Japanese nonsense syllable sequences, the accent pattern has little effect on the control of the velum.

As for the effect of speaking rate, it was found that the EMG peak values for /s/, /t/ and /k/ were approximately 1.3 times greater in the fast-rate speech than in those in the slow-rate, while the minimum values of EMG during suppression for nasals were the same so as to reach the resting level in both rate conditions. On the other hand, there observed a tendency that the velar height for the high-velum consonants were lower in the fast-rate speech than in the slow-rate. There was also a tendency that the extent of velar lowering for nasals became smaller in the fast-rate speech compared with that in the slow-rate.

These findings would reflect the characteristics of dynamic control of the articulatory organ with mechano-inertial factors, although further analysis is needed in this respect. Detailed studies using our X-ray microbeam system (3) combined with EMG are under way.

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INTRODUCTION

The "ideal" vowel characteristics are influenced by neighbouring speech sounds in different contexts, this effect is called vowel coarticulation. The ideal vowel characteristics can be specified on the basis of a neutral or null context like h(vowel)t. They can be measured as articulatory features, or as acoustic features like duration, fundamental frequency, and relative power, but in this project we are mainly interested in the spectral features.

The changing spectral characteristics are generally presented as deviations from target vowel positions in the formant plane (e.g., Stevens et al., 1966; Ohde and Sharf, 1975; Bond, 1976). An improvement is obtained when not just the deviations in average vowel positions are specified, but also the complete dynamic spectral behavior (Broad and Fertig, 1970). For that purpose, formant analysis is very time consuming and needs human control.

EXPERIMENT

We advocate a straightforward real-time spectral analysis, using a parallel set of bandfilters sampled once every 10 msec. Lists of 270 CVC words spoken by three male speakers were processed.

A principal-components analysis on the variance-covariance matrix of the bandfilter spectra is then used to achieve a data reduction to two or more dimensions. In the two-dimensional vowel space the dynamic spectral aspects of vowel coarticulation can be displayed just like the deviations from the target positions.

Fig. 1 gives an example of the potential of the procedure, demonstrating the large effect final /r/ has on different vowels in monosyllabic CVr words. The initial stable vowel parts (open circles in Fig. 1) have extreme positions relative to the average vowel positions (closed circles in Fig. 1), suggesting, by the way, a pre-programmed articulatory action. The large dynamic excursions, following the stable initial parts of the vowels on their way to the /r/, are also presented in Fig. 1.

From subsequent identification experiments, using the same isolated vowel segments as in the spectral analysis, it appeared that especially for the Dutch vowels /e, o, ë/ a spectral transition is almost a "must" for correct identification. For more details of this research project we refer to Pols (forthcoming).

REFERENCES

INTRODUCTION

For studying the temporal coordinations of the articulations of the adjacent segments in speech, a large set of articulatory data covering various phonetic conditions is necessary. For the articulatory observations, cineradiography appears to be the most useful method, but there are difficulties in obtaining sufficient amount of data. A computer controlled x-ray microbeam method developed at our Lab. overcame to a large extent these difficulties. In the present study, tongue movements in the production of Japanese and American English containing a comprehensive set of C-V combinations have been observed by our new x-ray method and their temporal characteristics have been analysed.

METHOD

Three or four lead pellets were attached on the tongue, one on the lower incisor and another on the lower lip and their movements were tracked by the x-ray microbeam. The pellet movements were subsequently reproduced by the computer in various forms such as slow motion display, plot of the time function and XY-trajectories.

The speech material studied here contained the following test words. Japanese; MVjCVjæ, CVjVjæ (C=m,t,k,s V=i,e,a,o,u) in a carrier phrase "desu". English; pVP (V=10 English vowels), CVC(C=p,t,k,s,ʃ,ɹ,V=10 English vowels) in a carrier sentence "It's a___ again".

RESULTS AND COMMENTS

Measurements were made on the variations of the pellet positions at selected time-moments of consonant with regard to the difference in the adjacent sounds. For Japanese VCV sequence, the effect of the following vowel on the articulations of the consonants were in general greater than that of the preceding vowel. However, the temporal pattern of the vowel-consonant coarticulation varied depending on the vowel difference, in that some vowels (e.g. /i/) were overlapped more strongly with the consonant than other vowels.

In English CVC words, effect of the vowel was greater on the syllable final consonants than on the syllable initial consonants for the tense vowels, whereas the effect was more symmetric for the lax vowels. It appeared that, for the tense vowels, the transition to the following sounds began at later phase of the time period between the two consonants than for the lax vowels.

Effects of other factors such as suprasegmental structures, are also being investigated.

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PARAMETER DESCRIPTION OF THE TONGUE POINTS MOVEMENTS IN
THE VOWEL PRODUCTION

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INTRODUCTION

A method of multivariate analysis of the tongue shapes appears to be
useful for deriving a small number of articulatory parameters which
effectively describes tongue gestures in speech. (1) In the present
study, the movements of the tongue pellets in the production of the vow
els in Japanese and English observed by the x-ray microbeam method (2)
have been analysed, and the correspondence between the derived para-
eters and the formant frequencies of speech has been examined.

METHOD

Movements of the several pellets attached on the tongue, lower incisor
and the lower lip were recorded using an x-ray microbeam technique at a
rate of about 130 frames/sec. The speech material analysed here were
the vowel sequences in Japanese /ieaou/, /iaueo/, /iuoe/ and /ioeua/
(1000 time samples) and ten English vowels in "It's a pVp again" (500
time samples).

The movements of the pellets were approximated as a linear sum of
the independent components, 
\( (x) = J^* (xJ) + L^* (xL) + T_1^* (x_{T1}) + \ldots \)
where \( (x) \) was a vector listing the displacements of all pellets, \( (xJ),
(xL), (x_{T1}) \) etc. were the base vectors for each component, and \( J, L, T_1,
T_2, \) etc. were the articulatory variables representing the degree of
each components. The movement of the jaw pellet was approximated as
linear and represented by the parameter \( J \). The component \( (xJ) \) included
the movements of other pellets linearly correlated with the parameter
\( J \). The component \( T_1, T_2 \) etc. were derived by the principal component
analysis of the movements of the pellet on the tongue, after subtrac-
ting the effect of the jaw component. The lip movement independent
from the jaw component was approximated as linear.

Speech signal was recorded simultaneously and the relationship be-
tween the formant frequencies and the articulatory parameters was also
examined.

RESULTS AND COMMENTS

Both for the Japanese and English vowels, the components \( J \) and \( T_1 \) were
clearly related to the high-low and front-back dimensions of the vowel
gestures, respectively. The component \( T_2 \) corresponded to the bulging
of the tongue dorsum accompanied by the down and backward movement of
the tongue tip. It appeared that the \( T_2 \) component was necessary for
characterizing the vowels /u/ and /o/ in English, whereas for the Japa-
nese vowels, its contribution rate was smaller.

For the Japanese vowel sequences, in the third order multiple re-
gression estimation of the formant frequencies using four articulatory
parameters, \( J, L, T_1 \) and \( T_2 \), the contribution rate was 99% for \( f_1 \) and \( f_2 \)
and 92% for \( f_3 \).

Analysis of the more comprehensive set of data are now in progres

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(2) S. Kiritani et al.. J. Acoust. Soc. Amer. (1975) 57. 1516
This paper introduces a new application of the magnetometer to measuring the tongue movements in speech production and presents certain preliminary results of data analysis in order to investigate the dynamic characteristics of speech production process.

The monitoring system consists of two magnetometer units and a small permanent magnetic rod which was attached at a selected position of the tongue. The magnetometers detect the magnetic field produced by the permanent magnetic rod. Two sensors, \( S_y \) and \( S_y \), of the magnetometers are set at right angle of the sagittal plane of the head (Fig.1). With the long axis of rod and the sensor part of magnetometer oriented parallel, the voltage induced in the magnetometer is inversely proportional to the cube of the distance between the rod and magnetometer. Since the range of tongue motion in speech is small compared to the distance, the relationship between voltage change and distance change is nearly linear.

Movements of the tongue point were approximated by output responses of a second-order linear system of hypothetical stepwise commands for successive vowels represented as:

\[
H(s) = \frac{a \cdot e^{-TS}}{(s/\omega)^2 + 2\zeta(s/\omega) + 1}
\]

The four parameters \((a, \tau, \zeta, \omega)\) are estimated by an optimization technique which minimize the total mean square error over a time interval of vowel-to-vowel transition. Fig.2 shows an example of approximated movements as compared with observed tongue movements. Some estimated values of damping factor \(\zeta\) during the utterance of /e V e/ \((V = a, i, u, o; \text{Japanese vowels})\) are rather clustered in the limited range of less than 1.0. In this case, overshoot behavior is observed in the tongue point movement, that is, the tongue point moving from the position for the first vowel, moves beyond the steady-state position for the following vowel.

REFERENCE

INTRODUCTION

Recent physiological investigation have revealed that the activity of
the extrinsic lingual muscles, the genioglossus (GG) in particular, is
closely related to tongue movements in both speech and non-speech ges-
tures. The aim of the present study was to examine the electromyograp-
ic (EMG) patterns of GG in vowel production and to compare its results
with the patterns of tongue movements observed by means of an X-ray
microbeam system (1).

METHOD

The EMG signals were recorded during the production of various speech
materials using bipolar hooked-wire electrodes inserted to GG percuta-
neously, simultaneously with speech signals. The data were subsequent-
ly computer-processed with reference to pertinent speech events in the
test utterances taken as a line-up point of averaging procedure.

Movements of the tongue was examined by means of pellet tracking
technique using an X-ray microbeam system.

RESULTS AND COMMENTS

In the production of the five Japanese vowels /i/, /e/, /a/, /o/ and /u/,
GG was active for those vowels characterized by either the property
'high' or 'front', i.e., /i/, /e/ and /u/, while it showed only a very
low level of activity of the vowels /a/ and /o/. It was also found
that EMG signals obtained from five different locations of GG differd
characteristically in their activity patterns. These differences were
interpreted to be in correlation with differences in tongue configu-
rations in the production of Japanese vowels (2).

In order to investigate more comprehensive vowel sets, EMG studi-
es were performed on a native Swedish subject of Stockholm dialect who
produced Swedish long and short vowels, each of which was embedded in
a frame. It was observed that the patterns of GG activity also differd
depending on the location of the electrodes, with general tendency
to high activity for high-front long vowel producion. The same tend-
ency was also found in the production of the short vowels with lesser
amount of GG activity when compared with each of their longer counter-
parts. Specifically, GG was inactive for [e] production, whereas it
was quite active for [e:]. The result seemed consistent with the de-
scription that the long/short set of [e]/[e] was different from the others
(3). More detailed studies are being performed with respect to differ-
ences in the patterns of tongue movements observed by means of pellet
tracking technique in the same subject. Simultaneous recording of EMG
and X-ray data are also in progress.

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SOME RELATIONSHIPS BETWEEN ARTICULATION AND PERCEPTION OF TENSE AND LAX VOWELS IN ENGLISH

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INTRODUCTION

Electromyographic studies of the American English vowel pairs /iy/-/i/ and /ey/-/e/ reveal two different production strategies. In the first, speakers appear to differentiate the members of each pair of vowels primarily on the basis of tongue height, while in the second the basis of differentiation appears to be tongue tension.

METHOD

In order to ascertain if these differences in production correspond to differences in perception, vowel identification tests were given to the subjects of the electromyographic study. In both identification tests, subjects were asked to label the members of a randomly presented seven-step synthetic vowel continuum, /iy/ through /i/. In the first test each of the items in the continuum had equal probability of occurrence. In the second test there was unequal probability of stimulus occurrence; the first stimulus in the continuum, /iy/, was heard four times as often as any of the other stimuli.

RESULTS AND DISCUSSION

Compared with the labelling function on the equal probability test, the phonemic labelling boundary in the unequal probability test was displaced toward the anchor stimulus. However, the magnitude of the perceptual shift was greater for those subjects whose production strategy was based on tongue height than for those subjects who used tongue tension to differentiate the vowels.
The main purpose of this study (1) is to develop a most effective aid for the articulatory training of the deaf. In the first stage of this project a program for computer generation of a dynamic sagittal articulatory model has been developed. 'Dynamic' means that such a model should change and move in time according to articulatory control parameters. This goal has already been reached and nearly any desireable articulatory movement can be produced on screen not unlike an animated cartoon. (If possible examples will be demonstrated here). The main principle of this computer program is the parameter controlled calculation of intermediate points between stored data of coordinates of extreme articulator positions. Fig (1) shows the transition for the syllable 'ma' with two intermediate positions in superimposed representation of coordinate points.

As an example of a more complicated articulatory movements fig (2) shows the single pictures generated for synthesis of the German word 'male'. A Tektronix 4010 terminal with vector generation was used. In the second stage of the project the control parameters are to be taken from the acoustical output of a speaking person and should move the articulatory model in real time. The parameters for voice (movement of the glottis) and nasality (movement of the velum) are already detected effectively and employed using contact microphones at the side of throat and nose. The detection of all necessary parameters out of the acoustical signal of course bears some problems. The advantage of this procedure however lies in the possibility of a direct and real time control of the efficiency of parameter extraction. The use of the known simple procedures of counting zero-crossings and of maxima detection is at time in experimental stage.

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In the continuing search for efficient electronic aids for deaf children, modern techniques have concentrated on two aspects: the display of voice parameters (1) and the exercising of articulation features (2). It is felt that insufficient attention has been paid to fundamental exercising of the intrinsic laryngeal muscles influencing glottal vibrations. The deaf child is in an analogous situation to the young pianist who must learn finger control via scales and exercises before complicated works can be attempted. Should basic pitch movements be exercised regularly over a period of years, there is a strong likelihood that the brain's proficiency in organizing neural information to and from the larynx would increase, enabling an easier transition at a later stage to some of the socially acceptable contours of normal speech. Since regular exercising (e.g. once daily for ten minutes) can only occur after school hours, a pitch exercising apparatus must conform to a number of criteria, among which are: ease of use in a home situation; economical; and some inherent motivation factor.

It seems evident that once undue tension has been removed from the speaking process, faster progress would be possible. Tension can possibly be reduced by regular use of an apparatus in game form; and by ensuring adequate breath control. The pitch exerciser illustrated in the photograph is designed to perform the functions mentioned above by means of progressive exercising from basic voice duration to complex pitch contour control. It uses an economical yet accurate pitch extractor (3) and a pilot study has started at a school for the deaf in Pretoria using twelve children to evaluate its long term effectiveness.

REFERENCES

INTRODUCTION

A speech training technique has recently been described (1, 2) which employs a computer to analyse the speech waveform into an areas model of the speaker's vocal tract. By displaying on a CRT the area function for a correctly vocalised sound as a target, and also the area function corresponding to the pupil's current attempt, the pupil is guided to correct vocalisation by attempting to match the two shapes.

ASSESSMENT

A portable version of this aid has been developed for the training of single sounds and is being used in a Cambridge school to assist in speech training of the deaf. An assessment project has been set up and this is described in the paper. This divides into several parts. The selection of subject groups and control groups and the establishment of their speech and hearing impairments. The training procedures to be used in the classroom - the range of sounds trained and their frequency. The acquisition of data which in this case is achieved by continuous recording of the speech attempts and the displays. The assessment of progress, quantitatively from the recordings and subjectively by panel assessment of speech quality.

The assessment will finally cover a wide range of subject, age and disability groups. In the paper preliminary results are given from a 6-8 year group of profoundly deaf children.

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SOME REMARKS ON THE SPEECH RECOGNITION

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Speech and communication research embraces many aspects and many purposes. One of the chief technical tasks is being the automatic recognition of speech. This task can be but treated as a very complicated problem the solution of which should not be achieved within reasonable time.

This statement can be reinforced by the fact that all up-to-date automatic recognition equipments operate according to pseudorecognition methods. The apparently attractive practical results relate without exception to a limited vocabulary and depend very highly from the voice of the operator - also in the case of the ingenious combination of the speech analysis with the learning possibility of a computer.

We define the recognition of the speech by the possibility of a correct transformation of the heard information in a quantized system like writing by letter symbols or by Morse-signs. So, the real automatic recognition ought to be the quantization of the fluent speech into units, the determination of the linguistic quality of these units, and to transcribe the determined units into a contractual signal system. This whole process can be treated as a real automatic recognition and the result of this process can be a really recognized speech.

The difficulty of the mentioned procedure is unquestionable because first, we are forced to create a quantized information series from a continuous one and second, we must pick out the only information series of phonemes from an information parcel which contains many attributes of the actual speech sounds.

To make steps towards the solution of this very difficult problem - according to our opinion and conviction - one must first know the speech recognition process of the human brain.

One of our efforts in this direction is the investigation of the intelligibility of stop consonants /p,t,k/. The idea of hearing and evaluating palindrome texts played back in opposite direction was the first step of experiments. The intelligibility of texts remained 100%. As next step we heard variously composed syllables in back direction and surprisingly we observed 75% intelligibility of reversed p, t, k.

This last observation means that brain disregards - at least for a short time - the direction of a time-oriented signal and evaluates it only or chiefly according to its spectral content and for its correlation attributes. Further information of these results will be presented in the next paper.

Another result is the statement that we can cut the tapes with stop consonants into two parts, one of which contains the main information data of the stops. By this, to determine the phoneme quality of speech sounds seems to be a very effective method to choose "good" words for this purpose like "takoteni" and much more worse to choose "bad" ones like "how-are-you". The consequent application of this principle leads to the construction of artificial languages with adequate phonetics.
The DAWID Speech Recognition System is conceived for the speaker independent real time recognition of isolated spoken words. By hardware analysis seven binary features are derived from the incoming speech signal. These aggregates serve as input for segmentation and labeling. The reference patterns are derived by the system in a learning phase. The maximum vocabulary size is about 150 words with a success rate of about 90%. A finite state grammar guides the system successively to different vocabularies, each containing words of a given syntactical category.

As recognition system for isolated words DAWID is particular suitable for certain applications in man-machine interaction, namely for voice programming and the query task in information retrieval. The strong syntax of programming languages and in information systems allow the easy decomposition of the total vocabulary such that at one time the actual vocabulary is small, usually considerably less than 150 words.

Two such applications are being tested:

1. Voice Programming
   Since the DAWID System is installed on a PDP-15 computer, the interpretative on-line language FOCAL has been selected. By talking to the computer any program can be generated, assembled and executed. Via FOCAL the user has control of the total computer system.

2. Information Systems
   The information exchange is usually initiated by inputting guiding parameters to the system. Using voice control these parameters are generated by spoken words.

For easy performance and general applicability the DAWID software package has been rewritten in the form of a standard device handler. Thus it can be used instead of any other alphanumeric input device.
To be able to influence speech intelligibility in a room, in an electro-acoustical system etc., it is needed to know how understanding, that is recognition of messages, depends on the acoustic information offered. The acoustic information consist of a multitude of acoustic cues.

**INFORMATION INDEX AND MESSAGES**

If in an ideal case all acoustical cues, that can be spoken, reach the listener, the information index \( i \) is considered to be unity. If for whatever reason, be it distortion or masking, a part of the cues will not reach the listener, the information index \( i \) will be in proportionality smaller.

Speech communication is based on the recognition of messages. There are very complicated messages, as are ideas, conceptions etc. that one can only recognise by the correlates of not only its acoustic but also its linguistic and even logical features. Phonemes are simple messages that have the advantage of being well known and for most listeners almost -but never quite- identical in its content. Moreover, if heard in logatoms or other nonsens word, they are neutral that is not recognisable by linguistic, logical or any other than acoustical cue.

**RECOGNITION OF MESSAGES**

The chance that one misunderstands a certain message becomes larger the smaller the number of cues is that one has its disposal.

The increment \( d(1-A) \) in the error probability will be proportional to the increment in the information index \( i \), but also proportional to the error probability.

In formula:

\[
d(1-A) = C_1 (1-A) i \]

or:

\[
\ln (1-A) = -C_1 i + C_2
\]

wherein:

- \( 1-A \) = the chance that the message is misunderstood, otherwise known as the error probability or the articulation loss
- \( i \) = the information index
- \( C_1 \) and \( C_2 \) constants, depending on the message.

The probability error, or the articulation loss can be measured. The information index is much more difficult to determine. If the articulation loss for a speech message that has passed a (ideal) low pass filter, is equal to the articulation loss of the same speech material that has passed a high pass filter with in both cases the same cutt-of frequency, one can argue that at that frequency the information is bisected. With an analogue procedure starting with half the information, one can find the articulation loss for a quarter and three quarter of the cues etc.

Stewart, Fletcher e.a. found experimentally the above mentioned relation for phonetical balanced phonemes. Using own research and data out of the literature we found the relation to be valid for all kind of messages. For every message we find values for \( C_1 \) and \( C_2 \). If those values are near to each other for different messages, these messages constitute what we define as a homogeneous group. With regard to frequency filtering all phonemes belong to a homogeneous group.

It has been found that the articulation loss depends also on the signal to noise ratio (in dB) according to formulae 1 and 2. The values of the constants \( C_1 \) and \( C_2 \) are now for the different phonemes different. The consonants are still enough alike in this respect, that we can consider the consonants to belong more or less to the same homogeneous group. This is not longer the case for the vocals. A same division between consonants and vocals is found in the case of other distortions, e.g. reverberation etc. that reduce the information index.

It can be argued, that for speech intelligibility tests one needs messages belonging to a homogeneous group. As the vocals are much less affected by a low signal to noise ratio, by reverberation etc. as the consonants, it can be easily understood that the articulation loss of consonants (\( AL_{cons} \)) has been found to be a good measure for the speech intelligibility.
Introduction. La stéréophonie nous a habitué à la perception de l'espace sonore et l'on a pu remarquer que les sons provenant de directions différentes sont perçus par l'auditeur, non seulement dans des régions différentes de l'espace, mais avec des "présences" différentes.

Il est possible de lier cette dernière sensation à l'époque d'arrivée des ondes sonores et à la perception subjective du temps.

Bases théoriques. La perception d'un phénomène sonore n'est jamais instantanée; elle s'étend pendant une certaine durée. Comme cela veut pour des phénomènes de différente rapidité, on conçoit que la perception de plusieurs d'entre eux, émis simultanément, s'étage dans le temps. En particulier, la notion de simultanéité subjective est relative et fonction de la plus ou moins grande stationnarité. Elle dépend aussi d'un certain conditionnement du récepteur humain, qui, étant donné la signification affective de certains sons complexes, les perçoit plus aisément. C'est l'effet de présence, qui introduit un masque subjectif sur les sons moins "intéressants". Cet effet est à rapprocher de l'effet dit d'antériorité ou prédominance de la perception directionnelle pour les sons atteignant la tête les premiers. Quand les directions sont les mêmes, on obtient l'effet de présence, qui dans ce cas particulier est lié à un décalage temporel.

Il est logique de penser que ne doit pas être considéré l'instant d'arrivée \( t_0 \) des ondes physiques sur la tête, mais bien l'instant \( t \) où les sensations sont intégrées en perception consciente, le délai intermédiaire \( \Delta t = t - t_0 \) étant le "retard psychophysique". Mais alors \( \Delta t \) dépend de phénomènes certainement liés à la perception directionnelle. On sait bien, en particulier que la "présence" dépend de l'orientation (cône d'attention), et plus précisément de la place des images sonores dans l'espace auditif. Un couplage existe donc entre \( t_0 \), \( \Delta t \), et le repérage spatial.

Techniques expérimentales.

Lorsque l'on s'écarte de l'axe auditif (avec du cône d'attention), les images sonores paraissent, toutes choses égales d'ailleurs, moins présentes. On peut rétablir une présence équivalente en effectuant une correction temporelle \( \Delta t \). Si \( \phi \) est l'angle d'arrivée, les essais doivent permettre d'établir, pour chaque type de modulation, une carte du \( \Delta t (\phi) \) corrigée du défaut de présence.

Application. Le travail en cours concerne l'étude de la perception des réflexions successives du son dans une salle, dont l'ensemble se traduit par la sensation de réverbération. La direction d'arrivée des ondes est en relation avec les retards physique et psychologique. La perception de la "qualité" d'une salle, moins "présente" que celle de la "réverbération", est liée à la perception spatio-temporelle des échos successifs.

Références.


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LA PERCEPTION D'UNE LANGUE ÉTRANGERE EN MILIEU BRUYANT

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INTRODUCTION
Cette étude s'inscrit dans le cadre d'un programme de recherches sur l'intelligibilité de la parole en milieu industriel. Dans les usines et sur les chantiers, on peut observer de nombreuses situations de travail où la présence du bruit gêne considérablement la compréhension des messages verbaux. Mieux connaître les possibilités réelles de communication, en particulier pour les travailleurs étrangers, devrait contribuer à améliorer, à la fois, la sécurité et l'efficacité du travail.

METHODE ET RESULTATS
Des mesures d'intelligibilité ont été effectuées en laboratoire, en masquant des listes de mots français par un bruit uniforme, réglable en intensité. Deux séries d'expériences ont été réalisées : la première en utilisant des mots abstraits ou concrets de faible fréquence (listes de la C.F.A.) et la deuxième avec un matériel verbal concret de fréquence élevée (listes du C.D.F.). On donne les résultats des tests, dans le silence et dans le bruit, en fonction de la connaissance du français par les divers sujets étrangers : Portugais, Brésiliens et Chiliens. Pour les sujets qui ont une mauvaise connaissance de la langue, il faut tenir compte des pourcentages de mots répétés, ou non répétés, dans les parties connues ou inconnues de la liste présentées. La distinction entre "intelligibilité" (répétition correcte) et "compréhension" (reconnaissance correcte) est donc essentielle. Avec la liste abstraite, le pourcentage de mots répétés parmi les mots inconnus peut être supérieur à celui des mots répétés parmi les mots connus. Ce résultat paradoxal nous a amené à réaliser un nouveau test audio-visuel, dans le but de mieux séparer les niveaux de perception de la parole : acoustique (signal sonore), phonétique (signal de parole), phonologique (signifiant), sémantique (signifié). Dans ce travail, on présente les premiers résultats de ce test, qui s'apparente à la fois au test d'images de L. Kantzer et au test de rimes de W. Voiers. A l'aide de ce test on peut obtenir une estimation plus exacte du niveau pratique de perception d'une langue étrangère, c'est-à-dire de la manière dont elle est perçue, non seulement dans le calme mais dans un bruit intense. Les résultats acquis en laboratoire sont utilisés pour étudier les améliorations possibles dans la réalité industrielle en matière de communication verbale. L'étude présentée est par nature pluridisciplinaire, puisqu'en pratique, l'efficacité de la communication est souvent liée à des contraintes physiques, physiologiques, linguistiques et psychologiques.

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INTRODUCTION
Methods used for judging the influence of noise on speech intelligibility are in many cases inexact. Among others, the influence of the mutual position of speech and noise sources is not taken into account. As in the preceding research this influence was proved in free field conditions (Fig.1), the paper deals with the intelligibility of Czech masked by the wide-band white noise in a reverberant sound field in dependence on the mutual position of noise and speech sources.

DESCRIPTION OF THE EXPERIMENT
Intelligibility tests were performed in 2 rooms with different reverberation times: 0.4s and 2s. Recorded syllabic test lists (30 pseudo-open, equally difficult PB-lists, each having 100 items) were reproduced by a loudspeaker placed in front of the listener. The conventional speech level CSL determined on the basis of the statistical distribution of amplitudes in read test lists, amounted always to 70dB. The noise signal was emitted from loudspeakers placed outside the radius of reverberation (d=2m), namely in front of the listener coincidently with the speech source (1), behind the listener (2), on his left (3) and above him (5). The disturbing noise level was switched in steps in the range 45 - 80dB(C). Each listening crew consisted of 20 untrained test persons (normal hearing, aged 18 - 45 years, higher IQ). The test items were written down in normal Czech orthography.

RESULTS AND CONCLUSIONS
Results of the realized intelligibility tests are shown in Fig.2 (syllabic intelligibility I versus disturbing noise level L_n dependences; reverberation time and position of the noise source as parameters). The results prove that the influence of the mutual position of speech and noise sources on intelligibility becomes evident in the reverberant sound field, too. With rising reverberation time, of course, the intelligibility decreases and the differences between the particular noncoincident positions of speech and noise sources diminish. The order of directions as to the noise interference and above all, the significant difference between intelligibility of speech masked by coincident and noncoincident noise sources correspond but fully with dependences established in the free field. The general invalidity of the till now used multiplicative intelligibility equations (Knudsen, Reichardt) was also proved; the reverberation and noise factors are not mutually independent.
In order to investigate the effect of reverberation on speech quality in the telephone transmission system, the quality of reverberant speech in room and the quality of artificial reverberated speech were evaluated in the viewpoint of preference. To simulate the simplified behavior of reverberation in this experiment, colorless artificial reverberator proposed by Schroeder was composed. Because the reverberator has simple structure of feedback system with three amplifiers and one delay element, it is easy to give the varieties on some physical parameters of reverberation.

For subjective evaluation, preference test was carried out by the pair comparison of reverberant speech and noise added speech. The signal to noise ratio at 50% preference point is defined as equi-preference S/N, and is shown in Fig.1 with reverberation time $T_r$. The solid circles and other symbols respectively indicate the equi-preference S/N of reverberant speech in room and artificial reverberated speech. Since the former corresponds to the results by the reverberator with delay time about 25 to 50 ms., it is apparent that there is similarity between them as to preference. The difference of S/N caused by delay time is taken out from Fig.1, and plotted in Fig. 2 related to delay time of 12.5 ms.

According to Haas or Lochner and Burger, the reflected sounds within the period approximately 30 to 50 ms. are subjectively masked by the direct sound, and the speech quality is improved. After the period, they produce an overlapping of speech and this acts as masking noise, causing a reduction in speech quality.

From these consideration, it can be assumed that the relative difference of S/N at 30 ms., for example, is equal to 0 dB as a reference, because neither reduction nor improvement of speech quality takes place at the period.

Then the normalized equi-preference S/N is obtained. Furthermore normalized reverberation time $T_r$ for artificial reverberator, denoted by $T_r^*$, can be expressed experimentally as $T_r^* = -0.14 \tau_{12}/ \log_{10} g$ where $g$ and $\tau$ are gain and delay time in feedback loop of the simulated reverberator.

Through these interpretation, the results in Fig. 1 can be rewritten to the normalized representation as shown in Fig. 3. Then the speech quality of the reverberator is expressed by a single curve uniquely independent of delay time, which is applicable for telephone transmission to estimate the quality of speech with reverberation.

REFERENCES
Improvement in the Signal to Noise Ratio of Speech Signal by Use of Auto-Correlation Function

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INTRODUCTION

SPAC and SPOC recently developed are effective for improvement in the quality of noisy speech because correlation techniques are used. This paper discusses the equalizers required in SPAC and the accumulated time of data, taking into account the effects of theoretical noise reduction by correlation function on the basis of listening evaluation of speech processed by SPAC and SPOC.

SPEECH MATERIALS

Speech materials are the news lasting about 3.3 seconds recorded from FM broadcasting. They are digitized into 11 bits at a sampling rate of 10kHz after band limitation (0.2-3kHz). The succeeding procedures are simulated on a computer. Speech materials, whose signal to noise ratio (S/N) is 0, -5 and -10dB, are synthesized by reference to the amplitude cumulative distribution of speech and noise.

EXPERIMENTAL PROCEDURES

The equalizers proposed in SPAC are as follows: Cl consists of a Band Pass Filter (BPF) and an instantaneous compressor (COMP), C3 the three-channel BPF and each subsequent COMP, D3 the three-channel BPF followed by each AGC circuit. The accumulated time of data is fixed at 10, 20, 30 and 40ms by reference to conventional analysis time. Speech stimuli processed by the system with each accumulated time are presented to 7 listeners and evaluated by pair comparison test. Preference score obtained is transformed into Thurstone's psychological scale.

RESULTS AND DISCUSSIONS

Fig.1 shows the relation classified by equalizers between the quality of processed speech and the accumulated time of data. Fig.2 indicates the results by SPOC, where the S/N of input signal are shown on right side of each Figure. The accumulated time of 10ms is evidently ineffective to the present study. In the case of 40ms, the lower the S/N becomes, the higher the evaluation is. In the cases of 20 and 30ms, the highest evaluations are given by reason that the degree of improvement in S/N and the fidelity of processed speech are satisfactory. Fig.3 shows the quality comparison of speech processed by three types of equalizers. The evaluations by SPAC (C1) and SPOC are superior to those obtained by other systems. This fact presents advantages in constructing hardware because the system is very simple. SPAC and SPOC should be tested quantitatively for practical use.

REFERENCES

Our general purpose is to investigate the relations, if any, intermediary among various parameters of the same vocal signal (phoneme, word, phrase or sentence) uttered by the same subject under the same experimental conditions, but with different loudness. The vocal emissions are practically unaffected by emotion, the experimental conditions being those of a subject who reads or repeats a text in a quiet laboratory, acoustically conditioned. The increase of loudness is induced in a rather natural way in the speaker either requesting from him to be intelligible for a listener farther and farther in an anechoic room or by the masking effect of continuous stationary signals of various level and shape the speaker perceives by earphones.

The experimental material is magnetically recorded: specifically, beyond the vocal signal the glottal signal and the airflow are recorded to consent different kind of analysis that allow to relate various aspects of the phenomenon. The kind of analysis used depends mainly on the text: preliminary experiments show a different behaviour in different subjects, even if some characteristics, such as the increase of the fundamental frequency, is generally observed in all the subjects. The analysis usually employed concern the fundamental frequency, the formant frequencies, the transitions, the spectrum, the rhythm and various statistical characteristics.

The final purpose of our investigation is to find out the general and individual variance laws, if any, in vocal emissions of different loudness and the relations between normally loud voice and crying (in some subjects particularly loud voice has some characteristics proper of the singing). It is also necessary to state which parameter, among the number proposed, can be used to express the loudness of the voice to take into account for instance the statistical distribution of instantaneous or integrated values.

Having in mind the various aspects of the phenomenon, a more specific investigation considers the details of the analysis of sentences from a statistical point of view, by a long term integration, in order to get results that can be significant in the problem of the speaker identification.

A further purpose is to correlate the parameters currently used in speaker identification also for legal applications, when the voice under test is produced in various conditions, some of which induce, among other changes, an increase of loudness. The same investigation may apply to problems proper of voices taken from telephone calls and duly recorded.
The human listener can "tune in" to a certain speaker in a noisy environment, e.g., other speakers (cocktail-party effect). This is achieved by exploiting the differences between the two ear channels, in a way much more efficient than mere averaging would be. For impulse-like noise, a model was constructed by Mitchell et al. (1). Another case that can be treated relatively easily is noise uncorrelated between both channels as would occur in a room with a diffuse random-noise field. The signal \( s(t) \) is to be the same in both channels: \( x_i(t) = s(t) + n_i(t) \), \( i = 1, 2 \). According to Schroeder (2), an estimate \( \hat{s}(t) \) is obtained as a weighted mean of both channels, decreased in amplitude whenever the S/N-ratio is low. The signal power \( S \) and the noise powers \( N_i \) have to be estimated beforehand as

\[
S = \frac{x_1 x_2}{2}, \quad N_i = x_i^2 - S \quad \text{(short-time averages)}.
\]

Also the spectral properties of the signals can be accounted for (3). In effect, those parts of the spectrum where the S/N-ratio is low are suppressed, as in a Wiener filter. This can be done by applying Schroeder's method to several frequency bands of the signals \( x_i \). The bandwidths over which the power estimates are averaged should be small for good frequency resolution but wide for statistical significance. A reasonable compromise will be the critical bandwidths of the human ear. Instead of working with distinct filterbank channels as in (3), an FFT technique including a convolution of the estimated power spectra with a basilar-membrane excitation curve on a Bark scale is preferable and faster. Best averaging time constants are 20 to 40 ms.

A different method, exploiting the special structure of speech spectra rather than the averaging properties of the ear is possible when the noises \( n_i \) are nearly white; \( \hat{s} \) is estimated by a Kalman filter based upon a time-varying autoregressive system of 12th order (3). The system parameters and noise variances again have to be determined beforehand, employing an estimated acf of \( s(t) \). The results for both methods are similar, the Kalman method is slightly better.

More difficult to treat is the case of correlated noise signals, e.g., composed of speech from other speakers in different directions. Then methods of adaptive noise cancelling (4) may be applied. When the desired speaker is on the symmetry axis of the "ears", the difference \( x_1 - x_2 \) does not contain the desired signal. The difference is compared with the sum signal, again using narrow-band averaging methods. Then a suitable cancelling signal is constructed from \( x_1 - x_2 \) and subtracted from the sum signal to yield an estimate \( \hat{s} \). A single off-axis "noise" speaker can be exactly suppressed; when more speakers are present, an optimal suppression for each frequency band is obtained. Speaker separation may be further improved by taking into account the pitch structure of speech signals.

REFERENCES

INTRODUCTION
Dans le cadre du problème de reconnaissance d'un locuteur, basé sur la recherche de paramètres caractéristiques du langage, nous allons présenter les résultats obtenus par l'examen des consonnes occlusives localisées au début de paroles du type CVCV et CVCCV. Ces consonnes, à savoir k, g; p, b; d, t, sont de très bons éléments pour effectuer une discrimination entre locuteurs, étant donné qu'elles reviennent souvent dans le langage italien et qu'on peut aisément les caractériser par la durée, qui est un paramètre pas affecté par le moyen de communication et dont la mesure est très facile.

TECHNIQUE EXPERIMENTALE
Au cours de chaque session d'essais, on a enregistré sur un ruban magnétique, 10 phrases prononcées par un locuteur. Dans chaque phrase on a localisé, à l'aide d'un appareillage approprié, (1) les phonèmes à examiner, dont la durée et la forme ont été déterminées à l'aide d'un oscilloscope à mémoire. En même temps, on a déterminé, à l'aide d'un calculateur (2), le spectre de ces échantillons, dont l'observation nous a permis de vérifier les données temporelles et les caractéristiques des transitions par rapport à la voyelle qui suit la consonne occlusive.

RESULTATS ET CONCLUSIONS
Quelques résultats préliminaires, relatifs à la consonne K suivie par la voyelle A, sont résumés dans l'histogramme de fig.1. On voit que la durée de ce phonème telle que résulte des déterminations effectuées sur 50 locuteurs des deux sexes, varie entre 10 e 60 ms, tandis que, pour chaque individu, la variation de la durée moyenne, au cours de différents sessions d'essais, est de l'ordre de 3+4 ms. Cefait nous permet de considérer ce paramètre très intéressant pour l'identification du locuteur. Il faut observer, toutefois, que aussy pour le même locuteur, la durée de la consonne examinée est influencée par les consonnes qui suivent la syllabe initiale KA. Les résultats conclusifs relatifs à toutes les consonnes occlusives examinées, seront présentés au cours du Congres.

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INTRODUCTION

The selection of decision rules is one of the important problems in speaker identification experiments. It seems that an application of quasi-Bayesian decision procedures may provide a useful tool for speaker identification purposes. In the present study several decision criteria based on Bayesian statistical model of recognition were applied to a given set of acoustic parameters /extracted from the speech samples of a given population of speakers/ in order to investigate the influence of different decision criteria on speaker identification scores.

THEORETICAL BASES

If, as a measure of speaker identification quality, we take the mean value of risk given by the equation:

$$R(\psi) = \mathbb{E}_{x, \omega} \{ L(\psi(x), \omega_i) \}$$

where:

- $L(\psi(x), \omega_i)$ - loss function,
- $\psi$ - decision rule,
- $\Omega = \{\omega_1, \omega_2, \ldots, \omega_m\}$ - space of classes,
- $X = \{x_1, x_2, \ldots, x_p\}$ - space of parameters,

then minimizing $R(\psi)$ with regard to $\psi^*$, we get the optimal Bayesian decision rule.

EXPERIMENTAL TECHNIQUES

Speech samples from 20 Polish male speakers were tape recorded and analyzed for zero crossing /1/. Several speech samples of 40 seconds duration were analyzed for each subject. Each speech sample was represented by a 16-dimensional vector presenting the statistical distributions of time intervals between successive zero crossings in the speech wave. The recognition algorithm contained as subroutines several decision rules based on optimal Bayesian decision rule; these rules were as follows: classic Bayesian, Siegert-Kotelnikov's, maximum posterior probability, maximum likelihood, Neyman-Pearson's, linear combination and minimax.

RESULTS

The primary results obtained indicate that decision criteria based on Bayesian model of recognition may be successfully used for speaker identification. There are some differences between speaker identification scores obtained for different decision rules, but they are not significant.

REFERENCES

SPEAKER RECOGNITION BY MEANS OF LINEAR PREDICTION COEFFICIENTS

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INTRODUCTION

An automatic recognition of speaker may be effective if there is a method to find the reference pattern for each speaker. If the number of speakers to be recognized is as big as 10000 or more, the exclusive use of some acoustic description will not be so effective as might be expected. It is because of lack of possibility to divide the feature space into very small different subsets. It is possible to generate practically unlimited number of patterns if the feature space will be spanned by set of acoustic feature parameters, augmented by set of phonetic parameters.

In the paper an approach to this problem is presented. In proposed method, the phonemes are represented by set of prediction coefficients obtained for an all-pole model of speech and minimum prediction residual principle /1/ is applied to automatic recognition of phonemes spoken by individual speakers.

THEORETICAL BASES

The patterns of linear prediction coefficients of all/some allophones uttered by each talker are stored. Decision, that a given input utterance belongs to the class of utterances spoken by given speaker, is made on the basis of the Itakura distance D/1/ i.e. log ratio of prediction residuals computed for all of the corresponding allophones of the input and model utterances.

EXPERIMENTAL TECHNIQUES

In the preliminary experiments the reference patterns were derived from the same utterances spoken by 50 males, and repeated 10 times. The utterances were sampled at 10 kHz using 8-bit A/D converter. Hamming window of length 500 through 10000 samples was applied to the digitalized segment of speech signal. The reference pattern consisted of the first 12 linear prediction coefficients computed from first 12 mean autocorrelation coefficients, and stored into the memory. Estimated and reference signals were compared on the basis of Itakura distance.

RESULTS AND CONCLUSIONS

The recognition rate in speaker recognition experiment carried out with the use of only one vowel represented by one set of 12 linear prediction coefficients is of the order of 0.8. The recognition rate rapidly increases as the number of analysed phonemes increases. For example, the recognition rate for two vowels is of the order of 0.96.

The presented method of speaker recognition seems to be promising enough that possible application of this method will be further investigated.

REFERENCES

INTRODUCTION

The lack of a standard for Spanish on Intelligibility Tests of Rooms makes the comparison of results among researchers quite difficult.

Our experience at CIAL allow us to suggest a scheme for consideration by our colleagues.

Our method only pursues to evaluate the characteristics of the room as transmission channel excluding the biological receptor and its neurophysiological implications.

PROCEDURE

1) Material: 25 lists of 50 monosyllables each, with the following characteristics:
   The syllables belong to the phonological system of Spanish.
   Each list represents the average construction of our language.
   The different lists have equivalent comprehension difficulty.

2) Criterion for selection of listeners and speakers:
   Male and female speakers.
   Audiologically normal listeners.

3) Tests:
   Minimum number of tests in each room: 25
   Number of listeners: 25 upon 500 seats; 50 listeners for bigger halls.
   Groups of 3 syllables preceded by a short sentence.
   Control of sound level at 1 meter from the speaker with a minimum of 60 dB (A), and in the furthest seat.
   Previous training of operators: 10 lists.
   Correction of errors by the listeners themselves
   Final check by the leader of the tests

CONCLUSION

Our extensive experience indicates that following the proposed schedule good correlation can be expected between the articulation and subjective judgements of the quality of the room for speech.

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Reassessment of cues and processes in perception of the initial voicing distinction in English

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Some feature distinctions are acoustically simple and articulatorily complex while others, including the voicing distinction are acoustically complex. The latter class permit demonstrations of the sophisticated way in which perceptual processes reconcile the statistical structure of speakers' productions, the constraints of the vocal tract as an acoustical network, and the general capabilities of the auditory system.

Voice onset time (VOT) has been identified as an articulatorily simple dimension that classifies the voicing feature including (fortis/lenis and aspirated/unaspirated) in many languages (1). However a more voiceless token with a long VOT has many aspects including (a) longer separation interval between a burst and the onset of intense periodic excitation (b) more intense aperiodic (aspirated) energy in the formant transitions (c) lower intensity of F₁ transition excitation, leading to an effectively later onset of F₁ than of higher formants (d) lower absolute detectability of an F₁ transition and (e) higher average frequency of an F₁ transition onset. Other cues such as burst intensity, duration of occlusion in intervocalic position, and F₀ trajectory are relatively independent of this subset, both articulatorily and perceptually. Any cue may be traded with any other when locating the phoneme boundary.

To answer the question of which of the subset (a)-(e) are psychophysiological real, sets of synthetic syllables must be created permitting independent variation of each variable. Cue (c) was established many years ago by this criterion independently of (e) but not of (a) (2). Cue (a) was established only recently (3) as a crucial aspect of VOT. Variable (b) probably differs between languages but we found (3) that it has slight cue value in English in the reverse of the predicted direction, probably by influencing sensory extraction of (a). Stevens and Klatt (5) and ourselves (3) attributed some importance to (d) but we have recently shown that F₁ and higher transitions exert no significant effect when F₁ onset frequency is controlled, confirming Lisker's (4) advocacy of F₁ onset. Thus both separation interval (a) and F₁ onset (e) have cue value, partly because they are both discriminable in the psychoacoustical sense. But the dimension they reflect is an acoustically coherent one rather than a set of articulatorily compatible dimensions. Ecological validity then demands that variation in F₁ onset conditioned by vowel environment be handled in one of two ways. Either the voicing decision could be delayed pending vowel processing and contingent upon it, or a simple summation of the two cues could occur before decision on a vowel-independent criterion. Our results from trading one cue for another at boundary show the latter. But this in turn implies a constraint or adjustment in production offsetting interval against F₁. This occurs; low F₁ vowels such as /l/ give long separation intervals, maintaining the principle of summative trade between F₁ onset and separation interval. This is probably through active learned adjustment rather than passive aerodynamic constraint. While other production constraints involve complex, context-dependent decoding by the listener this subsystem has a strict perceptual logic requiring concessions from the speaker. The speaker is himself a listener so this is not too fanciful.

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INTRODUCTION
The object of our study is to investigate the auditory areas of Spanish vowels in correlation with the acoustic parameters of speech.

METHOD
To carry out our research the analysis by synthesis method was used. For the production of the synthetic speech we made use of an OVE III terminal analog synthesizer controlled by a PDP 12 computer.

MATERIAL
We used the structure /Vra/ as basis. V is the variant and in this place we included different acoustic vowel sounds. The Spanish vowel system consists of 5 vowels and the structure used gives meaningful lexical items with the five vowels: /ira, era, ara, ora, ure/.

The frequencies of the formants were established at intervals approximately equidistant on a logarithmic basis. The frequencies for the F1 were 250, 300, 350, 400, 450, 500, 550, 600, 700, and 800 Hz., for the F2 were 750, 850, 1000, 1250, 1500, 1750, 2000, 2300, and 2700 Hz., and for the F3 were 2300, 2600, and 2900 Hz. An arrangement of all the possible combinations of the formants at those frequencies was made. Those cases in which the frequency of F1 was equal or higher than that of F2 or the frequency of F2 was equal or higher than that of F3 were not included in the material.

PROCEDURE
The 231 items synthesized were recorded on a REVOX tape recorder. The items were randomized in order to put them to a listening test. Each item was preceded by a number for a better control. The number was said making use of real language.

LISTENING TEST
Forty student from Leon, renowned for its good Castillian accent, were asked to listen to the items on the tape and write on a sheet of paper the word they heard. On the sheet of paper we wrote a number which corresponded to the spoken number on the tape and after it " ra". The students were asked to write the missing letter.

RESULTS
The answers given were tabulated and the results obtained will be presented. We shall present, too, the conclusions at which we arrived.
INTRODUCTION

This paper deals with two specific characteristics of the singing voice spectrum. Its purpose is to investigate and to estimate the values of "singing formant"/SF/ in particular of the female singing as well as to study the formants $F_1$ and $F_2$ of various kinds of singing voices.

THEORETICAL BASES

The main characteristic of the male trained singing is the generally observed SF(1). Another characteristic of singing is the "compensation of vowels" that causes the so-called "concentration of the vowel formants" (2).

EXPERIMENTAL TECHNIQUES

We recorded the trained singing voices of 17 subjects with various kinds of the voices, from baritone to the different types of sopranos. Our samples represent the Slovak vowels [a], [e], [i], [o], [u] sung at the sustained tones as well as those in short melodic fragments. The tape recordings have been analysed by the spectrometer and the formant frequencies were determined from the total spectra gained by this analysis. The formant values $F_1$ and $F_2$ of singing voice were compared with those in Slovak speech in male voice given in (3).

RESULTS AND CONCLUSIONS

SF depends on the kind and type of voice. The frequency of SF is increasing from the value for the low and dark voice /having the lowest value/ to the higher and lighter voice types. SF did not occur in some cases of the coloratura soprano. The mean frequency peak intervals of SF were determined as follows: baritone 2.9-3.1 kHz; tenor 3-3.2 kHz; alt and mezzosoprano 3-3.5 kHz and soprano 3.2-3.8 kHz.

The frequency values of $F_1$ can not be determined in the cases when $F_0$ is approximately identical /or lies above it/ with $F_1$ in the speech. It is the case of a great part of the vocal range in female voices. The value $F_2$ of sung vowels confirm the phenomenon of the so-called "concentration of the vowel formants" in all investigated kinds of voices /see fig(1)/.

REFERENCES

INTRODUCTION

A study was conducted to evaluate the viability of the choice reaction time as a measure of phonetic similarity. The logic of the response metric was predicated on the assumption that correct identification of one stimulus in a stimulus pair requires response time proportional to the complexity of the task. Thus the more similar pairs would be characterized by the longer response latencies.

EXPERIMENTAL PROCEDURE

The stimulus set consisted of twelve initial consonants in a CV syllabic frame, consonant plus the vowel /a/, spoken by a male native English-speaking Canadian. The stimuli were recorded, digitized, and stored for subsequent auditory presentation through a PDP-12 computer (1). The testing was carried out under computer control. Two response keys were located below a remote display screen. These keys were specifically labelled for each trial by a display presenting an orthographic representation of the test and the comparison stimulus. After two seconds the test stimulus was presented through earphones. The subject's task was to respond as quickly as possible by hitting the response key corresponding to the sound presented. The responses were checked for correctness and the correct response latencies stored. A total of twenty subjects were tested on an individual basis.

DATA PROCESSING

The raw data were compiled into a symmetric proximity matrix for each subject. These were analyzed by means of a multi-dimensional scaling program INDSCAL (2) for assessment of the underlying dimensions that may account for the stimulus discriminations. This program also permits an examination of individual differences subjects exhibit in assigning weight to those perceptual dimensions.

RESULTS AND CONCLUSIONS

The dimensionality analysis indicates that the perceptual space for our stimuli is three-dimensional. In this space the stimuli are clearly separated into groups representing the fricatives /z, c, s, s/; the stops /b, d, t, p/; and the resonants /m, n, l/; plus the isolated phoneme /h/. The first dimension orders the stimuli with respect to the consonant duration, which is thus strongly suggested as the primary cue in consonant recognition. The second dimension orders the stimuli in a manner consistent with the degree of resonance, or voicing, of the consonantal portion of each stimulus. Accordingly, some measure of signal periodicity provides a secondary but still significantly informative cue. Interpretation of the third obtained dimension is at the present obscure. From the variations of the saliency of the three perceptual dimensions for individual subjects it is fairly evident that different subjects weight the same signal features differently.

These results indicate that the application of the choice reaction time methodology in investigating the perceptual basis of speech sounds appears warranted.

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A MATHEMATICAL APPROACH TO THE ACOUSTICS OF DIPHTHONGS

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INTRODUCTION

Diphthongs can be represented acoustically on a formant chart by inserting as many points for F1 and F2 as one has measured on a spectrogram, see fig. 1. These points may be assumed to define a curve which can be described mathematically.

THEORETICAL BASIS

There are principally two kinds of curve approximation which do not exclude each other: one by ways of a parabolic model, see eq. (1), or another one by ways of a polynomial formula, see eq. (2):

\[ y^2 = 2px \quad (1) \]

\[ y = a_0 x^n + \ldots + a_3 x^3 + a_2 x^2 + a_1 x + a_0 \quad (2). \]

EXPERIMENTAL TECHNIQUES

The diphthongs of the recordings from eight Portuguese speakers were sonagographed (Sona-Graph 7029A), F1 and F2 were visually ascertained, drawn by hand, and measured in Hz at five equidistant points in time. The F1 and F2 means were computed and inserted on formant charts. On the basis of the five pairs of mean formant values the vertex P0 of the "mean curve" can be roughly estimated as P0 = (521, 1175). The formula

\[ y^2 = -2px \quad (2a) \]

is obtained from eq. (1) by axis rotation (coordinate transformation), where \( \psi = \pi + 180^\circ \). It evidently describes a parabola with an axis almost parallel to that of the "mean curve". After vertex shifting and calculation of p, the final formula of the parabola approximating Portuguese [oi] is

\[ y = \pm \sqrt{351137.84 - 673.969x^2} + 1175 \quad (2b). \]

It produces the dotted parabola in fig. 1 which is an acceptable approximation.

Fig. 1: Portuguese [oi]: measurements, mean curve, approximating parabola

When the rotation angle \( \psi \) is neither \( \pm \pi \) nor \( \pm \pi \), the final formula results to have the polynomial form of eq. (2).

RESULTS AND CONCLUSION

A diphthong can be mathematically described by using the following variables: (a) rotation angle \( \psi \), (b) vertex P0 = (x0, y0), (c) factor p, (d) direction of vectorial movement. The method can be used both for diphthong analysis and synthesis. Moreover, we assume we have developed a useful tool to be applied to diphthong comparison and typology.
SEGMENTATION AND IDENTIFICATION OF PHONEMES IN CONTINUOUS SPEECH

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INTRODUCTION

This communication presents the general principles of the phonetic analyzer of our speech understanding system. The input of the phonetic analyzer corresponds to a speech signal analyzed and coded by a fourteen-channel vocoder.

The output of the phonetic analyzer is represented by a sequence of segments. A measure of confidence in the existence of each segment is calculated. The boundaries of each segment are also given together with a string of candidate phonemes ordered by the degree of likeness (Fig. 1) under the condition that this similarity remains above a fixed threshold.

DESCRIPTION OF THE METHOD

The main components of this module are the following:

- A speech-noise distinction using as criteria:
  - a measure of energy and spectral evolution.
  - A segmentation into syllables based on the variations of the energy spectral distribution allowing for the location of vowels - a segmentation into phonemes based on the temporal variations of the spectrum.
  - A more precise classification inside each class by means of linear functions, the coefficients of which are computed during a learning stage by a stochastic approximation method.

Simple phonological rules are about to be introduced in the analyzer in order to improve the recognition of phonemes whose spectral characteristics depend highly upon the neighbouring sounds.

RESULTS AND CONCLUSIONS

The percentage of errors due to segmentation into syllables varies between 5% and 10% for different speakers and speaking-rates.

The first results of segmentation and identification of phonemes obtained on test-sentences extracted from one application are summarized in fig. 2. Percentage A corresponds to the percentage of phonemes coming in first position in the ordered set of answers which are well-recognized. Percentage B, corresponds to the percentage of phonemes correctly identified independently of their position in the set of responses. The learning data were made of a set of disyllables: CV or CVVC. The percentage A is improved by about 10% when the unvoiced plosives [p, t, k] are considered as a single class. The results indicate the importance of adapting the classification parameters to each speaker. Present work is focused on the improvement of these results by introducing additional phonological rules and a more exhaustive learning set.

<table>
<thead>
<tr>
<th>number of speakers</th>
<th>number of different phonemes</th>
<th>total number of phonemes</th>
<th>Percentage of correct identification; A</th>
<th>Percentage of detection + B</th>
<th>Omission</th>
<th>Insertion</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 (learning speaker)</td>
<td>32</td>
<td>160</td>
<td>50.8%</td>
<td>87.2%</td>
<td>4%</td>
<td>14%</td>
</tr>
<tr>
<td>3 (including the learning speaker)</td>
<td>32</td>
<td>359</td>
<td>44%</td>
<td>83%</td>
<td>5%</td>
<td>13%</td>
</tr>
</tbody>
</table>

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(2) J-Y. GRESSER and G. MERCIER Auditory Analysis and perception of speech 1975 G. FANT and M.A.A. TATHAN) p. 357, 382
An approach for the effective segmentation and first-order recognition of continuous speech has been evolved using Pattern Recognition techniques (1). The following parameters are used for segmentation: total energy, voice energy (80-300 Hz), low energy (300-1000 Hz), mid energy (1000-3200 Hz), high energy (3200-7500 Hz) and zero crossing rate. The speech waveform is weighted by a 12.2 m sec. Hamming window and a 256-point Fourier transform is computed using a Fast Fourier Transform algorithm, from which energies in the different frequency bands are extracted. Each 10 m sec. 'frame' is independently categorized by a weighted minimum distance classifier into one of six mutually exclusive classes, viz., vowel (vowels and diphthongs), vowel-like (nasals and laterals), voiced stop, aspirated fricative, fricative and silence. Contiguous frames falling in the same category are grouped together to form a phoneme size segment. Composite segments where two contiguous phonemes belong to the same class are detected by their relatively long duration. These are divided into two equal segments and treated independently.

For recognition, ten parameters are used. The six parameters mentioned earlier and duration are used in common for all sounds. In addition, the first three formants are used for vowels and vowel-like sounds, energies in the ranges 3.0 to 4.2, 4.2 to 5.8 and 5.8 to 7.5 KHz for fricatives and in the ranges 0.8 to 1.8, 1.8 to 2.5 and 2.5 to 3.5 KHz for other sounds. These energies are extracted directly from the power spectrum. For the extraction of the first three formants, a selective linear prediction technique (2) using 12 predictor coefficients is employed to smoothen the power spectrum in the range 0-3500 Hz. The three peaks in this smooth spectrum are taken to be the first three formants. When the number of peaks is not equal to three (this happens 20% of the time), information from the earlier frame is used for assigning values to formant frequencies in the current frame. The vocalic sounds are further segmented into stationary and transition parts using second formant information. All the ten parameters are averaged over the stationary part of the phoneme-like segment. Recognition at this stage is accomplished by matching of parameters with stored samples only within the class, thereby reducing the matching time taken by the classifier.

The present system is trained for a single speaker and utilizes only the information in the stationary part of the acoustic segments for recognition. The performance of the segmentation and recognition systems has been found satisfactory. Information in the transition parts will be used for the more refined synthesis-based recognition (3) where inter-phoneme context will also be taken into account.

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VERSUCHSREIHEN ZUR PERZEPTION FREMDSPRACHLICHER
AKUSTISCHER FAKTOREN

Kroes-Hecht, B. M. Gruppe für Kommunikationsforschung,
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EINLEITUNG

Eine bekannte Schwierigkeit im Fremdsprachenunterricht besonders bei erwachs
senen Anfängern ist die Abhängigkeit der Perzeption von phonologischen Sy
stem der Muttersprache. Dem Kongress werden die Ergebnisse von Versuchsrei
hen mit schwedischen Studenten vorgelegt, deren Ziel es war, die Perzeption
fremdsprachlicher akustischer Faktoren zu erleichtern. Als Fremdsprache wur
de das Russische gewählt.

THEORETISCHE GRUNDLAGEN

Ausgangspunkt für die Versuchsanordnung A waren ein Vergleich des schwedi
schen und russischen Systems distinktiver Eigenschaften, der kombinatori
schen Regeln und Segmentierung und die Überlegung, dass Perzipientenfaktoren,
die auch in der eigenen Sprache die Perzeption negativ beeinflussen, vermie
den werden sollten. Die Grundlage für Versuchsanordnung B bildeten Oszillo
graphkurven, d.h. die Tatsache, dass die zu unterrichtenden Laute auf einem
bestimmten Grundton anschauliche Kurvenbilder ergeben.

VERSUCHSANORDNUNGEN

(A) Einer Gruppe schwedischer Studenten wurde ein Tonband mit schwedischen
Text aber systematisch durchgeführter russischer Aussprache vorgelegt.
(B) Der Effekt der Versuche A wurde durch Diskriminationsübungen im Sprach
labor verstärkt, bei denen die Laute vom Tonband zum Oszillographen und von
dort mit Hilfe einer einfachen ITV-Kamera auf einen Fernsehschirm übertragen
wurden.

ERGEBNISSE UND SCHLUSSFOLGERUNGEN

Während in den Kontrollgruppen noch nach einem Jahr Unterricht 12,5 -15 %
der Teilnehmer negativ auf gewisse fremdsprachliche akustische Faktoren rea
gierten, wurden sie in der Versuchsgruppe schon nach einer Doppelstunde zu
90% erfasst. Das bedeutet: 1. Fremdsprachliche akustische Faktoren in die
eigene Sprache projiziert werden dort bedeutend leichter perziptiert als in
der fremden; 2. Kurvenbilder erleichtern dieDiskrimination.

LITERATUR

Kroes-Hecht, B. Maria, Successive Additions, Proceedings of the 9th FIPLV Congress, Group 5 (Subgroup 5:4), Uppsala, 1965

Anm.1: So wurde jedes /l/ durch [¥] bzw. [ blatant entsprechend den russischen
kombinatorischen Regeln ersetzt; [ ] statt [¥] ist der häufigste Aus
sprachefehler schwedischer Russischlernender (relative Fehlerfrequenz 15) und [¥] statt [ ] liegt an 5. Stelle mit der Frequenz 7; vgl. hierzu Thelin, Nils B., Fremdsprachendidaktik, linguistische Theo
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A NEW METHOD FOR SPEECH ANALYSIS BASED ON POLE-ZERO MODEL

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INTRODUCTION

The entire process of conversion from the source to the speech signal can be approximated by a linear system with variable numbers of poles and zeros within a finite frequency range of interest. This paper presents a method for determining the optimum numbers of poles and zeros as well as their values.

THEORETICAL BASIS

The speech production process can be regarded as consisting of an input \( u \) which is either a train of volume velocity impulses or random pressure fluctuations and a quasi-stationary autoregressive moving-average process that converts the source into the speech signal \( s \):

\[
R_i + \alpha_1 R_{i-1} + \cdots + \alpha_n R_{i-n} = \beta_1 u_{i-1} + \cdots + \beta_m u_{i-m} \quad (m \leq n),
\]

where both the order \( n \) of the autoregressive part and the order \( m \) of the moving-average part vary depending on the vocal tract configuration and the source location. Deviations from the above formulation are regarded as an additive white noise with zero mean and unknown variance applied at the receiving end.

The optimum value of \( n \) can be defined as minimizing the r.m.s. error in the estimation of the speech signal by Kalman filtering, and is utilized for estimating the autocorrelation function of the source, whose spread is then used to determine the value of \( m \). The coefficients \( \alpha_i \) and \( \beta_i \) are determined from the short-term autocorrelation function of the speech signal for the values of \( m \) and \( n \) thus determined.

EXPERIMENTAL RESULTS

Utterances of CV syllables by a male speaker were sampled at 10 kHz and analyzed by a digital computer. Figure 1 illustrates (a) short-term spectrum, (b) spectral envelope given by the optimum pole-zero model, and (c) spectral envelope given by an all-pole model (\( n=10 \)), respectively for [m] and [s], and clearly indicates the advantage of the present method of analysis over conventional methods. The results can be applied to efficient coding and noise reduction in speech transmission.

REFERENCE

INTRODUCTION

The impact of inborn or acquired deafness upon children appears to be very severe and it is of great importance to try to help them in the communication with other persons. Oral language is the most natural means of communication; so notable efforts have been done during the past years to take advantage of modern technology for the design of speech understanding and speech production aids. Such devices have been designed for the reeducation of one or several parameters of the human voice. A new trend is to use the computation facilities of machines in order to perform faster and more sophisticated analyses and to store great amounts of information. We present in this paper some aspects of our SIRENE project which works on the realization of a computer-controlled reeducation system. Basically, this system provides the child with picture representation of speech and uses extensively the results of a speech recognizer (1).

SPEECH ANALYSIS IN THE SIRENE SYSTEM

The speech analysis uses classical and specific methods. In the former we especially use a FFT algorithm for the computation of correlation functions, frequency spectra and the interpretation of transfer functions (e.g. the one of the vocal tract). A bank of filters gives information about spectral density in various frequency ranges and a pitch detector supplies voicing or non-voicing indication as well as the value of the fundamental frequency of voiced sounds. We also use a linear predictive coding algorithm in order to compute the impulse response of the vocal tract model. In addition the area function of the vocal tract is computed. The specific methods have been designed to perform the analysis of the utterances of deaf children. So we evaluate the projections of time-varying contours (like pitch or intensity contours) upon a set of Tchebycheff polynomials; the obtained results permit a classification of the child voice from specific procedures. We also settle a set of exponential filters adapted to a voice or to a vocabulary on which we project the speech time-varying signal. The projections give a kind of skeleton of the sound.

USE OF AUTOMATIC SPEECH RECOGNITION

Our system uses an automatic word recognizer which performs a dynamic comparison between reference and input patterns. The similarity rate thus obtained makes it possible to estimate the proximity between the sound produced by the child and the sound expected by the teacher. This can be expressed on a visualization screen by a spot more or less closed to a target. This system is also used for the speech reeducation of limited vocabulary (e.g. the pronounced words control the movement of a vehicle in a labyrinth) or to distinguish sounds like a vowel and the corresponding nasalized vowel. In all cases, it gives an objective interpretation of the child productions according to some predefined criteria.

Speech recognition is also used as a complement to lipreading. In this case, an indication of phonemes or classes of phonemes is given from the determination of duration, voicing, spectral density or others parameters.

REFERENCE

SYSTÈME D'ANALYSE DU SIGNAL VOCAL

Le centre de ce système est constitué par un mini-ordinateur microprogrammé MULTI 20-06 d'Intertechnique (32 K octets, cycle mémoire 400ns). Nous utilisons une entrée microphone ou magnétophone, la parole codée peut être analysée directement ou stockée sur disque. Pour l'étude et la mise au point des algorithmes de reconnaissance les programmes d'analyse ont été doté de sorties graphiques qui permettent une interprétation visuelle immédiate des résultats. Nous avons pu ainsi analyser une variété de signaux de parole aussi grande que nécessaire.

CODAGE DU SIGNAL VOCAL

Nous utilisons une technique basée sur les passages par zéro de la dérivée du signal. Un périphérique spécialisé permet d'abord d'échantillonner le signal à 10 K Hz sur 64 niveaux d'amplitude, puis, à chaque extrémité du signal, d'envoyer au calculateur l'amplitude de l'extrémité et le nombre d'échantillons écoutés depuis l'extrême précédent. Le codage de l'amplitude est fondamental car il conserve des informations sur les composantes fréquentielles du signal (fondamentale et formants) et sur leurs énergies relatives.

ANALYSE DU SIGNAL VOCAL

Les programmes développés permettent :
- d'isoler les zones de signal vocal ;
- de déterminer le caractère voisé ou non du signal ;
- de détecter précisément le début de chaque cycle de voissement, quelles que soient les transitions entre phonèmes, la rapidité d'élocution et le locuteur ;
- d'analyser la composition d'un cycle de voissement en extrayant des paramètres liés aux composantes fréquentielles et à leurs énergies respectives ;
- de segments le signal en unités phonétiques.

Toutes ces analyses sont basées sur des méthodes heuristiques utilisant les extréma (amplitude et durée). Les divers paramètres extraits et leur évolution au cours du temps permettent d'identifier les phonèmes.

Le système reconnaît actuellement en temps réel une cinquantaine de mots isolés prononcés par des locuteurs masculins différents.

REFERENCES

M. BAUDRY, B. DUPEYRAT, 7ème Journées d'Etudes sur la Parole NANCY, 19-21 Mai 1976
Le développement des techniques radars des communications et des études sur les systèmes variables dans le temps nécessite l'utilisation de la théorie du spectre instantané qui est liée d'une manière univoque à la fonction d'ambiguïté.

Nous avons appliqué la théorie de la fonction d'ambiguïté à l'analyse du signal de parole. La fonction d'ambiguïté a été définie pour la première fois par Woodward dans le cas du signal radar. Elle est représentée par l'intégrale :

$$H_w(\tau, \nu) = \int_{-R}^{R} A(t) \cdot A(t-\tau) \exp(-2j\nu t) \, dt$$

La fonction d'ambiguïté généralisée est représentée par :

$$H_x(\tau, \nu) = \int_{-\infty}^{+\infty} x(t) \cdot x(t-\tau) \exp(-2j\nu t) \, dt$$

Nous disposons d'un programme de simulation d'un ambiguimètre et d'un ambiguimètre 100 points fonctionnant à 200 fois le temps réel.

Comme nous nous intéressons à l'évolution des paramètres du signal de parole, nos ambiguimètres calculent en fait :

$$H(t_0, \tau, \nu) = \int_{t_0}^{t_0+T} A(t) \cdot A(t-\tau) \exp(-2j\nu t) \, dt$$

On montre que la fonction d'ambiguïté permet d'obtenir les principaux paramètres du signal de parole tels que la mélodie et les formants.

La mélodie est obtenue par la fonction de corrélation qui, elle, est la fonction d'ambiguïté à \( \nu = 0 \).

Le premier formant est obtenu sur l'axe fréquentiel à la fréquence \( 2F_1 \) par laquelle la fonction d'ambiguïté est constante. Les maximums de la fonction d'ambiguïté à \( \tau = 0 \) correspondent aux fréquences \( 2F_i, F_i - F_j, F_i + F_j \). La fonction d'ambiguïté est indépendante de \( \tau \) pour les fréquences \( 2F_i \) et est modulée respectivement par \( F_i + F_j, F_i - F_j \) aux fréquences \( F_i - F_j \) et \( F_i + F_j \).

Des résultats obtenus pour des voyelles orales de synthèse sont présentés.
ACOUSTIC ANALYSIS AND PERCEPTUAL EVALUATION OF SUNG VOWELS

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INTRODUCTION

Singing voices of male operatic singers have often been observed to possess prominent spectral components around 3 kHz. The present study is an attempt to obtain quantitative understanding of the underlying mechanism as well as the perceptual significance of the phenomenon.

ACOUSTIC ANALYSIS

A 27-year-old high baritone singer provided the acoustic material. Samples of five Japanese vowels, /i/, / e/, /a/, /o/, and /u/, both sung and spoken, were recorded, converted into frequency spectra, from which five formant frequencies were extracted by Analysis-by-Synthesis. As shown by the examples of Fig. 1, greater enhancement of spectral components around 3 kHz in sung vowels, as compared to spoken vowels, is found to be caused by concentration of the three higher formants (F₃, F₄, and F₅), whose standard deviation σ around their mean may be used as an index for their concentration. The ratio of σ for spoken vowels to that for sung vowels ranges from 1.5 for /e/ to 2.7 for /u/.

PERCEPTUAL EVALUATION OF SONORITY

In order to examine perceptual significance of σ, two experiments were conducted. A 27-year-old female voice trainer served as subject. In Experiment 1, samples of sung vowels were presented in random order for perceptual evaluation of sonority based on five-point rating scores and values of σ obtained by acoustic analysis. The use of natural vowels, however, did not allow complete control of other factors. Experiment 2 was then performed using synthetic sung vowels as stimuli, generated by using a source waveform obtained by inverse filtering of a sample of natural sung vowels as input to a vowel transfer function, whose first three formants were held constant while F₄ and F₅ were varied systematically to produce systematic changes in σ. The results, as shown in Fig. 2, indicate an almost perfect correlation (-0.99) between subjective evaluation of sonority and σ, confirming the relationship between acoustic and perceptual characteristics of sonority in sung vowels.

REFERENCES

1. H. L. Helmholz, Braunschweig, 1862.

Fig. 1. Comparison of spectral envelopes of sung and spoken vowel /e/.

Fig. 2. Evaluation score of sonority vs. σ for various versions of synthetic vowel /e/.
Parmi les travaux relatifs à l'aide aux handicapés auditifs, il est d'usage de distinguer ceux qui visent à l'amélioration de la perception auditive (prothèses) et ceux qui s'intéressent à la production de la parole. Le présent travail s'inscrit dans cette dernière catégorie et s'applique à l'apprentissage de la prononciation de voyelles chez de jeunes enfants sourds de moins de 5 ans.

Plusieurs études ont déjà été entreprises dans ce domaine, mais elles font généralement appel à des moyens d'analyse et de reconnaissance puissants donc lourds et difficilement utilisables dans le domaine qui nous intéresse.

Le travail présenté ici vise à n'utiliser que des moyens simples, mais permettant toutefois une bonne reconnaissance des voyelles. Il est à remarquer que l'aspect rééducation a prévalu sur l'aspect reconnaissance dans le but d'obtenir un appareil utilisable par un enfant, c'est-à-dire facile d'emploi et attrayant.

A partir de l'enregistrement de voix d'enfants (le nombre d'enfants est de 60) prononçant 5 voyelles, isolées de leur contexte, nous avons obtenu un spectre moyen, les analyses étant faites, dans un but de simplicité, par des filtres de 1/3 d'octave.

La comparaison des spectres des 5 voyelles nous a permis de faire ressortir des fréquences caractéristiques de chaque voyelle (par leur amplitude relative par rapport aux autres). Ces fréquences correspondent sensiblement aux fréquences des formants, bien que les fréquences des formants n'aient pas été systématiquement recherchées.

Pour chaque fréquence retenue, et pour tenir compte des différences d'amplitude observées, il a été nécessaire de disposer des seuils, dans chaque filtre, chaque voyelle est donc reconnue comme étant au-dessus ou au-dessous du seuil fixé. Une logique de décodage permet d'obtenir l'information souhaitée.

L'examen des spectres a permis de se limiter à 4 fréquences et donc de n'utiliser que 4 filtres.

Il a d'abord été procédé à un test par filtre ; c'est-à-dire : le niveau dans ce filtre est significativement au-dessus ou au-dessous du seuil, en fonction de la voyelle. Pour un filtre, les pourcentages sont de 100 % pour toutes les voyelles. Pour les autres, le pourcentage, pour chaque voyelle, ne descend pas au-dessous de 75 %.

Cette méthode permet, à partir d'un dispositif très simple, de reconnaître les voyelles et ce, quel que soit le niveau de la parole, ce qui lui rend très intéressante dans le cadre de la rééducation de la parole.
1. Introduction:

In this contribution an attention will be devoted to a principles of the synthesis of adaptive filters that are realized as active filters RC with transforming twoports. In the paper also some results of theoretical research and practical experiments with selective networks containing voltage controlled gyrators will be introduced.

2. Block diagram of the filters:

In the practical acoustic measuring systems we need very often such selective networks that are in the dependence on the properties of the input signal automatically controlled. One simple example of such filter is shown in Fig.1. In Fig.2 a block schematic of the frequency analyser is introduced.

Fig.1.

In this example the filters $VTF_1, VTF_2$ are controlled by voltage $U_0$, that is derived from the changes of the frequencies of first harmonic components of input signal. These pass-band filters $VTF_k$ are realized by means of the voltage controlled gyrators.

3. Experimental results:

In Fig.3a is introduced a practical configuration of the voltage tuned selective network. The dependence of the measured resonant frequency of this filter is shown in Fig.3b. In the structure of the filter is connected a gyrator that is in hybrid microelectronic technology realized and produced by the TESLA Corporation in Czechoslovakia.

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ADAPTIVE LATTICE METHODS FOR LINEAR PREDICTION

Makhoul, J. 
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INTRODUCTION

In all-pole modeling of the speech signal, it has been popular to use the autocorrelation or covariance methods of linear prediction to compute the parameters of the all-pole model. However, it has become clear recently [1] that lattice methods offer significant simultaneous advantages. In this paper we present adaptive lattice methods for sequential estimation of reflection coefficients, and compare the results to more traditional block methods.

ADAPTIVE ESTIMATION

From Fig. 1, we have: 

\[
\begin{align*}
\phi_0(n) &= b_0(n) = s(n), \\
\phi_{m+1}(n) &= \phi_m(n) + K_{m+1}(n)b_m(n-1), \\
b_{m+1}(n) &= K_{m+1}(n)b_m(n), 
\end{align*}
\]

(1)

where \( s(n) \) is the original speech signal, \( \{K_m\} \) are the reflection coefficients, and \( p \) is the predictor order. There are many possible methods for the estimation of \( K_m \) [1]; here, we shall choose one particular method [2]:

\[
K_{m+1}(n) = -2 \sum_{k=0}^{n} \beta^{n-k} f_m(k)b_m(k-1) / \sum_{k=0}^{n} \beta^{n-k} [f_m(k) + b_m^2(k-1)], \quad 0 < \beta < 1. 
\]

(2)

The lower limit \( k_0 \) in the summations in (2) depends on the type of estimator memory to be used. We differentiate two types of memory: (a) Growing memory (\( k_0 = 0 \)), and (b) fixed memory (\( k_0 = n-M+1 \), where \( M \) is the fixed memory size). \( \beta = 1 \) represents a nonfading memory, and \( \beta < 1 \), a fading memory. It is common to use either a fixed-nonfading memory or a growing-fading memory.

Equations (1) and (2) determine a recursively adaptive filter whose reflection coefficients change at every time instant \( n \). This is to be contrasted with the usual "block estimation" method, where the coefficients are estimated for a block of data, one stage at a time, i.e., \( K_p \) is determined for that block of data, then \( K_{m+1} \), etc. Then, a new block of data is used to obtain a new set of estimates.

This paper will compare the performance of the adaptive and block methods.

REFERENCES

INTRODUCTION

The information of a speech signal can be transmitted by Fourier coefficients obtained by expansion of short-time magnitudes of speech in terms of Laguerre functions (1). The Laguerre functions \( l_n(p,\tau) \) (2) can be stretched or compressed by a suitable choice of the scale factor \( p \). In general, the scale factor \( p \) should be chosen in accordance with the general behavior of the approximated function with the objective of obtaining the best approximation with the smallest number of terms. Some methods for determining the optimum value of the scale factor have been reported (3). But, the nature of speech requires a different approach to this problem.

THEORETICAL BASIS

In general, in the frequency domain, the short-time magnitudes of a speech signal have several peaks. Then, if in the frequency domain the short-time magnitude is peaked at one frequency, in the time domain the short-time corresponding magnitude is a damped cosine wave. Real and continuous short-time magnitudes in the time domain can be represented in the range \( 0 < \tau < \infty \) by the Fourier series of the complete set of Laguerre functions. Then, if we set an upper limit of terms, \( N \), the integral-square error can be minimized by an optimum choice of the scale factor \( p \).

Of course, it is not possible to determine the scale factor \( p \) such that the integral-square error will be simultaneously the minimum for all formants.

Consequently, it is interesting to know which value of scale factor \( p \) will be nearly optimum when the short-time magnitude of the voiced speech sound is approximated by the first \( N \) terms of the set of the \( l \)-Laguerre functions, and when priority is given to the first two formants. It is evolved that this value be \( p = 1.35 \times 10^{-2} \).

EXPERIMENTAL TECHNIQUES

In order to estimate the intelligibility of synthetized speech, statistical measurements were carried out. The analysis was made of various difficult Serbo-Croatian sentences pronounced by an adult male speaker. Apparently, there was a clear cut deterioration of speech intelligibility when the number of Laguerre functions used for synthesis was less than seven.

RESULTS AND CONCLUSIONS

The method for choosing the scale factor for the approximation of the speech signal using the orthonormal Laguerre functions is reported. Using this method priority was given to the first two formants of voiced speech signal.

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DISTANCE MEASURE FOR ADAPTIVE FRAME RATE SPEECH TRANSMISSION - A TIME DOMAIN APPROACH

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In digital transmission and synthesis of speech signals, a high degree of data reduction without relevant loss of speech quality can be achieved using a variable frame rate technique /4/. During stationary parts of the signal, when there is little change in the relevant speech parameters, the time interval between subsequent frames is enlarged, whereas smaller time intervals between the frames are applied within the dynamic signal sections. In frequency domain, distance measures, such as rms log spectral distance /1/, are used to select the frames for transmission. On the other hand, the frame selection problem is closely related to the segmentation task which subdivides the signal into stationary and dynamic segments and which can be performed in time domain as well as in frequency domain /5, 2/.

In this paper, the results of a time domain segmentation algorithm /2/ are applied to variable frame rate transmission systems. Three speech signal parameters (absolute average of the speech signal, absolute average of the differenced signal, and the ratio of these two) are measured pitch-synchronously and interpolated to "microsegments" of three to four milliseconds. Out of these parameters, the following distance measure is evaluated (Fig. 1):

\[
r_p(n,k) = \max \left( \frac{|P(n+k) - P(n)|}{P(n+k) + P(n)}, \frac{|P(n-k) - P(n)|}{P(n-k) + P(n)} \right)
\]

From this distance measure, where \( P \) stands for any of these three parameters, the "segment length function" SLF is determined:

\[
SLF(n) = k \quad \text{where} \quad r_{p_i}(n,k) > q_{p_i} \quad \text{for at least one parameter} \quad P_i
\]

and

\[
r_{p}(n,k-1) \leq q_p \quad \text{for all parameters} \quad P
\]

Thus, the SLF is defined as the minimum distance from a given microsegment \( n \) so that at least one of the parameters is subject to a "significant" change ("significant" means that the parameter distance \( r_p \) for the particular parameter \( P_i \) exceeds the given threshold \( q_p \) for this parameter). In segmentation, this function is used as an intermediate function for the detection of stationary and dynamic segments /2/. In transmission systems, however, where an adaptive frame rate is to be applied, the SLF can be directly converted into the local time interval between subsequent frames. The procedure does not depend on a particular speech transmission system. It has been successfully applied to a transmission system developed by Längle /3/ which uses a pitch-synchronous time domain Karhunen-Loève expansion of the speech signal.

References:

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Fig. 1: Segment length function and parameter selection.
A Mouth and Ears for Computers

Accumulated fundamental understanding in the analysis and synthesis of speech is providing key impetus in enabling machines to communicate with man in the way he finds most convenient -- by natural, spoken language. In so doing, the immense capabilities of digital computers (for the storage and access of voluminous amounts of information and for fast logical decisions) are becoming as accessible as the nearest telephone.1

This paper outlines current state of progress in giving computers a "mouth" to speak and "ears" to listen to man's voice. The paper focuses on newly-devised techniques in three areas of man-machine communications:

- computer voice response
- automatic speech recognition
- automatic talker verification

Examples of emerging applications of these techniques are given and present research frontiers are delineated. Specific illustrations include experiments on: voice-directed wiring and installation of communications equipment, automated telephone directory-assistance, talker authentication by computer measurements on telephone speech, and automatic word recognition for a travel information system.

INTRODUCTION

Cette communication a pour but de présenter le système de compréhension de la parole continue actuellement développé au C.N.E.T. L'originalité de cet ensemble réside dans le fait qu'il doit être intégré à un centre de renseignements téléphoniques et qu'il doit en assumer les contraintes :
- qualité médiocre du signal
- multiplicité des tâches

L'expérience préliminaire aujourd'hui implémentée permet à l'utilisateur de composer oralement un programme en langage de conception assistée par ordinateur (C.A.O.). L'application suivante lui permettra de consulter l'annuaire téléphonique d'un département français.

MÉTHODES

Dans sa version actuelle, le système travaille aux niveaux phonétique, lexical, syntaxique et pragmatique. Il est prévu d'intégrer les niveaux prosodique et phonologique à moyen terme.

Chaque niveau est autonome et s'adapte automatiquement à l'application choisie. La parole est codée, soit à travers un vocoder à 14 canaux, soit à travers un analyseur à codage prédictif.

- L'analyse phonétique opère une segmentation de la parole et identifie les différents phonèmes (méthode qui fait l'objet d'une autre communication).
- L'analyse lexicale opère une sélection sur le dictionnaire en calculant un indice de ressemblance entre la forme phonétique et chaque mot. Ce module assure également le cadrage des mots dans la phrase.
- L'analyse syntaxique et sémantique recherche une structure syntaxique complète à travers le treillis lexical fourni par le module précédent. L'algorithme utilisé est ascendant de gauche à droite, il dérive de celui proposé par Earley. Le processus d'accélération est du type heuristique. Cet algorithme a été choisi à cause de sa souplesse vis à vis de la grammaire de l'application.
- L'analyseur pragmatique est un automate d'états finis qui exploite la question de l'utilisateur et assure une réponse vocale par synthétiseur à diaphonèmes. En cas d'incompréhension du système, l'analyseur pragmatique dirige le dialogue pour préciser tout ou partie de la phrase non reconnue.

En plus de ces modules de base, un algorithme de programmation dynamique permet de reconnaître plus sûrement un mot isolé dans la phrase.

RESULTATS

Le système a été testé pour l'application C.A.O. en temps différé sur un corpus de 160 phrases. Le dictionnaire est de 50 mots et la grammaire hors contexte du langage est de 50 productions. Le pourcentage de bonne reconnaissance de phrases est de 65 % en première position et passe à 89 % si l'on accepte deux réponses.

Ces résultats sont encourageants, car l'introduction d'une analyse prosodique et phonologique réduira considérablement le nombre des réponses des niveaux phonétique et lexical, ce qui permettra d'augmenter la taille du dictionnaire de l'application sans réduire les performances (temps et score).
INTRODUCTION

A programming language has been defined which conforms as closely as possible with phonetic and linguistic terminology. Such a language is presented in the first part of our paper. The second part indicates how this tool is used in a grapheme to phoneme transformation rules system for French, and presents some results which are very accurate.

THE GRAPHEME TO PHONEME TRANSFORMATION RULES SYSTEM

It is based on the application of an ordered set of letter-to-sound rules. The left handside of each rule indicates the grapheme involved by the rule. The right handside of each rule specifies the corresponding phoneme and eventually the preceding and/or succeeding graphemic context. The context-sensitive rules are processed in such a way that the rules related to exceptionnal pronunciations (particular pronunciation of certain words, foreign words,...) are first examined in the set. The last examined rules are the more general ones (no context is specified for the grapheme to be translated). Additional rules have been implemented to deal with the problem of (a) mute 'e' (the grapheme 'e' may not be pronounced, depending on its position in the word), (b) 'liaisons' (a final consonant silent in the isolated word is articulated before a word beginning with a vowel), (c) 'linking' (the final consonant, pronounced in the isolated word is linked with the initial vowel of the following word), and (d) numbers written as numerals.

In French, for instance, the 'S' character is pronounced /z/ if preceeded and followed by a vowel. This rule is expressed in our language by:

\[ S \rightarrow /z/ \quad \text{'vowel'} + \text{'vowel'} \]

where 'vowel' is the subscript representing the french graphemic vowels.

RESULTS AND CONCLUSIONS

results obtained in the grapheme-to-phoneme transcription of isolated words, and of words embedded in whole sentences indicate a phoneme error rate of about 0,4 %. The fact that the program makes no reference to the syntactic category of the words leads to an erroneous translation of certain words (such as the words "couvent" or "est" which are pronounced differently if they are verbs or nouns); some erroneous liaisons and linkings produced by the program could be suppressed only through a knowledge of the syntactic structure of the sentence (for example, no linking or juncture is allowed in French between the noun subject and the verb or between a singular noun subject and its following adjective).

This program has been designed primarily to serve as a flexible research tool. The set of rules can be easily defined, revised and expanded.

(a) An example of a liaison we make in French between an ending consonant and a word beginning with a vowel or an aspired 'h' is like in "nous attendons".

(b) the concatenation we make in connected speech occurs for instance in "partir" en courant".

(c) the elision of mute 'e' is apparent in "forme", "tellement".

(d) the instance, "32" is transcribed as "trente deux".

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(2) M. DIVAY, M. GUYOMARD CNET Recherches/Acoustique - Volume III (1976) 267, 290 ;
An x-ray system implemented at the University of Tokyo (1,2) has been used for collection of articulatory data including tongue, lip, and velum movements in words and sentences in English and Japanese. This paper discusses some features of our digital data base and a new data analysis program recently developed at Bell Laboratories. Typical examples of such data and some salient characteristics of articulatory gestures will also be discussed.

The articulatory data are automatically obtained in digital x-y coordinate values, as time functions of several pellets placed on the pertinent parts of the articulators. Sound is simultaneously recorded on an analog tape with frame-synchronized pulse trains on another channel. The sound waveform is later digitized and reorganized together with the pellet data.

Three recording sessions, one in English and two in Japanese and amounting to a net total of nearly 10 minutes of speech, resulted in a data base of 88 million bits.

An interactive display program allows the experimenter to select, through knob control, any portion of the specified utterance for examination of pellet movements in comparison with sound. In one mode, up to six coordinate time functions together with a desampled waveform can be displayed. In another mode, the laterally-viewed trajectories of the pellet movements are displayed with crosses showing the pellet positions for the time frame that is specified by a cursor for the waveform.

By analysis of Japanese disyllabic words containing nasal consonants in syllable initial and final positions, we have found that the final nasal /N/ shows deep lowering of the velum, whereas /m/ (or /n/) in initial position showed an inherently restricted degree of lowering of the velum. Thus, for example, the velum for a sequence /Nm/ showed a higher velum position than a single final /N/. The same point is found for English, e.g. in an utterance of /damp/ and /nap/ in a sequence with a breathing pause in between. In spite of the short duration of the nasalization in the former and the fully lowered velum before /n/ in the latter, the latter shows a significantly higher velum position.

REFERENCES

A STUDY OF VOWEL NASALIZATION USING A COMPUTER MODEL OF THE VOCAL TRACT

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In many areas of speech signal processing (such as acoustic analysis in vocoders and automatic speech recognition) the performance in nasalized vowels is not satisfactory. This paper describes work using a fairly straightforward vocal tract model with a nasal branch, to help us understand the acoustic effects of nasalization, and is an extension of similar work by Fant (1), using analogue instrumentation. The impulse responses from the glottis to the lips and nostrils are computed using a time domain method which assumes plane-wave propagation in hard-walled tubes composed of short cylindrical sections each 5 mm long. Such approximations lead to a sampled-data representation in which forwards and backwards pressure waves suffer delay in the elemental tubes and partial reflections at their boundaries. Transfer functions are computed using the discrete Fourier transform. An implementation of an earlier version of the model has been described (2).

At the junction with the nasal tract we have areas $A_p$, $A_o$ and $A_n$ of the ends of the pharyngeal, oral and nasal tracts. The corresponding incident pressure waves have amplitudes $P_p$, $P_o$ and $P_n$. The resulting pressure at the junction is $P_j = \frac{2(A_p P_p + A_o P_o + A_n P_n)}{A_p + A_o + A_n}$ and the resulting pressure waves into the three branches have amplitudes $(P_j - P_p), (P_j - P_o)$ and $(P_j - P_n)$. A first order low-pass filter is inserted in each tube section to simulate high frequency losses, which are particularly important for the nasal tract. The sampling interval is twice the transmission time of one section of the tube. Terminations at the lips and nostrils are dealt with by providing a reflection corresponding to junctions with large tubes. The glottis is kept almost closed, and an impulse injected to probe the response. A band-limited impulse is used, allowing down-sampling of the output before Fourier transform.

Currently the outputs of the model are being compared with real oral and nasal acoustic outputs, separated using a 'nose trumpet' (3).

REFERENCES

A self-oscillating two-mass model of the human vocal cords\(^1,2\) is allowed to vibrate with simultaneous lateral (\(x\)) and longitudinal (\(y\)) displacements. Bilaterally-symmetric, stiffness-coupled masses, \(m_1\) and \(m_2\), respond to forcing functions which depend upon:

- \(P_{sg}\) the acoustic pressure just beneath the vocal cords,
- \(U_{gs}\) the acoustic volume velocity entering the glottal opening,
- \(U_{g}\) the volume velocity exiting the glottal port, and
- \(P_t\) the pressure at the entry to the vocal tract.

The cord model (Fig. 1) is programmed on a DDP-516 laboratory computer, along with a complete transmission-line representation of the vocal tract and subglottal system. The dynamic behavior of the cord model is examined for a variety of glottal, subglottal and articulatory conditions.

For steady-state conditions of cord and tract controls, the \(x\)-\(y\) trajectories of the cord masses (Fig. 2) show the mechanical behavior adequately, and these data can be compared with physiological observations.

For dynamic conditions and transient conditions, the calculated behavior is better displayed by high-speed motion pictures (or a video monitor display on the computer).

We present in this paper high-speed motion pictures, made directly from the computer graphical output, of the dynamic behavior of the vocal cord model.

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The sound pattern of some German vowels can be changed by a computer programme i.e. by varying the duration of the quasiperiodic parts of their time functions. This phenomenon can clearly be observed at the vowel groups /e/, /ɪ/, /ø/, /ʊ/ and /o/, /ʊ/. In order to clarify the cause of this effect these (naturally spoken) vowels were compared with respect to their time functions and their formant frequencies. It was found that there are only small differences in the formant frequencies of the vowels of each group. The above-mentioned changes in the sound character cannot be explained in this way (1).

EXPERIMENTS AND RESULTS

The utterances of one speaker only were used for further investigations. He excited his vocal tract (VT) by an electronic speaking aid. Thus the individual speaker properties concerning articulation and excitation of the speech generating system had not to be taken into account. On the other hand the articulation positions of the VT and its correlated transfer functions could be compared when pronouncing the vowels concerned.

To describe these transfer functions completely (e.g. in their magnitude and phase characteristics), both the excitation function of the speaking aid and the output signals perceived as articulated sounds were registered simultaneously. Then the required transfer functions, which specifically describe the vowels, could be calculated with the aid of a computing system for signal analysis. In all cases except that of the pair /o/, /ʊ/ good agreement was found when comparing the complex transfer functions of the vowels of each pair. As the vowels of each pair differ only in their duration and not in the articulatory position of the VT it seems that the difference in interpretation of these vowels by the human auditory system is to be attributed only to the functioning of auditory perception (2).

In current investigations it shall be clarified whether the differing auditory interpretations of these vowels are only a special problem of the German language. A report on the forthcoming results will be given at the conference.

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(2) W. Döring, W. Endres, Acustica 31 (1977) September, to be published.
A METHOD FOR INVERTING ARTICULATORY-TO-ACOUSTIC TRANSFORMATION IN THE VOCAL TRACT BASED ON A COMPUTER-SORTING TECHNIQUE

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INTRODUCTION

We present, in this paper, numerical methods for studying the relationship between the shape of the vocal tract and its acoustic output. For a stationary vocal tract, the articulatory-acoustic relationship can be expressed as a multidimensional function of a multidimensional argument: \( y = f(x) \), where \( x \) is a vector describing the vocal-tract shape, \( y \) is a vector describing the resulting acoustics, and \( f(x) \) is a function relating these vectors. Assuming that \( y \) may be computed for any \( x \), we describe a procedure for inverting \( f(x) \). Inversion by computer sorting consists of computing \( y = f(x) \) for many different values of \( x \) and of sorting the resulting pairs \( y, x \) into a systematic ordering based on the vectors \( y \). Finding a value of \( x \) corresponding to a given \( y \) consists simply of looking up the desired \( y \) in the sorted data and obtaining the \( x \) associated with it. Figure 1 indicates the process graphically. Points \( x_1, x_2, ..., x_n \) are mapped from the \( x \)-space to the \( y \)-space to obtain points \( Y_1, Y_2, ..., Y_n \) as shown. The \( y \)-space is sorted into cubes. If two or more points in the \( x \)-space produce identical, or nearly identical values of \( y \), these values will be sorted into the same cube or into neighboring cubes. Thus, in the sorted data, it is easy to find all values of \( x \) associated with a given \( y \).

The local nature of \( f(x) \) is determined by linearization in a small neighborhood. Larger regions are explored by extending the linear neighborhoods in small steps. Application of this method for determining parameters of an articulatory model corresponding to a given set of formant frequencies is presented. A method is also described for finding articulatory regions (fibres) which map into a single point in the acoustic space.

RESULTS

We used two types of representations to define the vocal-tract shape in our study. In the first case, the vocal-tract shape was described by 20 cross-sectional areas specified at equidistant points along its length. In the second case, a four-parameter articulatory model was used to define the vocal-tract shape. The articulatory parameters were: (1) location of constriction, (2) area of constriction, (3) area of mouth opening, and (4) length of the vocal tract. In computing the transfer function of the vocal tract, losses due to finite mass of the vocal-tract walls, glottal leakage, friction, heat conduction, and radiation at the lips were included. Our results show that widely different vocal-tract shapes can have same set of formant frequencies and amplitudes. Examples of different vocal-tract shapes having identical frequencies and amplitudes of the first three formants corresponding to vowel /a/ are shown in Fig. 2. The right side of the figure shows frequencies, amplitudes, and bandwidths of the first four formants. Sounds produced by various articulations along a fibre were synthesized and were compared by informal listening tests. These tests show that often a given sound could be produced by many different vocal-tract shapes.
INTRODUCTION

This paper discusses a method for improving the estimation of vocal tract area functions either directly from acoustic speech waveforms on the basis of the linear prediction analysis method or from given acoustic data.

METHOD AND RESULTS

An important problem in obtaining a better estimate of the vocal tract area function in this situation is how to handle the losses involved in actual speech production. The linear prediction model assumes a lossless vocal tract with all the losses lumped together at the glottis end (1). This causes rather substantial differences in boundary conditions at both the lip and glottis ends as well as in the vocal tract itself, assuming the most realistic speech production model. Since the current technical level of the LPC analysis does not allow an easy mathematical handling of the individual losses, such as those due to friction, heat conduction, wall vibration, and lip radiation, a simple approach is to find a transformation of the most realistic model into the LPC model by making certain corrections to formant frequencies and bandwidths. For this purpose, a synthesis model as proposed by Ishizaka and Flanagan (2) was utilized. A frequency correction chart from measured formant frequencies to resonance frequencies of the LPC model was made by using the synthetic vowels generated by the above synthesis model. The chart is given in Fig. 1(A) and estimated area functions for /a/ and /u/ are shown in Fig. 1(B) and compared with the original ones. Similar corrections to bandwidths further improve estimation accuracy. However, it is necessary to automatically and reliably process the closed phase of a glottal cycle for the better estimation of bandwidths. The corrections applied according to each vowel category are expected to give better area function estimates for known input vowels.

REFERENCES


Fig. 1. (A) Formant frequency correction chart. (B) Estimated area functions.
INTRODUCTION

Acoustical diagnostic methods in laryngology and phoniatry consist in a detailed analysis of the information content of the speech signal, viz. the acoustic pressure $p(t)$ of the speech wave or its spectral presentation $R(\omega) = \mathcal{F}[p(t)]$. The time and/or frequency characteristics of the voiced segments of the speech signal contain essential information which is related both with the physiological properties of the larynx source and with the anatomical structure of the vocal tract (1).

THEORETICAL BASES

The human larynx source is a feed-back aerodynamic oscillator whose output, i.e., the volume velocity $U(t)$ through the glottis is related by the differential flow and motion equations with the equivalent mechano-acoustical constants of the vocal folds. Any anomaly of the larynx or restriction of its motive capability is expressed in terms of the physical parameters of the larynx model, which thus influence the shape of the glottal wave $U(t)$ being analysed both in the time and in the frequency domains (2).

The nasalization of an oral vowel caused by cleft palate results in theoretically predicted and experimentally proved modifications of its spectral characteristics. Besides additional formants which do not exist in purely oral articulation, some antiformants appear due to the shunting effect of the nasal tract (3).

EXPERIMENTAL TECHNIQUES

A two-mass model of the larynx source described by the differential flow and motion equations was analyzed and the relations between its physical parameters and the equivalent constants of the biological system were determined. An analogue model of the human vocal tract including the shunting effect of the nasal tract was designed and built and the influence of cleft palate on the spectral characteristics of oral vowels was investigated both analytically and experimentally.

The results of the analytical and model investigations were then verified under clinical conditions. The normalized utterances of 100 subjects were analyzed in the time and frequency domains using both the analogue and digital techniques. The investigated speech organ's disorders involved the unilateral and bilateral paralysis of laryngeal nerves and the cleft palate (med.: palatoschisis molle).

RESULTS AND CONCLUSIONS

Essential diagnostic information concerning the considered larynx disorders is contained in the distributions of the following parameters: (a) the larynx tone frequency $F_0$, (b) the instantaneous and peak values of the $p(t)$ function within the voiced speech segments, (c) the respiratory and/or articulatory pauses within definite utterances.

The nasalization effect due to cleft palate may be evaluated by the measurable modifications of the formant-antiformant structure of oral vowels under definite phonation and articulation conditions. It confirms the usefulness and convenience of objective acoustic methods in computer-aided medical diagnosis of speech organ disorders.

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The vocal cord should be regarded as a layer-structured vibrator as described in a separate paper in this congress. The purpose of this film is to demonstrate the behavior of the layer-structured vibrator in different laryngeal adjustments of normal subjects and also in various pathological conditions. The conditions are normal heavy voice, normal falsetto, recurrent laryngeal nerve paralysis, sulcus vocalis, vocal cord polyp, polypoid vocal cord, amyloid tumor of the vocal cord, epithelial hyperplasia of the vocal cord, and glottic carcinoma.

From a mechanical point of view, the vocal cord can be regarded to have three layers, i.e. the cover, transition and body. As a first step of model studies, the three-layer structure was simplified into a two-layer structure which consisted of a cover and a body. A standard shape of the vocal cord during phonation was determined on the basis of morphological evidences. Actual vibration in the various conditions described earlier was investigated with ultra-high speed photography. Changes in mass and stiffness of the cover and the body in those conditions were estimated from the findings of histopathological, electromyographic, and ultra-high speed cinematographic investigations. With all of these, behavior of each layer in a frontal section within one vibratory period was schematized.

The film shows the actual vibration viewed with ultra-high speed photography and the schematized behavior of the layer-structured vibrator in each condition. Two examples of the schemata are shown in Figs. 1 and 2.

![Fig. 1 Normal heavy voice](image1)

![Fig. 2 Vocal cord polyp](image2)
INTRODUCTION

The human can produce voice of a great variety of fundamental frequency and tonal quality using only a single pair of vocal cords. This indicates that the vocal cords can become vibrators with many different properties. The purposes of this paper are to describe the structure of the vocal cord which is adequate to the task and to discuss its mechanical property.

STRUCTURE OF THE VOCAL CORD

Light and electron microscopic observations were conducted with human vocal cords. The networks of the blood vessels were investigated with human and canine larynges. The results led us to the conclusion that the vocal cord consists of multiple layers of tissue, all having different mechanical properties. Histologically, the vocal cord consists of mucosa and muscle. The mucosa, in turn consists of epithelium and lamina propria. The lamina propria has three layers: the superficial, intermediate and deep layers. The epithelium can be regarded as a capsule whose purpose is to maintain the vocal cord's shape. The superficial layer of the lamina propria is poor in fibrous components and appears loose and pliant. It may be regarded as a mass of very soft gelatine. The intermediate layer is primarily composed of elastic fibers which are like rubber bands. The deep layer is dense with mostly collagenous fibers which are more like cotton thread. The muscle, i.e. the vocalis muscle, is the main body of the vocal cord and is like a bundle of rather stiff rubber bands. Fig. 1 shows the structure of the vocal cord schematically.

From a mechanical point of view, we differentiate the layers into three sections: the cover, consisting the epithelium and the superficial layer of the lamina propria; the transition, consisting of the intermediate and deep layers of the lamina propria; and the body, consisting of the vocalis muscle.

STIFFNESS OF THE VOCAL CORD

With excised human and canine larynges, stiffness of each layer of the vocal cord and that of the entire vocal cord in varying laryngeal adjustments was measured. The ratio of the stiffness for the cover to that of the transition to that of the body can be estimated approximately 4 to 2 to 5. Contraction of the cricothyroid muscle results in an increase in the stiffness of the entire structure. Activation of the vocalis muscle appears to cause an increase in the stiffness of the body and a decrease in that of the cover.

Fig. 1
INTRODUCTION:
In the studies of intonation, the fundamental frequency (Fo) contours of speech are often decomposed into localized components (such as rises and falls) and a non-localized one: the baseline. We describe here a possible mechanism generating the baseline, that characterizes a general Fo fall along the sentences. A superposition of the Fo contours of 11 American English sentences spoken by one speaker is shown in Fig.1, as an example. The general falling characteristic of the contours seems to be evident, and may be represented by the straight line on Fig.1. In our analysis, the magnitude of the Fo drop in the baselines varies from 20 Hz to 30 Hz, depending on the subject, and not significantly on the length of the sentences. The subglottal pressure drop along the sentences may be responsible, in part, for the specification of the baseline; however, it is not presumably sufficient to generate such amount of Fo drop. Therefore, it is reasonable to speculate some other mechanisms related to the baseline.

EXPERIMENT AND RESULTS:
The variations of the length of the laryngeal ventricle, which roughly corresponds to the vocal-fold length, has been measured using a cineradiographic technique. The same speaker read four short sentences in this experiment. In Fig.2, the data points representing the frame to frame variations in the ventricle length and the corresponding Fo contours are shown for one of the four sentences. These points exhibit a considerable amount of jitter. Nevertheless, the straight line determined by a least square criterion for those points indicates a negative gradient, as seen in Fig.2, for every sentence. The average of the magnitudes in the shortening of the vocal-folds is estimated to be 3.8 mm for a entire sentence. According to published data (2), a 'Fo-vocal-fold length sensitivity' is in the order of 10 Hz/mm in the interested Fo range. Thus, the shortening of the folds along the sentence is sufficient to account the Fo drop of the baseline.

It is speculated that the 'tracheal pull' acting on the cricoid cartilage may be responsible for the shortening of the folds. A decrease in the lung volume along a sentence, perhaps, causes an increase in the tracheal pull, which leads to a gradual downward tilt of the coida, and thus, to the shortening of the vocal-fold length.

REFERENCES:
(1) The study reported here has been carried out while the author was at the Research Laboratory of Electronics of the Massachusetts Institute of Technology.
ANALYSIS OVER THE CLOSED GLOTTIS INTERVAL

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INTRODUCTION

For speech recognition, it is necessary to select a set of parameters that describes best the characteristics of the articulation process rather than those of phonation. Speech analysis deals with the most appropriate method for extracting these parameters and the conditions that emphasize most the articulatory features. In this respect, the extraction of the vocal tract (VT) area function of non-nasalized voiced sounds using LPC appeared worthy of interest. This paper deals with the detection of and the analysis over the closed glottis interval (CGI). It is known that the VT resonances are different in the closed - and open - glottis conditions. In the CGI, the subglottal tract is decoupled and the resonances are those of the VT proper.

DETECTION OF THE CGI

Previously, we presented a highly reliable, non-iterative algorithm (1) for the detection of the CGI, which proved to be heavily time-consuming. In consequence, a second approach has been devised. It is based on the location of a very short interval of the speech signal characterized by an average spectral slope that is minimal, whilst the signal energy is close to maximal. The algorithm calls for a gross estimation of the CGI location, which is given by the peak value of the signal trajectory length index

\[ I(kT) = \sum_{n=0}^{N-1} (e_{k+n} - e_{k+n-1})^2 \]

\( I(kT) \) increases with average energy and high frequency oscillations. The instant of glottal closure (IGC) is located a few sample periods prior to the instant \( kT \) for which \( I(kT) \) is maximal. Its location is determined by calculating the very short term spectral slope index \( I_p(nT) \) for the 10 preceding instants. \( I_p \) is evaluated as the algebraic sum of three real adapted pre-emphasis zeros computed successively. The IGC corresponds to the minimum of \( I_p(nT) \).

ANALYSIS OVER THE CGI

If \( N \) represents the IGC, then the analysis sequence is bounded by \( n=M-p \) and \( M+N-1 \), where \( p \) is the predictor order used. The average spectral slope of the sequence is fully equalized, then the signal is partially windowed. The covariance method of LPC is applied for parameter estimation. Finally, BW damping (=50Hz) is applied to reduce filter instability probability.

RESULTS AND CONCLUSIONS

The method for CGI detection has proved to be reliable for male voices. Its application to French oral and nasal vowels indicates that the minimum slope criterion together with full equalization improves considerably the conditioning of the covariance matrix. Filter instability is rare. We apply a unique equalization strategy for all sounds which was not the case with other analysis modes (1). Finally, analysis over the CGI emphasizes the phonemic characteristics of a speech sound (see figure (1)).

REFERENCES

The solution of the problem of automatic speech recognition has been one of the most important aims of speech research. Although physical parameters of some speech sounds /for example vowels/ are known well enough in these days. These parameters can be useful for recognition, but the distinction of stops by traditional methods knowing only their physical parameters is not possible.

In our experiments we try to examine these stops and at the same time we hope, that we can give some data for the understanding of hearing process at speech perception, too.

Our work has been based on the experimental observation, that listening backword to the meaningless words picked up on tape, the intelligibility of stop consonants are much more better than expected. Two-syllablic, meaningless words were recorded on tape in an anechoic chamber by studio tape recorder. All Hungarian stops took place in various positions and combinations. These tests were played backward and forward for 20 to 30 years old men and women listened to the test under dynamic headsets or by electrostatic loudspeaker.

The intelligibility of Hungarian stops in meaningless words at various sound connections was examined by subjective method in a series of experiments. The average intelligibility of stop consonants played backward was found as 75%. The question is, how the reversed sound-effect can give nearly the same sound impression, when these are aperiodical, quickly varied and impulse type ones /Figure 1/.

A certain transient part of unvoiced stops were cut out by mechanical or electronical way. The obtained words were played both normal and backward directions. The intelligibility of these incomplete stops was examined with the aim to know, which acoustical segments had been responsible for the identification of stops, and which segments could have been taken into account as secondary or just redundant factors.

At last we wanted to examine the minimal duration of a single segment, which is enough yet to identify the stops.

Faulty-matrixes were made from the intelligibility data obtained by this subjective method. Considering the experiences about masking effect in time, the obtained faulty-matrixes were compared to the physical parameters of stops.

INTRODUCTION

Discrete linguistic units, such as phonemes and morae, are converted through the process of speech production into continuous time-varying trajectories of formant and fundamental frequencies. This paper describes experiments conducted to elucidate the perceptual process through which these trajectories are segmented to correspond to their underlying units.

EXPERIMENTAL METHOD

Synthetic speech sounds were generated with formant and fundamental frequency trajectories simulated by step responses of critically-damped second-order linear systems to produce close approximations to those of natural utterances of two-mora, two-vowel words. Perceptual boundaries between segmental or suprasegmental units were determined by the method of constant stimuli; by truncating each synthetic word at various points in time to be presented in random order, and by asking subjects whether they heard each stimulus as consisting of one unit or two. Perceptual boundary between two units was defined as the point of truncation where the two answers were found to occur with equal probability. Boundaries obtained using heads of truncated words were generally different from those obtained using only the tails.

RESULTS

Figure 1 shows perceptual phoneme boundary in synthetic [au], indicating that perceptual onset of the second phoneme is delayed by 40 msec or more as compared to onset of formant transition, and is considerably affected by the rate of formant transition. The effect of fundamental frequency transition was found to be rather small.

Figure 2 shows perceptual mora boundary in two word accent types of synthetic [aa], indicating the existence of a similar delay between onset of fundamental frequency transition and perceptual onset of the second mora, as well as its dependency upon word accent types.
INTRODUCTION

The purpose of the present study was to investigate the perceptual cues to the word-final fricative-affricate contrast in English. Previous research (Gerstman, 1957, unpublished) has shown that in word-initial position, friction rise-time and friction duration cue the contrast. The experiments reported here sought to evaluate the perceptual relevance of these cues in word-final position. In addition, the perceptual salience of a closure interval (i.e., the interval between the offset of the vocalic transition and the onset of friction) was assessed.

EXPERIMENTS AND RESULTS

In the first experiment the sentence "Put it in the ditch." was recorded. Under computer control the closure interval preceding the friction in the word "ditch" was systematically varied over a range of 100 msec. Ten adult listeners identified the final word in the sentence as "dish" at closure intervals of less than 40 msec, and as "ditch" at intervals greater than 60 msec.

In the second experiment the sentence "Put it in the dish." was recorded and the (inserted) closure interval was varied as in the first experiment. At closure intervals greater than 80 msec listeners reported hearing "ditch," while the originally intended "dish" was heard at closure intervals of less than 50 msec.

In the third experiment slow and rapid rise-times of the friction in "dish" were generated and the closure interval was varied over a range of 20 to 150 msec. Listeners identified the fast rise-time stimuli as "ditch" at smaller values of closure duration than the slow rise-time stimuli.

In the final experiment three durations of friction in the word "dish" were prepared and the closure interval was varied as in the third experiment. With shorter friction durations, a smaller closure duration was sufficient to afford identification of stimuli as "ditch."

CONCLUSIONS

The results of these experiments indicate that friction rise-time, friction duration, and duration of closure interval are perceptual cues to the fricative-affricate contrast in word-final position in English.
Consonant-vowel-consonant (CVC) syllables manifest time-varying formant frequencies but listeners generally assign a unique vowel-color to the context-embedded vowel. This paper is concerned with characterizing a listener's perceptual mapping from dynamic formant patterns to static vowel images. The formant values of the static vowel which best matches in vowel color the dynamic vowel are determined by an adjustment procedure. Vowel matching by the method of adjustment allows the study of vowel-color shifts both within and across categorical boundaries. Our experiments employed short synthetic CVC stimuli roughly 200 msec in duration. The consonant context was always symmetric and up to now we have explored the effects of the semivowels /w/ and /j/ and the stops /b/ and /g/. The opposition in the place of production is signalled in each case by equivalent but oppositely directed transitions of the second formant. The first formant variation is identical for each pair of semivowels or stops, respectively.

Lindblom and Studdert-Kennedy [J. Acoust. Soc. Am. 42, 850-843, 1967] reported on vowel-categorization of [wVw] and [jVj] stimuli and found that subjects adjusted their categorization of the vowel continuum in the semivowel contexts in such a way that the transitions were permitted to undershoot the target frequencies for the vowel. Identification responses indicated that listeners tended to compensate for the formant frequency undershoot associated with vowel reduction. I have reported recently [P. Mermelstein, J. Acoust. Soc. Am. 58, 556, 1975] that in vowel-matching experiments the second formant frequency of the vowel matched to the dynamic vowels of [wVw] and [jVj] stimuli is shifted in the direction of the formant transition. Instead of compensating for an expected undershoot by extrapolation as found by Lindblom and Studdert-Kennedy, these subjects underestimate the extreme vowel formants and shift the categorical boundary so that the steady-state formant boundary is exceeded for the desired vowel category to be attained.

We can identify two groups of subjects with significantly different response patterns. Group A matches the /ɪ/-/ʊ/ vowel continuum in essentially a continuous pattern irrespective of consonantal context. The effects of context are to shift the formant frequency of the match in the direction of the formant transition by amounts ranging up to 300 Hz for the semivowel context when the transitions range up to 1200 Hz. These subjects exhibit a corresponding categorical boundary shift in the opposite direction. A second group of listeners, group B, matches the same vowel continuum in a highly categorical manner, i.e., the matches shift abruptly from a frequency range appropriate for /ɪ/ to one appropriate for /ʊ/ as we cross the corresponding categorical boundary for that subject. These subjects exhibit a categorical boundary shift of nearly 100 Hz in F₂ in the direction of the initial and final F₂. Vowel matches at the extremes of the continuum, where the formants are well within the regions appropriate for those categories, show shifts directed towards the center of the vowel category irrespective of the transition imposed by the context. Matches that correspond to extrapolation of the formant trajectories for both semivowel contexts are found only between the two context-shifted categorical boundaries. Listeners belonging to both groups A and B exhibit undershoot in matching the same vowels in a [bVb] or [gVg] context. Context dependent shifts of the match may be significant but rarely exceed a 200 Hz movement from the extreme formant frequency attained.

One may conclude that the predominant result is a vowel-shift that corresponds to a time-integration of the formant-frequency information in the region of the vowel segment. Compensation for expected undershoot is found only for some listeners in contexts where the transitions are sufficiently slow and when the vowel is otherwise ambiguous. These listeners cannot match vowel color without reliance on phonetic processing.
ON THE INFLUENCE OF CONTEXT UPON PERCEPTION OF VOICELESS FRIкатIVE CONSONANTS

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INTRODUCTION

The acoustic characteristics of voiceless fricative consonants /s/ and //, expressed in terms of frequencies of one zero and two poles within the frequency range of 0-5 kHz, are found to be affected by their vowel context, which in turn suggests that perception of these sounds may be influenced by context. The present paper describes results of perceptual experiments on the influence of adjacent vowels upon identification of synthetic voiceless fricative consonants.

EXPERIMENTAL TECHNIQUES

Synthetic /CV2/ syllables, generated by computer simulation of a terminal-analog speech synthesizer, were used in identification tests to determine the phoneme boundary between /s/ and //. The consonant synthesizer consisted of a random noise generator followed by one zero and two pole characteristics, while the vowel synthesizer consisted of a buzz source followed by five cascaded poles and an emphasis characteristic. A set of ten points were selected over the entire range from /s/ to //, represented by a straight line in the three-dimensional parameter space constructed by frequencies of the zero and the two poles. The formant patterns used for the vowel part were obtained by simplifying those found in utterances of a male speaker.

Effects of V; upon consonant perception were investigated using synthetic syllables of /VCa/-type, while effects of V, were examined by using /aCV2/-type syllables. These syllables were presented in random order at intervals of 4 sec for forced identification. The subjects were two male adults with normal hearing.

RESULTS

Figures 1 and 2 respectively indicate influences of the preceding and the following vowels on the phoneme boundary between /s/ and //, expressed in terms of frequency of the spectral zero. While rather small variations are found in influences of V;, a marked difference is observed between /u/, /o/ and /a/, /e/ as V, suggesting that the effects of lip articulation of V; are reflected on the perception of the preceding consonant. These results are applicable both to synthesis and automatic recognition of voiceless fricative consonants.

REFERENCES


Fig. 1. Influence of the preceding vowel on phoneme boundary between /s/ and // in /CV2/.

Fig. 2. Influence of the following vowel on phoneme boundary between /s/ and // in /CV2/.
AUTOMATIC RECOGNITION OF SEMIVOWELS IN SPOKEN WORDS

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INTRODUCTION

Based on an approximate formulation of the coarticulatory process at the acoustic level, a scheme for reliable segmentation and recognition of connected vowels has been established. The present paper describes its extention to recognition of semivowels in word context.

EXPERIMENTAL TECHNIQUES

Preliminary analysis of a number of connected utterances of vowels and semivowels indicated that target formant frequencies of semivowels /j/ and /w/ were respectively quite similar to those of vowels /i/ and /u/, while durations of semivowels were significantly different from those of vowels, suggesting a recognition scheme in two stages.

At the first stage, the input speech is divided by Analysis-by-Synthesis of formant trajectories into segments that possess a set of target formant frequencies corresponding to the five Japanese vowels /a/, /i/, /u/, /e/, and /o/, under the condition that target formant frequencies of the semivowels /j/ and /w/ are respectively identical to those of the vowels /i/ and /u/. The vowels /a/, /e/, and /o/ can be uniquely recognized at this stage, but segments which possess target formant frequencies of vowels /i/ and /u/ require further processing for their ultimate recognition.

At the second stage, these segments are classified into semivowels, /j/, /w/, vowels /i/, /u/, and their sequences /ij/, /uw/ on the basis of their durations and the information concerning the speech rate.

EXPERIMENTAL RESULTS

The validity of the proposed scheme was tested by a recognition experiment on 270 utterances of meaningful words containing vowels, semivowels, and their sequences pronounced by two male speakers.

At first, instants of command onset were extracted from the formant trajectories, and the segments thus obtained were temporarily classified as one of the five vowels. Segments corresponding to /j/, /i/, and /ij/ were then classified by using optimum linear discriminant functions in a two-dimensional plane of segment duration versus duration of the following phoneme.

Figure 1 shows distribution of durations for /j/, /i/, and /ij/ in the two-dimensional plane. The same technique was also applied for the separation of /w/, /u/, and /uw/. For the present speech materials, percentages of correct recognition of these phonemes after the first stage and the second stage are 96.8% and 94.4% respectively.

REFERENCES
IDENTIFICATION OF CONSONANTS BY A FINGER-TIP TACTUAL VOCODER

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INTRODUCTION

A finger-tip tactual vocoder composed of a vibrator array about the size of a matchbox was designed for the deaf. By using our tactual vocoder for daily training, the identification of 5 Japanese vowels was improved up to almost 100%. On the other hand, that of consonants was scarcely improved. In this report, based on the identification tests of various consonants, we investigated why the consonants could not be so much recognized as vowels.

APPARATUS AND EXPERIMENTS

In our tactual vocoder, 200-4400Hz components of voices were divided by a 16-channel filter bank based on the critical bands of the auditory sense and the output of each filter was transformed into a mechanical vibration of one of 16 bimorph ceramic vibrators with the 200Hz frequency. As shown in Fig.1, these 48 vibrators were arranged in 16 columns with 1mm pitch and in 3 rows with 3mm pitch. After a 20-30 minute training, the identification tests were carried out by randomly pronounced 5 Japanese consonants followed by a common vowel /u/. There were three groups composed of 5 consonants in these tests. The 1st group consisted of the consonants (/k/, /s/, /n/, /h/, /r/) that were difficult to be identified by lip reading. The 2nd was the explosive consonants (/k/, /g/, /t/, /d/, /p/) that had impulsive changes in sound pressure. And 3rd was the nasal and glide consonants (/m/, /n/, /r/, /y/, /w/). Furthermore, by storing these consonants in a memory and displaying them on a finger-tip slower than real time speed, identification scores were obtained as a function of speed ratio.

RESULTS AND CONCLUSIONS

From experimental results as shown in Fig.2, (1) identification score for the 1st group of consonants showed the highest percentage though they were difficult to be identified by lip reading, (2) that for the 2nd group of the explosive consonants, which had short and impulsive changes at the beginning, showed the worst percentage, (3) the identification of every group showed the maximum percentage when it was displayed 3-4 times slower in speed. These results make it clear that tactile sense is 3-4 times as bad as hearing sense in time resolution. Because of high redundancy of spoken language, the most of daily usages may be understood without perfect cognition of consonants. However, to improve our instrument, we are trying to find a suitable method for the cognition of consonants by an apparent extension of display time by scanning technique using 3 columns of vibrator array.

![Fig.1 Head of Vocoder](image1.jpg)

![Fig.2 Identification Scores.](image2.jpg)
TEMPORAL ORGANIZATION OF SEGMENTAL AND SUPRASEGMENTAL FEATURES

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INTRODUCTION

Temporal organization of speech calls for synchronism between segmental and suprasegmental features. The present paper describes methods and results of quantitative analysis of temporal relationships between these features in the realization of word accent of Japanese.

METHODS OF ACOUSTIC ANALYSIS

Two-mora words of the Osaka dialect, consisting only of vowels, were listed in random order and pronounced in isolation by two speakers of the Osaka dialect. The speech signal was recorded and analyzed by a digital computer.

Segmental features were extracted as formant frequency trajectories obtained from short-time frequency spectra, and the onset of formant transition was determined by the method of Analysis-by-Synthesis based on a functional model for the control process of articulation. Suprasegmental features were extracted as contour of voice fundamental frequency obtained from short-time autocorrelation functions, and the onset/offset of accentual component, characteristic to each word accent type, was determined also by Analysis-by-Synthesis based on a functional model for the control process of phonation.

RESULTS

Analysis of the speech materials revealed that phonatory control lags behind articulatory control by approximately 50-70 msec, as shown by the two examples of Fig. 1, which compares fundamental and formant frequency trajectories in two accent types of [ai]. Though somewhat dependent on speech rate, the lag was consistently observed in both speakers regardless of phonemic constituents of the test words. The existence of the lag was also confirmed at the physiological level, by observations of electromyographic activities in the genioglossus and the cricothyroid muscles, recorded simultaneously with the speech signal in one of the subjects.

REFERENCES


Fig. 1. Comparison of formant and fundamental frequency trajectories, together with estimated instants of onset of articulatory and phonatory controls in two accent types of [ai].
LA COARTICULATION ET SES EFFETS ACOUSTIQUES SUR LES CIBLES DES VOYELLES ORALES FRANÇAISES

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INTRODUCTION

L'analyse spectrale des voyelles orales du Français a été effectuée sur divers échantillons de parole: voyelles isolées, syllabes CV, syllabes VC, syllabes CVC, dictionnaire de 700 mots lus par 5 locuteurs, et corpus de 50 conversations de 1/2 heure enregistrées par Malécot (1).
Cette étude montre les effets du contexte phonétique sur les formants mesurés sur un interval "quasi-stable" des voyelles.

METHODE D'ANALYSE

L'étude de la parole naturelle implique l'analyse d'un corpus de larges dimensions à cause de la grande variabilité des paramètres acoustiques. Les dimensions de cette tâche nécessitent l'emploi de l'ordinateur pour la localisation des voyelles, l'analyse spectrale, l'extraction des formants et l'analyse statistique des mesures.

Des algorithmes développés pour la reconnaissance automatique de la parole et des programmes standards d'analyse statistique ont été utilisés pour cette étude (2). En particulier, le signal a été échantillonné, quantifié, et mémorisé sur disques et bandes digitales. Les noyaux syllabiques ont été définis aux maxima d'une courbe lissée suivant l'énergie du signal. La méthode globale de prédiction linéaire, dite d'autocorrélation, a fourni une estimation optimale du spectre des portions stables du signal. Les pics de ces spectres correspondent aux formants caractéristiques du timbre des voyelles. Les fichiers contenant ces mesures pour 35 000 voyelles sont accessibles par des programmes statistiques standards tels BMDP (3).

DISCUSSION, RESULTATS ET CONCLUSIONS

Pour des raisons d'économie, on peut postuler que les locuteurs ont pour chaque voyelle une cible articulatoire à laquelle correspond une cible acoustique. Cette cible n'est pas toujours atteinte. La variabilité des formes spectrales et en particulier celle des formants des voyelles croît avec la complexité du type de parole analysé. En parole naturelle, l'accentuation est un facteur important (4). En position innaccentué, toutes les voyelles se rapprochent du timbre du e-muet.

Le contexte et plus particulièrement les sons immédiatement contigus influencent les valeurs des formants mesurés sur les portions stables des voyelles. On peut prédire qualitativement et quantitativement les effets de la coarticulation sur les voyelles. Un tel modèle pourrait être intégré, à profit, dans un système de synthèse ou de reconnaissance de la parole. Les contraintes imposées par le contexte influencent l'évolution des systèmes phonologiques et devraient figurer dans les modèles diachroniques.

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(2)G.F.Chollet, AAP Convention, San Diego, (Nov. 1976)
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(4)G.F.Chollet, JASA,(Fall 1976),50,Suppl.No.1,S44(Ab.)
This paper is concerned with a program for an automatic segmentation of continuous speech into words in French. The program uses the linguistic information contained in the fundamental frequency (Fo) variations along the sentences. The algorithm for the segmentation is based on the author's previous work on French prosody (1) and (2). We shall make the following assumption: a sentence is decomposed into breath group by pauses; a breath group is further decomposed into prosodic words by Fo fluctuations. A prosodic word corresponds to a single lexical word, or to more lexical words, which are closely semantically related. Figure 1 illustrates one example of a breath group decomposed into four prosodic words.

The goal of the program is to detect the locations of the large Fo rises and falls inside the breath groups. The threshold level for deciding whether a Fo movement has to be considered as relevant is speaker dependant. The program calculates the rate of the Fo changes for eliminating the very rapid ones as errors of the detector or due to the production of a consonant.

In an experiment, 4 speakers read 13 sentences, containing 67 lexical words. There are 268 lexical words in the material tested. The program detected 465 Fo movements as relevant, and categorises them as 'R' (rise) or 'F' (fall) - see Fig.1. Our analysis indicates that (1) the beginnings of 160 words (60.7%) and the ends of 196 words (73.2%) are marked by 'R' and 'F', respectively; (2) 58 rises marked by 'R' correspond to a continuation rise at the end of a word; (3) 9 Fo falls marked by 'F' represent a Fo fall along the words, and do not correspond to a specific offset of a lexical word; (4) 42 Fo movements marked either by 'R' or 'F' are due to underlying consonants.

It is interesting to note that 60% of the onsets and 73% of the offsets for the 268 words are detected correctly, by simply assuming that 'R' corresponds to the onset of a word, and 'F' to the offset. However, the performance may be improved considerably by implementing an algorithm to separate the rises corresponding to the onset of a word from the rises corresponding to a continuation rise, and further by combining a phonem detector which allows the program to delete the marks assigned on consonants.

In Ref. 1 and Ref. 2
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(2) VAISSIERE J. Further Note on French Prosody, PR 115, Massachusetts Institute of Technology, Research Laboratory of Electronics, 1975, pp 251-261.
ETUDE DES CARACTERISTIQUES ACOUSTIQUES INTRINSEQUES DES VOYELLES FRANCAISES

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1 - Modèle de simulation - Paramètres de commande

On a montré que l'on pouvait représenter l'impédance d'entrée par un ensemble de circuits accordés sur les fréquences formantiques, les autres caractéristiques peuvent être déduites de la fonction de transfert du conduit vocal (1). La source vocale est simulée par un modèle à deux masses (2). Une première étude a été nécessaire pour déterminer l'échelle des paramètres de commande de la source afin d'obtenir un signal glottique de forme et de fréquence en accord avec les mesures réelles. Pour une voix d'homme, dans la plage de fréquence 80 - 160 Hz, les valeurs retenues pour les paramètres de la source vocale sont les suivantes : K1 = 20 à 60 kdynes/cm² pour une pression de 2 à 12 cm H2O.

2 - Résultats

2.1. Fréquences intrinsèques

On sait que les voyelles naturelles "fermées" ont une F0 intrinsèque plus élevée que les voyelles "ouvertes". Ceci peut être expliqué par plusieurs hypothèses. Les résultats obtenus pour les voyelles du français sont donnés (fig. 1). Les variations de fréquence sont assez faibles mais dans un sens opposé à celui constaté dans la parole naturelle. Le couplage ne permet donc pas de rendre compte à lui seul des phénomènes observés à l'analyse. On remarque que la fréquence intrinsèque est liée directement à la fréquence du premier formant (fig. 2). Ceci peut être expliqué en considérant la partie imaginaire de l'impédance d'entrée du conduit vocal.

2.2. Puissance intrinsèque de l'onde de débit

Par simulation, nous avons également évalué les valeurs de la puissance intrinsèque de l'onde de débit. Les résultats sont donnés dans le tableau 1. Dans la parole naturelle, ce sont les voyelles les plus ouvertes qui sont les plus intenses. Ce résultat n'apparaît pas clairement dans les valeurs de l'onde de débit, ce qui implique que la fonction de transfert du conduit vocal joue un rôle pour cette caractéristique.

Tableau 1 : Puissance en valeur arbitraire de l'onde de débit :

<table>
<thead>
<tr>
<th>Voyelle</th>
<th>(u)</th>
<th>(a)</th>
<th>(i)</th>
<th>(o)</th>
<th>(e)</th>
<th>(ê)</th>
<th>(y)</th>
</tr>
</thead>
<tbody>
<tr>
<td>F0</td>
<td>31</td>
<td>30,75</td>
<td>30,47</td>
<td>31,7</td>
<td>30,5</td>
<td>31,2</td>
<td>31,7</td>
</tr>
</tbody>
</table>

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Nous présentons une unité à réponse vocale en temps réel utilisant un mini-ordinateur. L'onde sonore est calculée directement dans sa représentation amplitude-temps. Un programme engendre pour chaque phonème les caractéristiques acoustiques en fonction des phonèmes précédents et suivants. La transcription phonétique, le rythme et la mélodie sont assurés par programme.

TRANSCRIPTION PHONETIQUE, RYTHME ET MELODIE

Le programme de transcription applique les règles usuelles de la prononciation du français et utilise un dictionnaire de taille très réduite pour les cas d'exception.

Un découpage automatique de la phrase en groupes rythmiques permet de calculer le schéma intonatif et le rythme sans analyse syntaxique (2).

GENERATION DES PARAMETRES ACOUSTIQUES

Les trajets formantiques en fréquence et en amplitude sont déterminés par règles dans toutes les zones voisées ou bruitées en fonction des consonnes qui entourent les voyelles. Ces paramètres sont recalculés pour chaque période ou chaque zone élémentaire bruitée.

CALCUL DE L'ONDE SONORE

Chaque période est calculée par microprogramme comme la somme de plusieurs courbes représentant les formants et les bruits de friction ou d'explosion (1). Le signal est obtenu au moyen d'un simple convertisseur digital-analogique.

RESULTAT

Toute phrase écrite est transformée en une parole continue parfaitement intelligible et d'un timbré très naturel.

REFERENCES

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(2) X. RODET, C. SANTAMARINA, Journées d'Etude sur la Parole du GALF (1975) 1, 364
Le travail présente les résultats de la première phase d'une recherche qui se propose comme but final d'établir les rapports existantes entre l'intensité, le spectre et la perceptibilité des voyelles et des consonnes de la langue roumaine — phase où l'on effectue une interprétation statistique des données de la sphère de l'intensité globale des éléments respectifs de la parole.

Le matériel de référence comprend approximativement 600 mots, ainsi choisis à fin de pouvoir épuiser — autant que possible — toutes les combinaisons positionnelles admises dans la langue roumaine, considérant chaque voyelle et consonne comme élément initial, moyen et final des mots, dans des syllabes accentuées ou non accentuées.

Ainsi, — en mettant des sujets — (féminins et masculins, professionnels de la voix, parleurs usuels) — à prononcer les mots, constitués en phrases —, on a obtenu un total d'environ 14.000 éléments phoniques, qui ont été analysés à l'aide du "phonomètre" Bruel & Kjaer (2203) et, en même temps, inscrits sur "L'enregistreur de niveau" B.& K. (2.305).

À la fin de l'étude se trouve un tableau qui présente les rangs occupés par les éléments discrets de la langue roumaine par rapport à la fréquence de leur apparition ainsi qu'à leur intensité globale.
In automatic word recognition the parameters that describe a word have to be speaker-independent; whereas speaker recognition needs text-independent parameters. The present work gives the possibility to use the same parameters and comparable recognition methods for word recognition as well as speaker recognition.

In word recognition, experience has shown the difficulties of making the correct segmentation. This problem can be avoided if the parameters describe a complete word. In our research the two-dimensional (modified) cosine-coefficients $\gamma_{n,k}$ of the short-time spectrogram of a whole word are considered:

$$
\Gamma_{n,k} = \sum_{i=0}^{L-1} \text{sgn}(s(i)) \cdot s(i+n) \cos\left(\frac{\pi ki}{L}\right)
$$

with $n = 0, \ldots, 9$ and $k = 0, \ldots, 3$ (1)

where $s(i)$, $i = 0, \ldots, L-1$ is the (sampled) speech signal. The parameters used are: $\gamma_{n,k} = \Gamma_{n,k}/\Theta_{0,0}$, which are independent of the intensity. The coefficients with low indices have the considerable advantage of being relatively independent of the rate of speech production, of the fundamental frequency and also of the precise values of the formant frequencies.

For the recognition, the unknown parameter-vector $(X)$ is compared with sample-vectors $(G)$ by considering distances $(D)$:

$$
D = ((X - G) W^{-1} (X - G))
$$

where $W$ is a covariance-matrix, dependent on the various recognition methods. With Eq. (2), word as well as speaker recognition are realized. While covariance matrices that are both word and speaker dependent would theoretically give the highest recognition scores, they are apt to be singular or near-singular, thus obviating their theoretical advantage. We have obtained the best recognition scores with matrices that are linear combinations of covariance matrices that are either word or speaker dependent with covariance matrices that are both word and speaker dependent.

The ten numerals 0 to 9 of the German language are used, spoken by 21 male speakers. From information gained from many speakers, the average characteristic parameter-vector for each word is obtained. With 20 speakers, 93% of the spoken words are recognized correctly. If word recognition is made in a speaker-dependent way and is preceded by correct speaker recognition, the word recognition rate exceeds 99% for each of the 21 speakers.

Speaker recognition is realized with single words as well as combinations of words ("phrases"). The use of phrases yields better results than the use of single words: with one phrase (02678), spoken by 21 male speakers, all speakers are recognized correctly; with six other phrases the rate is 98%. Of the single words used, only one word ("eins") reached 90%. All words or phrases are assumed to be known in the speaker recognition process.
INTRODUCTION

A model for specifying nasality obtained on the basis of perceptual experiments showed quite reasonable results on Japanese speech data (Takeuchi 1975, unpublished) under the following assumptions: (a) sonorants are discriminated from obstruents; (b) the only liquid can be separated from nasals by duration; (c) relatively homogeneous speakers, such as male adults, are used. This paper describes an improved procedure for automatic detection of nasality from continuous speech which makes no such preliminary assumptions.

METHODS AND RESULTS

Input speech signals are analyzed using the linear prediction autocorrelation method to compute the RMS energy, reflection coefficients, and normalized residual signal energy (ERS). The back-to-total cavity ratio (BTR) and the front-to-back maximum area ratio (FBR) of the area function, voice bar parameter (VB), and smoothed frequency envelopes are also computed. The frequency for the smoothed envelopes is rescaled according to the estimated average vocal tract length to eliminate inter-speaker differences.

First, the center of a syllable nucleus is detected by use of the RMS energy, duration between RMS extremes, the BTR and FBR parameters, and voiced/unvoiced detector. Then, the intersyllabic segment bounded by two successive syllable centers is analyzed by taking the syllable structure of English into account. At the first stage of the analysis, the obstruent sound segments are excluded from the intersyllabic segment by applying the turbulence noise and voice bar detector. At the second stage, the first three spectral peaks, ERS, BTR, and spectral difference measure between the input smoothed envelope and the best matched vowel spectrum are computed. The nasal segment is assigned to those sound segments whose non-Euclidean distance to the reference point in the parameter space is less than a threshold. This decision process is followed by a smoothing process which takes the segment duration into account.

The procedure has been tested on ten sentences spoken by each of eight speakers (four males and four females) and found to be quite effective.

REFERENCES

INTRODUCTION

For the discrimination of different speech-communication systems it is necessary to run practical tests with subjective measurements of the 'quality' of the transmitted signal. In systems with a relative high-quality performance, where intelligibility advances, the effort for intelligibility tests used for discrimination of systems also increases. In this case another method of quality measurement is necessary to reduce time consuming tests. A method of preference judgement seems to be the most relevant to establish a rank order among different systems. In order to come to an absolute quality value for any communication system it is necessary to establish a common reference system for comparison purposes, including different types and levels of distortion.

MEASUREMENT PROCEDURE

A group of seven reference systems with decreasing quality levels similar to (1) was developed by computer simulation. These reference systems contain different types of distortions: A: Original signal, B: Band-pass, C: Additive noise, D: Echo, E: Dropouts, F: Peak-clipping, G: Rectified signal.

A short list of words, representing the most used german phonemes, spoken by a male speaker, served as testing material. Presenting all possible pairs of systems to a group of listeners and counting the preference judgements the rank order was established and the quality level was defined with 100% for the original signal; decreasing in 10%-steps of quality.

Any test-signal has to be presented with all these reference systems for paired comparison, based on the preference judgements an absolute 'value of quality' can be given related to these reference systems.

RESULTS

Several signals with different distortion levels were tested and quality labeled:

1) PCM-signal
   - 32Kb/sec: Quality level: 66%
   - 40Kb/sec: 80%

2) ADPCM-signal
   - 24Kb/sec: 70%
   - 32Kb/sec: 84%

3) KLT-signal
   - (Karhunen-Loève-Transformation (2))
     - 3.2Kb/sec: 57%
     - 6.4Kb/sec: 69%

CONCLUSIONS

The experimental results show that it is possible to state absolute quality values for different speech signals with reasonable testing efforts. The results are reproducible with different groups of listeners.

Because of the multidimensionality of quality judgements it is impossible to reach all dimensions of a complex signal distortion with such a single over-all decision, and improvement of the measurement method has to be subject of further investigations to come toward a short realistic testing procedure.

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ACOUSTICAL CORRELATES OF THE EXPRESSIVE STRESS IN SENSE-DISTINCTIVE INVERSION

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INTRODUCTION

Phrases frequently occurring in the colloquial speech where a given word is emphasized through suprasegmental characteristics instead of inserting an explanatory word like "just", "namely", etc., are treated in the paper with the purpose of defining the acoustical features through which these structures are realized.

THEORETICAL BASES

The experimental results are obtained on the basis of the acoustical comparative analyses of the fundamental frequency and the durations of those parts of the phrases which bear the expressive stress.

EXPERIMENTAL TECHNIQUES

In discussing that kind of phrases the emphasizing of one or another word is connected with the change of the word-order. To investigate these changes are analyzed phrases in which different parts of speech bear the expressive stress in the corresponding inversion. The linguistic material includes the following types of syntactical structures: predicate-subject, subject-object, predicate-adverb and two identical parts.

The recorded material consisting of 90 phrases is analyzed with pitchmeter. Three persons are employed as speakers. The data is processed by a statistical analyses.

RESULTS AND CONCLUSIONS

The results show equal difference in the fundamental frequency value for one and the same sound in either type of word-order. The average statistical value of this difference is 95 Hz with a small variation interval (10 Hz). The duration of the sound-bearer of the expressive stress is bigger by 0.3 cs.

The expressive stress frequency regarding the average frequency value of the whole phrase is approximately by 90 Hz higher in both phrase structures.
AN AUTOMATIC SPEECH RECOGNITION SYSTEM USING SYLLABLE NUCLEI AND CONSONANT CLUSTERS

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INTRODUCTION

In contrast to classical phonem recognition this paper describes an automatic speech recognition system which proceeds from the syllable as the speech unit and thereby avoids the big problems which accompany the localizing of single phonems in the speech signal. The number of syllables in a language is in fact limited, but very large. Accordingly the syllable unit is here divided into the syllable nucleus, which would correspond to a vowel or a vowel cluster in German, and generally into two consonant clusters whereby one precedes and the other follows the nucleus. Of course, in the physical pattern space the consonant clusters always contain a certain part of the adjacent vowel as well, see fig. 1. The thus defined units offer the advantage that, for example in German only about 50 initial consonant clusters (ICC) are formed, as well as 250 final consonant clusters (FCC) - if all flexions are taken into account - or only 50 FCC if the flexions remain unconsidered. In the case of the syllable nuclei there are about 15 vowels (including diphontongues) to be discriminated. This inventory already allows the identification of all monosyllabic words. As far as multiple-syllabic words are concerned an attempt is being made to separate the medial consonant clusters at a suitable position so that only a sequence of an FCC and an ICC is produced again. These units, which have already been described in previous papers /1, 2/ permit the representation of a practically unlimited vocabulary.

RECOGNITION PROCEDURE

A special loudness analyzer is used in order to generate 22 specific loudness functions covering a frequency range of 30 to 10 kHz, whose sum yields the so-called loudness level $L$. Additionally, a modified and smoothed loudness function $L_m$ is calculated as a weighted sum of all 22 channels, see fig. 1b. The peaks of $L_m$ mark the positions of syllable nuclei, whereas the $L_m$ minima between two consecutive nuclei indicate suitable positions for the separation points $S$. So, for example, when the word "cluster" (/klaster/) is considered, as in fig. 1b, the $L_m$ minimum at $S_2$ causes the following segmentation /kl/ /a/ /s/ /t/ /a/-/ and the minimum at $S_3$ causes /kl/ /a/-/ /st/ /a/-/. The notation /-/ indicates an "empty" consonant cluster, that is, the vowel is in a final position. All resulting segmentations are then processed further.

Recognition starts with the classification on the one hand of syllable nuclei and on the other hand of consonant clusters. For consonant clusters a special time normalization is carried out for this purpose. This allows the comparison of different length patterns and therewith the application of known methods of pattern recognition. Finally, every resulting syllable is checked against a list of possible syllables and every resulting sequence of syllables is checked with a lexicon entry of a predefined vocabulary for the best correspondence. The described system has been realized for the recognition of spoken German words, but the extension of the system to clearly spoken connected speech seems possible.

REFERENCES

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INTRODUCTION

In 1971, the Advanced Research Projects Agency (ARPA) of the United States government initiated a five year research effort to develop a speech recognition system that would meet, among others, the following specifications: work on connected speech with a 1000 word vocabulary, for multiple cooperative speakers, running in near real-time on a futuristic machine that executes 100 million instructions per second, and achieve at least 90% semantic accuracy. The Harpy system is the first system to meet the above specifications.

SYSTEM STRUCTURE

The Harpy system is an extension of two earlier systems developed at Carnegie-Mellon University, Dragon and Hearsay-1. The knowledge of the system which includes the language grammar, lexicon (dictionary of phonetic spellings of the words in the language), word juncture rules, and prosodic rules is combined into an integral state network. Each state in this network contains a lexical name (word in the language), a phoneme name, expected phoneme durations, a list of prior states, and a list of following states. All paths through the network (from the initial state to the final state) represent all pronunciations of all legal sentences in the language. A set of user-dependent phonetic templates are used for calculating matching probabilities (acoustic match probabilities) of the phonemes to the speech signal.

The recognition process consists of a heuristic graph searching technique that finds an optimum path through the network. Plausible states along the search paths are matched with the speech signal based on the acoustic match probabilities (and the phoneme durations). Only a few of the "best" paths through the network are searched where "best" is defined by a heuristic threshold on the state probabilities.

TASK AND TRAINING DATA

The task domain for the system is an AI information retrieval task which contains 1011 unique words or 815 base words (excluding words with endings such as ..ing, ..s, ..tion, etc.). Two example sentences are "List the abstracts by Newell or Simon" and "Do any papers discuss planner-like languages". The language grammar has a static branching factor of 10. The static branching factor is the average number of word choices for all partial sentences specified by the grammar. The average dynamic fanout is 35. This is the average number of word choices encountered during the recognition process for all the test data. The system was trained on five speakers, three male and two female. The training set for one of the male speakers contained 476 sentences (3568 words). This set was used to generate the phonetic spellings for the lexicon and to generate a set of speaker-dependent phonetic templates. The training set used by each of the other four speakers contains 256 sentences (1444 words). These data were used to generate a speaker-dependent set of phonetic templates for each of these four speakers.

TEST DATA AND PERFORMANCE

The combined test data for all five speakers contain 184 sentences (1138 words). The errors that the system made on these data were never analyzed nor corrected; the results on the test data represents a true indication of the performance of the system. The system had a 3% word error rate and a 9% sentence error rate on the test data. The semantic error rate (the sentence error rate minus trivial errors such as incorrect tense, etc.) was 5%. The speed of the system is 31 MIPSS (Million Instructions Per Second of Speech). If a machine exists that can execute 31 million instructions per second, then Harpy would run in real-time in this 1000 word AI task. A more complete description of the system is available in Lowerre [1].

REFERENCES

THE USE OF SYNTACTIC AND SEMANTIC INFORMATION IN A SPEECH UNDERSTANDING SYSTEM

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INTRODUCTION

During the past five years a great amount of work has been devoted to the design of automatic connected speech understanding systems. In such systems it is not necessary to recognize every phoneme of a sentence as far as the meaning of the sentence has been obviously understood. This paper is devoted to the presentation of the syntactic and semantic processing levels of the MYRTILLES speech understanding system which is presently under development in our laboratory.

SOURCES OF KNOWLEDGE AND BASIC PRINCIPLES

The system operates on multiple-answer, error-full strings of phonemes produced by the acoustic-phonetic component. The strategy relies mainly on a grammar considered as a parameter of the system. In addition it is possible to accept some syntactic errors in the input sentence, as mentioned below. The use of semantics and pragmatics makes it possible to avoid some errors in the parsing and to accept only meaningful sentences. Finally prosodic features (concerning pitch, intensity and rhythm) can be used as additional information at both syntactic and semantic levels.

Our system uses these various sources of knowledge with an hypothesis-and-test principle: the parser builds a set of hypothetical words which may appear in the sentence, according to the grammar, the context and other available informations. These hypotheses are then verified or cancelled by a lexical analyzer which uses phonological alternative descriptions of the vocabulary in order to spot a given word in the phoneme string. The parser then starts again with the most likely hypothesis. This algorithm is non-deterministic in several ways. Various heuristics are used in order to decrease the undeterminism.

A PRACTICAL IMPLEMENTATION

In order to improve some of these basic ideas a practical system has been implemented for the automatization of a telephone exchange. This preliminary version uses a top-down, left-to-right parser together with a thirty-five-rule grammar and a forty-word vocabulary. It also incorporates a dialog procedure in order to solve some ambiguities before acting the command asked by the user. Such a procedure is very general and is to be found in any speech understanding system.

FURTHER WORK AND CONCLUSIONS

Work is presently in progress for designing a more general and powerful system. An original approach has been stated for the incorporation of semantic information in the syntactic processing. The use of prosody at all levels of processing: acoustic-phonetic, syntactic and semantic is also investigated. In addition a new parsing algorithm is used concurrently with the classic left-to-right algorithm. The work herein reported, as well as other similar works in the world, have shown the feasibility of connected speech understanding system. However, one is still far from the industrial development of such systems.
Several systems for recognizing words have been implemented in our laboratory. These systems differ in vocabulary size (from 2 to 54 words), and in whether the words are spoken in isolation, or in a connected sequence. A common element of all these systems is a statistical framework for the recognition algorithm in which a statistical characterization of the features for each word in the vocabulary is stored. The features used for recognition in each of the systems are the set of LPC coefficients for each frame. Novel techniques for signal preprocessing and time warping are used to combine repetitions of the vocabulary words by both the same and different speakers. Also, an expanded form of the principle of minimal residual error is used in a new distance metric which incorporates the effects of both prediction residual, and inter-replication variation of the LPC parameters.

The standard LPC distance metric proposed by Itakura[1] is of the form

\[ d(a, \hat{a}) = \ln \left( \frac{aV\hat{a}'}{\hat{a}V\hat{a}'} \right) \]  

where \( a \) and \( \hat{a} \) are sets of LPC coefficients, \( d(a, \hat{a}) \) is the distance between the coefficient sets, and \( V \) is the correlation function of the speech whose LPC set is \( \hat{a} \). The new distance metric is of the form[2]

\[ d' = (\hat{a} - m)(S + V^{-1}(N\hat{a}\hat{a}'))^{-1}(\hat{a} - m)' \]  

where \( S \) is the covariance matrix of the LPC coefficients, and \( m \) is the mean vector of LPC coefficients. Both \( S \) and \( m \) are obtained by averaging LPC sets over repetitions of the same frame by the same or different speakers, and in different acoustic environments (e.g. coarticulation effects in speaking).

The recognition systems incorporate a wide variety of signal processing algorithms. The major components of the systems are:

1. Endpoint Detector
2. Voiced-Unvoiced-Silence Analysis
3. Segmentation Procedures
4. LPC Analysis
5. Time Alignment Procedure
6. Distance Calculation

Each of these components has been extensively studied and tested on a wide variety of speech utterances.

Using these analysis methods, three distinct recognition systems have been studied. These include a simple yes-no (binary) recognizer, a connected digit recognizer; and a computer vocabulary recognizer. Experimental evaluations of these systems have yielded recognition accuracies of from 95 to 100% across a variety of speakers, and for different transmission conditions including a computer room environment and a standard telephone connection.

References


INTRODUCTION

There are two ways in which splitting the information carried by the speech wave between the ears might help speech reception by the sensorineural deaf: (a) apply F1 to one ear and the higher formants to the other, and (b) apply F1 and F3 to one ear and F2 and F4 to the other. The first configuration (a) is designed to prevent the relatively intense F1 from masking the higher formants (1). It has been found by Rand (2) that in this configuration with normal listeners the intensity of the higher formants can be reduced by about 20 dB compared with binaural presentation before speech intelligibility is impaired. The second configuration (b) is designed to increase the frequency separation of the formants in each ear in order to improve speech reception by those with poor frequency resolving power.

EXPERIMENTAL TECHNIQUE

Before designing elaborate devices to separate the formants of speech it is desirable to know if either of these configurations has any deleterious effects on the perception of any speech sounds by listeners with normal hearing. To this end a series of experiments have been conducted using speech syllables synthesised by rule (3).

RESULTS

With configuration (a) it was found that perception of initial stop consonants and semivowels was essentially the same as with normal binaural presentation. With configuration (b), however, recognition of stop consonants was reduced by about 6% and recognition of semivowels and fricatives by about 4%. Recognition of vowels, either in isolation or in h-d words, was not significantly different from normal binaural presentation.

REFERENCES

RECOGNITION OF SPOKEN WORDS BY USE OF SPECTRAL PEAKS AND ITS EVALUATION

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INTRODUCTION

According to the investigations and the considerations on the processing of sound by auditory organ, we are led to a confirmation that spectral peaks are important features to express phonemes. And the spoken words recognition system by use of spectral peaks as acoustic parameters are developed. And the usefulness of the spectral peaks are proved by the experiment with the recognition system.

EXPERIMENT ON THE AUTOMATIC RECOGNITION OF SPOKEN WORDS

Input speech is frequency-analyzed by a filter bank composed of 29 single peak filters of low selectivity. Three major spectral peaks (P1, P2 and P3) bearing phonemic information and total power (W) are picked out every 10 ms. Fig. 1 shows how the input speech is processed. Tab. 1 shows the results of recognition experiment. Next, 25 different male adults uttered 51 city names. The recognition score was found to be 94%, when the speakers were permitted to repeat their utterance for three times.

RECOGNITION BY AID OF A MAN

In this experiment, one of the authors selected the most similar lexical item to an input speech, referring to the string of phoneme groups transformed from the input speech and the intermediate outputs of sequential matching. 60 samples randomly sampled from 166 city names were completely recognized which were uttered by one of the speakers for the experiments of Tab. 1.

CONCLUSION

The usefulness of the spectral peaks for the recognition of spoken words has been proved by the experiment with the recognition system by use of spectral peaks as acoustic parameters. Furthermore, from the results of above-described experiment, it is considered that the recognition score by the system will become nearly 100% when the effective use of linguistic information is completed.
INTRODUCTION

One approach to the automatic recognition of speech (for application to language identification, word spotting, etc.) avoids the problem of complete phonetic, or phonemic, identification by searching for gross acoustic elements selected to represent linguistic categories. In other words, rather than attempting to identify, for example, /p/, /t/ and /k/, the procedure tries to identify the category voiceless stop consonant. With such a procedure, the efficiency of identification increases as the target categories become grosser, but with a concomitant increase in word confusion as the number of different strings in the language becomes fewer. The number of false alarms — that is, identifications of the wrong word — is a function of a given language, as well as the number and type of categories used.

METHOD

A pronouncing dictionary [1] was used to convert a 1-million word text of American English [2] into strings of phonemes. Subsequently, these strings were converted into variously defined categories and the occurrence of such category strings for various target words was measured.

Initially, the targets were restricted to lie within word boundaries. All words in the text made up of four, seven and 10 elements were used as templates and the occurrence of strings corresponding to them was determined. The resulting statistics describe the frequency of occurrence of (1) variously defined strings of elements of these three lengths, and (2) of false alarms generated by these templates. Finally, estimates of the false-alarm rate across word boundaries were made.

Strings used the following degrees of categorization: (a) four categories (vowel, stop consonant, fricative consonant, nonvocalic sonorant); (b) six categories (vowels, voiceless stop, voiced stop, voiceless fricative, voiced fricative, nonvocalic sonorant); (c) eight categories (close vowel, mid vowel, open vowel, voiceless stop, voiced stop, voiceless fricative, voiced fricative, nonvocalic sonorant); (d) ten categories (close vowel, mid vowel, open vowel, voiceless stop, voiced stop, voiceless fricative, voiced fricative, nasal consonant, liquid consonant, glide consonant); and (d) 51 categories ('phonemes' of American English).

The findings are discussed in terms of application to the problem of automatic word identification.

REFERENCES


INTRODUCTION

The present experiments make inferences about the processes underlying the recognition of isolated natural speech sounds.

THEORETICAL BASES

It is hypothesized that at the first stage of the phonemic recognition process the encoding by categorical perception (1) takes place and that at the second stage the differences between phonemic categories do facilitate comparisons between probe and memory sounds.

EXPERIMENTAL TECHNIQUES

A technique developed by Stenberg (2) is used. Subject is auditory presented with a series of speech sounds which he tries to memorize, followed by a single probe speech sound. Subject must decide whether the probe item was one of the memory items.

RESULTS AND CONCLUSIONS

The mean reaction time (RT) of the responses as a function of memory set size is presented in Fig.1. Three different phonemic categories (stop consonants, nasals, and fricatives) were considered. Irrespective of the memory set size the RTs for nasals and stops are flat, whereas for fricatives there is a definite positive slope. The intercept of the RT function is the largest for nasals and the smallest for fricatives.

The pattern of results in Fig.1 suggests that category discrimination in the case of nasals and stops occurs at the comparison stage. Apparently, these phonemic categories are highly discriminable along an acoustic continuum. On the contrary, fricatives are relatively easily encoded but distinctive features of their phonemic category are not strong enough to enable a fast decision at the comparison stage.

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INTRODUCTION

There is some controversy in the literature about the relative contributions of various acoustical manifestations -temporal cues, excitation source contributions, and articulatory characteristics,- for speaker discrimination (1). An approach to solve this controversy was taken by Miller (2), who synthesized speech signals as if it were physiologically possible to interchange the larynges and vocal tracts of different speakers. Due to computer limitations, Miller's samples -the word /had/ and the sustained vowel /a/- and speaker population had to be severely limited. The data did not include temporal cues or natural sounding stimuli.

This study was an attempt to replicate Miller's approach but with a more sophisticated computer processing technique and more normal speech.

EXPERIMENTAL TECHNIQUES

Ten speakers (five male, five female) were asked to voice shadow a recording of a trained chorus speaking the "Rainbow Passage". The chorus was used so that all speakers would be prosodically synched to a single example that would be easy to shadow. This technique alleviated the necessity of computer warping the utterances so that interchanges could take place between a normally fast talker and a relatively slower one. The second and third sentences in the passage were processed using a Linear Prediction Coding (LPC) scheme that allowed the separation of the vocal tract influence, the LPC coefficients, and the source influence, the LPC residual, for each speaker. While it is true that the residual contains some vocal tract information and some error due to the inaccuracy of the LPC model, care was taken to keep these errors to a minimum.

The individual vocal tract models and source models were interchanged between the various speakers in all possible combinations. These hybrid combinations were presented to listeners in a paired forced choice paradigm i.e. are these speakers the same.

RESULTS AND CONCLUSIONS

When the interchanges were within the same sex, the results tended to show the vocal tract having the most influence. When across sex the source showed the most influence. These results seem to indicate fundamental frequency difference is used as a gross approximation, then formant structure is used as a finer determinate.

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Nous considérons une machine reconnaissant la parole comme un transformateur de débits d'information (bit/sec). Dans chaque seconde, l'onde téléphonique transporte 50,000 bits acoustiques, mais le cerveau n'en accepte que 50 bits phonétiques ou 10 bits linguistiques. Comment optimiser cette transformation 1000:1 ou 5000:1 ? Nous proposons l'analyse des redondances de systèmes symboliques comme fil directeur à travers le labyrinthe de la reconnaissance de parole (1). Selon les formules (1) à (4) du tableau ci-dessous, nous distinguons les débits d'information spectraux et symboliques (bit/sec), l'entropie négative (bit/symb) le rythme d'acceptance (symb/sec) et les redondances (bit/bit) qui caractérisent les lois pertinentes, les contraintes ou sources de connaissance.

a) Débit d'information spectral $D_d$ d'un canal (largeur de bande $L$ (Hz)), qui transmet des degrés d'amplitude, échantillonnés à la fréquence $F_d$ (Hz) :

\[ D_d = B_d(F_d) \]  

où $B_d = \log_2 d$ (bit), ou dynamique (dB)

b) Débit d'information symbolique $D_s$ d'un canal transmettant des signaux symboliques équiprobables ($p_4=0,5$) à la cadence $F_s$ (symb/sec), avec $B_s=\log_2 s$ (bit/symb) :

\[ D_s = B_s(F_s) \]

Si les symboles ne sont pas équiprobables, c'est-à-dire avec $p_4 \neq 0,5$ :

\[ B'_s = -\sum p_i \log_2 p_i \]

entropie négative moyenne

c) Redondance $R_{xy}$, quand on passe d'un débit d'information $D_x = B_x(F_x)$ (bit/sec) à un autre débit $D_y = B_y(F_y)$ (bit/sec), avec $F_x = F_y$ et $B_x \geq B_y$ :

\[ R_{xy} = 1 - \frac{B_y}{B_x} \]

rapport des entropies négatives

En appliquant l'analyse des redondances à la langue française avec un dictionnaire de 25,000 mots, on obtient le tableau des correspondances suivantes :

<table>
<thead>
<tr>
<th>domaine de contrainte</th>
<th>prosodique</th>
<th>phonétique</th>
<th>occurrence</th>
<th>lexicale</th>
<th>gramma-tique</th>
<th>sémantique</th>
<th>contextuelle</th>
<th>linguistique totale</th>
</tr>
</thead>
<tbody>
<tr>
<td>rapport entropie $B_y/B_x$</td>
<td>5/5000</td>
<td>5/5000</td>
<td>4,2/5</td>
<td>3/4,2</td>
<td>1,5/3</td>
<td>1,2/1,8</td>
<td>1/1,2</td>
<td>1,2/1000</td>
</tr>
<tr>
<td>redondance $R_{xy}$</td>
<td>0,999</td>
<td>0,999</td>
<td>0,16</td>
<td>0,286</td>
<td>0,5</td>
<td>0,2</td>
<td>0,167</td>
<td>0,9998</td>
</tr>
</tbody>
</table>

On constate que les contraintes de la prosodie et celles de la phonétique sont d'importances égales (0,999), mais que les autres contraintes linguistiques sont d'importance moindre (1=0,16,0,286,0,5,0,12,0,001=1=0,2,0,8). La contrainte linguistique totale donne $1 = (0,001.0,2 + 0,001) = 1 - 0,0012 = 0,9998$.

L'analyse des redondances peut détailler les composantes prosodiques, telles que dynamique, durée, pitch ou degré de voixement. Elle est applicable généralement aux formes diverses de langages naturels ou codés (phonocodes), avec des dictionnaires et des règles grammaticales plus ou moins étendues. Les analyses ainsi établies servent de fil directeur pour optimiser les performances des machines reconnaissant la parole.

**REFERENCE**


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INTRODUCTION

Traditionally data for planning telephone networks has been obtained from tests using human participants. Such tests are laborious and expensive and do not permit of rapid assessment of any proposed network change, nor are they suitable for network optimization. Progress in modelling of human telephonic behaviour has been made but a serious problem remains of defining the exact physical conditions at the user/telephone interface. The coupling between mouth and telephone is fairly well represented by existing artificial mouths but the corresponding problem at the ear has not had a satisfactory solution proposed.

THE ARTIFICIAL EAR

Presently the IEC artificial ear is used to obtain objectively measured data for use in the computer simulation of a telephone connexion. However the IEC ear is designed for audiometric use and a correction which is frequency dependent, and also dependent upon telephone handset design and manner of use, is required when it is used telephonometrically. The correction presently in use is not satisfactory and work is in progress leading to a better understanding of the problem and having the objective of providing a reliable and more suitable correction. The correction must not only be more accurate but also be flexible and applicable to the range of handsets in use or likely to be introduced in the future.

The reason for the unsuitability of the IEC ear for telephonometric use is that it is designed on the assumption of a perfect seal between earphone and artificial ear. Clearly this does not represent the behaviour of the average telephone user who will hold the handset loosely to his pinna introducing a leak. The amount of leak will be greater if the perceived speech is loud and loud side-tone will tend to cause a user to move the handset slightly further away from his head.

EXPERIMENTAL WORK

A series of couplers have been made for use with the IEC ear which introduce controlled amounts of leak and at the same time increase the size of the cavity. No attempt has been made to simulate any actual real ear situation but rather to cover a wide range of possibilities which will embrace all likely real ear situations. Several UK and overseas administration types of handset and earphone transducer have been used and earphone sensitivities obtained for each on the IEC ear using each of the couplers.

The sensitivities of the earphones have also been obtained when the handsets have been used by human observers listening to telephone conversation quality speech at different intensities ranging from very loud to very quiet. Sensitivities were obtained by the insertion of a probe microphone at the ear reference point to measure the sound pressure produced by the speech and subsequently to measure the sound pressure produced by a swept frequency audio tone.

Space does not permit the reproduction of any results here and they will be given in the full paper which will be available at the congress.
EXPERIMENTAL INVESTIGATION OF THE EFFECT OF INTONATION ON THE INTELLIGIBILITY OF SYNTHETIC SPEECH

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INTRODUCTION
Speech perception can be understood as an error-correcting process rather than a bottom-to-top analysis. According to this view, the hearer forms certain rough assumptions on basis of some cues and then tries to verify or reject these hypotheses with the help of additional data. It has been proposed that the hypothesis phase relies heavily on intonation, segmental timing, and allophonic variation (1). The aim of this study was to investigate the influence of intonation by separating sentence structure from intonation contour and manipulating them independently.

STIMULUS MATERIALS
The stimulus materials consisted of 6 pairs of sentences, taken from Garrett et al. (2), for which some string of lexical items was common to each member of a pair, whereas constituent structure of the one member differed from that of the other. For example:
(A) In order to catch his train // George drove furiously to the station.
(B) The reporters assigned to George // drove furiously to the station. (// = constituent boundary)
Tape recordings of these sentences, spoken by a native speaker of English, have been made, and fundamental frequency contour was extracted with the aid of a PDP 11/35 small computer. On basis of the extracted F₀ contour, two different kinds of stimuli were synthesized with the Edinburgh Speech Synthesis by Rule System (3, 4):
I. "SAME": F₀ contour and word string are taken from the same member of a pair.
II. "DIFFERENT": F₀ contour taken from one member of a pair is combined with text of the other. Additionally, 4 "dummy" sentences have been synthesized.

EXPERIMENTAL PROCEDURE
Two experiments, each including 48 subjects without any known hearing impairment, have been conducted. In both experiments, each subject is presented with 6 experimental and 4 dummy sentences in randomized order via earphones. 3 of the 6 experimental sentences match the SAME condition, the other 3 the DIFFERENT condition. Each sentence is applied 6 times.

In the first experiment, subjects had to repeat the heard sentence, and the responses were recorded on tape. It was hypothesized that performance (= number of correctly repeated words) would be better in case of the SAME than in case of the DIFFERENT condition.

In the second experiment, an ABX procedure was used. After each trial, the subject was presented with the written versions of the two members of a sentence pair, and had to decide, whether (A) or (B) was more similar to the experimental sentence (X). Firstly, the number of correct responses was expected to be greater in the SAME than in the DIFFERENT condition. Secondly, an interaction effect between stimulus structure (SAME vs. DIFFERENT) and number of trial was hypothesized in such a way that, in case of the DIFFERENT condition, the first trials should reveal a predominance of incorrect, whereas the last trials should reveal a predominance of correct responses.

Results of the two experiments are discussed in terms both of perceptual analysis of speech and implications for foreign language teaching and hearing aids for the deaf.

REFERENCES
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INTELLIGIBILITE DE PHRASES SYNTHETIQUES ALTEREES: APPLICATION A LA TRANSMISSION PHONETIQUE DE LA PAROLE(*)

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Introduction

On se place ici dans la perspective de la transmission phonétique de la parole (débit de l'ordre de 50 bits/s): le maillon d'entrée est un système de reconnaissance automatique de la parole continue, en temps réel ou légèrement différé, actuellement en cours d'étude; le maillon de sortie est un synthétiseur à commande phonétique actuellement réalisé au LIMSI (ICOPHONE 5, (1 - 3)). La question posée est la suivante: quels sont la nature et le taux maximum des erreurs phonétiques "tolérables" dans la reconnaissance, c'est-à-dire des erreurs rectifiables par l'auditeur humain qui utilise toutes ses facultés de compréhension, notamment sur les plans lexical, syntaxique et sémantique?

Expérimentation

On a choisi une liste de 80 phrases courantes, provenant du CNET qui les utilise pour des tests d'intelligibilité (exemples: "Le gamin est parti à l'école", "Le bouffon amuse le roi" etc.). Ces phrases possèdent une structure syntaxique simple mais variant de l'une à l'autre, et une unité sémantique, tout en étant totalement indépendantes les unes des autres. Leur longueur moyenne est de 18 symboles phonétiques. On s'est limité au cas des erreurs de substitution (erreurs du système de reconnaissance, la segmentation en phonèmes étant supposée correcte). On a distingué trois types d'erreurs:

- altérations de voisinage: la substitution porte sur des phonèmes voisins, c'est-à-dire différant par 2 traits distinctifs au plus. Ex. [lagam] au lieu de [lagamg]
- altérations de classe : toute voyelle peut être substituée à une voyelle, toute consonne à une consonne. Ex. [lagag] au lieu de [lagaam]
- altérations quelconques : n'importe quel phonème peut être substitué à n'importe quel autre. Ex. [lagamZ] au lieu de [lagaam]

On a également fait varier le nombre des substitutions effectuées dans chaque phrase, de 0 (étalonnage) à 4. Les phrases altérées ont été synthétisées, puis écoutes et prises en note par 10 auditeurs.

Résultats (figure ci-contre)

Dans l'ensemble, le nombre de mots correctement perçus en contexte de phrase décroît avec le nombre des altérations et avec leur gravité. Une erreur de voisinage passe inaperçue, mais deux ou trois entraînent une baisse sensible. Dans tous les cas la confusion d'une voyelle avec une consonne est catastrophique. On peut donc admettre un taux d'erreur phonétique de l'ordre de 15% sans dégradation trop importante de l'intelligibilité, à condition que les erreurs respectent au moins la distinction voyelle/consonne. Ce résultat est confirmé par nos expérimentations de reconnaissance.

(*) Travaillé effectué dans le cadre du contrat DRME 73/169
(1) E. LEIPP, M. CASTELLENGO, J.S. LIENARD - 6e ICA - Budapest (1968)
(2) J. SAPALY - 7e ICA - Tokyo (1971)
(3) D. TEIL - 8e ICA - Londres (1974)
HEARING AIDS AND THE TELEPHONE - A SEARCH FOR BETTER MAGNETIC COUPLING

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INTRODUCTION

Hearing aid designs often rely on a magnetic pickup, called a telecoil, in order to receive signals from a telephone. However, telephones are optimized to convert electrical signals into sound pressure, and thus any magnetic field created by the receiver has been more by accident than by design. Also, a telecoil in a hearing aid is primarily included for magnetic loop PA systems and in many cases not positioned properly for effective magnetic coupling to the telephone. The magnetic field from a telephone receiver or telephone adapter have not been properly characterized, making it difficult for manufacturers to improve this way of coupling.

A number of experiments were performed at BNR to characterize various magnetic fields and analyze calibration methods. The presentation describes results and conclusions; a few are summarized here.

EXPERIMENTS AND RESULTS

A sensor (search coil), with the same dimensions as a typical telecoil in a hearing aid, was used to characterize the magnetic field generated by commonly used telephone receivers (or couplers). This field can be presented in terms of its parallel (\(H\)) and perpendicular (\(\perp H\)) components. An example of these values versus the distance from a typical receiver's center is shown in Figure 1. The perpendicular component is always a maximum at the center, however the maximum of the parallel component depends on the angle \(\alpha\) between the telecoil's axis and the receiver's diameter. Its greatest value occurs when the axis is aligned (\(\alpha = 0\), i.e., radial) otherwise the telecoil's output decreases with \(\cos \alpha\).

The output voltage from telecoil vs frequency is shown in Figure 2 for both, the IEC-118-1 magnetic loop and a typical telephone receiver.

CONCLUSION

Presently hearing aid users are forced to search for the best pickup from the telephone receiver by displacing the handset from its normal position. The above characterization of the receiver's magnetic field may eliminate this problem by making it easier to understand and optimize magnetic coupling with better orientation of the telecoil within the hearing aid.
INTRODUCTION

An electronic model simulating the severely restricted amplitude range of hearing typical of profoundly deaf subjects has been used to investigate the effect of this one factor on speech intelligibility.

THEORETICAL BASES

The model rejects signal elements whose levels fall below electronic thresholds in each of 16 adjacent 1/3-oct bands. These thresholds are set to represent the average measured hearing thresholds of a group of profoundly deaf subjects. The input signal level is adjusted so that rms speech peaks fall just below levels representing the average measured discomfort levels of the group. Within the amplitude pass band (ca 15 dB at lower frequencies), amplitude expansion simulates the abnormal loudness growth of recruitment (1).

EXPERIMENTAL TECHNIQUES

Each of the frequency bands is processed separately, so that signal levels in one part of the spectrum do not open the threshold gates or affect the expansion of signals at other frequencies. The processing is performed by a computer, which continuously scans the output of peak detectors for each band and in turn controls digital attenuators in each band.

RESULTS AND CONCLUSIONS

The upper oscilloscope trace in Fig. 1 is the wave envelope of a 2-sec passage of unprocessed speech. The middle trace is of the same passage processed by the threshold/discomfort model without expansion, and the lower trace shows processing which includes expansion. Silent periods appear in the processed speech whenever signal levels drop below all of the thresholds. The processed speech is unintelligible.

The performance of this model suggests that attempts to improve speech reception for the profoundly deaf may be compromised by a failure to deal with the problem of a very narrow dynamic range of hearing. Some profoundly deaf subjects with a memory of normal hearing have indeed reported that amplified speech sounded "broken up". The remedy, which is a necessary but probably insufficient condition for providing such subjects with good speech reception, lies in amplitude compression.

REFERENCES

Murry, T. Veterans Administration Hospital, San Diego, California
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The general term, "hoarseness" has been used by clinicians and laymen alike to describe a variety of perceived vocal abnormalities. These may range from physiologically based problems such as laryngeal cancer to psychological manifestations associated with emotional stress. Attempts to provide specific perceptual classifications of voice disorders have met with little success due to lack of agreement on the definitions of perceptual terminology. It is apparent that "hoarseness" is a complex concept and probably not fully defined on any one perceptual dimension such as pitch or loudness. A voice that is hoarse may evoke perceptual constructs associated with physiological, acoustical, or psychological parameters. It would appear then that listeners are responding to more than one stimulus parameter when they are asked to judge the degree or existence of hoarseness.

Recently, a multi-dimensional technique known as INDSCAL has demonstrated its usefulness in the classification of stimuli with obscure perceptual parameters. Input to INDSCAL analysis is a proximity estimate between the stimuli. In this study an equal appearing interval scale was used to obtain similarity judgments from populations of normal and disordered voices. The specific purpose of the present research was to obtain the perceptual attributes associated with normal and non-normal voices.

Twenty samples from groups of normal and pathological voices were randomly selected from a library of 250 voice samples. Each of the subjects in the pathological group repeated the vowel /a/ for approximately 4 seconds. A 2.5 second segment (excluding the initial 50 msec.) was extracted and then each sample was paired with itself and the 19 other voices. For the normal subjects, the phrase: "These take the shape of a long round arch," provided the stimuli for analysis. Listeners were instructed to rate the similarity of each pair of voices on a 7-point scale, "one" representing maximum similarity and "seven" representing maximum dissimilarity.

For both the normal and abnormal data, a solution was obtained. The solution for the pathological group was five dimensional; for the normal group, it was a three dimensional solution. Interpretations of the INDSCAL solutions were made on the basis of physiological and acoustical data obtained from each talker such as speaking fundamental frequency, mean air flow rate, and fundamental frequency values. The interpretation of the data suggest that distinct perceptual features exist for normal and abnormal phonation and that these features may be explained on the basis of acoustic and physiologic phonatory parameters.
AUTOMATIC RECOGNITION OF TELEPHONE SPEECH FOR APPLICATION IN FULLY AUTOMATIC INQUIRY SYSTEMS

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The problem of the automatic recognition of telephone speech for application in fully automatic inquiry systems includes, in consideration of the technological state of the art and in the author's opinion, two very important subproblems:

- development of regimented instructions for the inquiring subscribers,
- elimination of disturbances in the telephone speech.

The regimented manner of speaking will be necessary for overcoming the difficulties resulting from the admission of a nearly unlimited number of different speakers and of the enormous number of words to be recognized. But this regimentation should be simple enough to be acceptable for the public. Experimental investigations have shown that spelling of words represents a kind of regimentation which is both generally well known and suitable for coding all words and also permits a reduction of the technical equipment.

It is less simple to solve the problem of eliminating the disturbances. There are two classes of disturbances,

- disturbances caused by the transmission line (noise, whistling, level variations, channel crosstalk, clicks, dialling noise),
- disturbances caused by acoustical influences at the site of the telephone station (environmental noise including traffic noise, music, competing talkers).

Besides this, telephone speech is always limited in bandwidth.

The evaluation of real-time-Cepstra permits only the detection of the "voiced" feature irrespective of disturbances (2). Further features such as formant-data are unmistakably extractable free off disturbances only provided that it is possible to separate formants and disturbances automatically. With an eye to this fact, a real-time procedure was developed and tested, which investigates unknown energy-concentrations in fine-resolution spectrograms to find out whether or not they contain a spectral-line-structure. The procedure permits the detection and localisation of formants even if the vocalic telephone speech is superimposed by different disturbances. In the cases of superpositions by music and competing talkers, a rejection is carried out automatically to avoid any recognition error.

The procedure was applied to the automatic recognition of spelled German words and to the automatic recognition of German digits 0 to 9. Examples will be given to demonstrate the reduced influence of disturbances on the recognition rate.

REFERENCES

INTRODUCTION

A certain speech sound is defined by the type of excitation function (voiced/unvoiced), by the individual anatomy and by the position of the vocal tract. According to the position of the vocal tract the anatomy will be projected more or less completely on the realization of this phoneme, i.e., the content of speaker-specific information in the speech signal is dependent on the articulated phoneme. The influence of this effect is registered by setting up a table showing the efficiency of phonemes for the discrimination of speakers. On this basis code sentences for speaker recognition can be selected.

METHOD OF INVESTIGATION

24 phonemes of the German language, spoken in isolation, were recorded in an anechoic chamber. Utterances of 12 speakers, each with 10 realizations per phoneme were available, altogether 2880 patterns. The speech signal was sampled at a rate of 12 kHz, and weighted by a 42.6 msec Hamming window.

The recognition rate of classification was chosen as the criterion for the phoneme quality. Recognition rates are not only dependent on the type of phoneme, but also on the efficiency of analysis. In order to eliminate this influence, the parameter sets of different methods of analysis were averaged.

RESULTS AND CONCLUSIONS

Fig. 1 shows the order of phonemes according to their efficiency for speaker recognition. The diagram (a) shows average recognition rates obtained by the Mahalanobis minimum-distance classifier. The diagram (b) shows the results obtained by a Euclidean minimum-distance classifier. Each parameter set consisted of 18 coefficients.

There are significant differences between the phonemes as regards their information content concerning the speaker. Especially suitable are the nasal phonemes, because they are mainly influenced by the nasal cavity, which can hardly be varied by the speaker.
INTRODUCTION

In automatic speaker verification based on code phrases, full use can be made of the temporal structure of the utterance, if corresponding speech events can be coordinated correctly. For that purpose a time registration is necessary, which normalizes the differences in speaking rate and establishes time equivalences between test and reference data.

Time registration allows to compare directly the speech parameters of the corresponding sounds or sound transitions. On the other hand evaluating the normalized time function provides additional features like temporal variations of the speech parameters and their warping functions.

EXPERIMENTAL TECHNIQUES

Several methods for time normalization are being investigated, for instance measuring the stationarity of the spectra or pattern matching in the frequency domain. The latter procedure, which searches for key points in the utterance /1/, proved to be qualified for this purpose. Here the wanted speech event is described by a typical sequence of spectra (reference pattern). Sliding this reference pattern along the test utterance the distance between reference and test spectral data provides a measure of similarity:

$$A_n = \sum_{k=1}^{L} \sum_{i=1}^{M} \left| a(n+k,i) - a_r(k,i) \right|$$

$$a(n,i) = \text{Intensity of the } i\text{-th channel in th } n\text{-th spectrum}$$

$$L = \text{Number of the spectra in the reference pattern}$$

$$M = \text{Number of the used channels}$$

RESULTS AND CONCLUSIONS

The procedure can be accelerated, if the number of the channels is reduced, that is if the channels $$a(n,i)$$ are grouped together:

$$a(n,j) = \sum_{l=j}^{M} a(n,l) , j = 1 \ldots N, N < M$$

This grouping is optimized by logarithmic scaling of the frequency axis. A comparison between logarithmic and linear scaling of the channel intensity has been realized.

Last results of the method will be given in combination with a procedure of measuring the stationarity of the spectra.

REFERENCES

INTRODUCTION

From the modular speaker recognition system AUROS, the module COMEX (Codewortbezogene Merkmalsextraktion) is described and results are discussed. COMEX, which derives speaker-specific features in the frequency domain, is used for the analysis of a prescribed utterance and is therefore best suited for speaker verification (1).

FUNCTION OF COMEX

First the speech signal is analyzed by an analog filterbank, the channels of which are sampled and digitized every 10 msec. An utterance is then described by a matrix

\[ X = \begin{bmatrix} x_{11} & \cdots & x_{1L} \\ \vdots & \ddots & \vdots \\ x_{K1} & \cdots & x_{KL} \end{bmatrix} \]

where the length \( L \) of the utterance is variable due to differing speaking rates. To obtain a precise alignment of corresponding speech sounds, utterances have to be time-normalized (2):

\[ X_{\text{time-norm.}} \rightarrow Y = \begin{bmatrix} y_{11} & \cdots & y_{1M} \\ \vdots & \ddots & \vdots \\ y_{K1} & \cdots & y_{KM} \end{bmatrix} \]

Finally several methods of feature-reduction are applied to the matrix \( Y \), for instance statistical methods like F-ratio (3) and more empirical procedures, which take into account the nature of speech production.

CONCLUSIONS

Currently time-normalization as well as feature-reduction techniques are optimized with regard to performance and expense. Generally speaking, processing of specific columns of the matrix \( Y \) means evaluation of the speaker's actual vocal-tract configuration, whereas processing of specific rows means evaluation of dynamic speech habits. Resulting feature vectors are classified by a Mahalanobis minimum-distance classifier, which is invariant to any regular linear transformation of the feature space. It is assumed that the influence of different transmission characteristics (such as dialed telephone lines) can be reduced by long-term normalization techniques. Newest experimental error rates of the entire system are compared and discussed in the lecture.

REFERENCES

The aim of automatic speaker recognition is to identify persons by characteristics of their voices, thus allowing the introduction of an "acoustical passport" for security systems in banking and law enforcement. The two fundamental problems related to that task namely the non-reproducibility of human speech and the large amount of data to be processed will be overcome by combining real-time speech analysis techniques and statistical methods of pattern recognition. This is being done within three phases:

Preprocessing phase: The highly redundant speech signal is analyzed by real-time signal analysis and data compression techniques, and is described by a voice typical feature vector.

Training phase: Due to the non-reproducibility of the speech signal, one utterance cannot be regarded to be representative for a speaker's voice. So, from the feature vectors of at least 10 training utterances, an individual voice reference is evaluated using pattern recognition techniques.

Reliability test phase: Utterance, being unknown to the system will be classified automatically. From the number of correct classifications and mix-ups, the system specific recognition rate is being calculated.

For solving the problem, the large modular speaker recognition system AUROS with a structure shown in fig. 1 has been developed within a government sponsored project by the "Philips Forschungslaboratorium Hamburg" and the "Heinrich-Hertz-Institut" Berlin.

It consists of a large number of different signal analysis processors mainly performing two-stage statistical measurements respectively devices for evaluation of three dimensional time-frequency-amplitude matrices (paper of Jesorsky). For classification various different pattern recognition algorithms for identification and verification tasks are available.

A supervising program allows combination of any analysis technique with any classification method for comparative experiments to investigate certain aspects like finding suitable segmentation methods (paper of Kriener) or suitable vocabularies (paper of Höfker).

Up to now, recognition rates of 99.9% could be obtained for a large population codeword-related as well as text-independent. This paper will describe the structure of the speaker-recognition system, results are being discussed.
SPEECH SYNTHESIS IN THE TIME DOMAIN IN CONSIDERATION OF PROSODIC PARAMETERS

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INTRODUCTION

Concatenation of small sections of sounds is one possibility to get synthetic speech with an unlimited vocabulary. There are several advantages of this method, for instance: the easy and economical implementation and the high speed of synthesis (one can multiplex 100-200 channels). But we also found some special difficulties when taking into account the prosodic parameters.

PRINCIPLE OF SYNTHESIS

Following Küpfmüller's synthesis procedure (1) by using diphones in the time domain (storage volume in the case of a digital system: more than 1 MByte if only monotonous speech is required it was possible to considerably reduce the necessary memory by the following steps:

1. Transformation of vowels e.g. \( e \rightarrow i \) (2)
2. Transformation of consonants e.g. \( f \rightarrow p \)
3. Storage of only one pitch-period of each voiced sound (3)
4. Storage of only the high informative transition part of the diphones
5. Reversal of these transition elements. Therefore we need only one transition for e.g. the two diphones "mo" and "om".

In this way it is possible to synthesize an unlimited vocabulary with a memory of only 64 KBytes, prosodic elements included (4).

PROSODIC PARAMETERS

Variation of duration of sounds is very easy. The stored single pitch period has only to be repeated as often as necessary. In case of unvoiced sounds the whole phone can be shortened by cutting the signal or lengthened by repeating.

Variation of the sound volume can be achieved by multiplication with a weight function. But variation of melody (pitch) presents some difficulties because the elements are stored in the time domain. By changing the sampling rate during reproduction of speech it is, indeed, possible to vary the pitch, but the formants are changed, too (similar to the so-called Mickey Mouse speech). These difficulties can be avoided by the "predictive coding" scheme. One can change the periodicity of the difference signal and use this processed signal for exciting the vocal tract filter. To save the speed of synthesis and simplicity of the system design, the melody segments have to be stored in an analysis phase and are only read out during the synthesis.

We are presenting a first outline of a hardware-implementation of the synthesizer described above. The quality of the synthesized speech will be demonstrated acoustically.

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RULE SYNTHESIS OF SPEECH FROM DYADIC UNITS

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Peterson, Wang, and Sivertsen (1) suggested the use of the units called "dyads" which are characterized as (a) containing parts of two phones with their mutual influence in the middle, and (b) beginning and ending at the phonetically most stable position of each phone as the basic unit for speech synthesis. This paper describes an approach to speech synthesis by rule using as the basic element a unit smaller than a dyad as defined by Peterson et al.; we propose to use only the transition between the two phones of the dyad. The steady state portions can then be obtained by connecting the end points of adjacent transitions with straight lines. This is thought possible because of recent work by Olive and Spickenagel (2) which has demonstrated that continuous speech can be resynthesized from spectral parameters which have been linearly smoothed between phonetic boundaries: phonetic boundaries were defined as the end points of the "steady states" of the phonemes, and of the transitions associated with successive phonemes. This work lead to further simplification of the dyad; since linear smoothing maintains the speech quality, only the end points of the transitions need to be specified.

Such a dyadic procedure was then used to provide segmental values for a complete rule synthesis scheme. The entire scheme consists of a dyad dictionary for obtaining segmental data, of a word pronouncing dictionary which also includes certain prosodic data, and of concatenation and further prosodic rules.

The dyad dictionary contains the segmental data for the dyads: it is structured as a matrix with the various phoneme pairs as entries. In this scheme the segmental values are specified in terms of LPC derived area functions for the end points of the transitions.

The word pronouncing dictionary contains the phoneme sequences corresponding to the orthographically specified text input. This dictionary also contains the timing information for the steady state and transition intervals for each phoneme in the word. Pitch prosody is obtained by the rules for the fundamental frequency described in a previous paper on rule synthesis by word concatenation (3).

REFERENCES

INTRODUCTION

In the synthesis-by-rule scheme that has been developed at Haskins Laboratories, the phonetic syllable, rather than the phone, is taken as the unit. This approach reflects the traditional Haskins view that the correlates of the phonemes at the acoustic level are "overlapped, or shingled, one onto another," yielding "irreducible segments of approximately syllabic dimensions"(1). The syllable is considered as a cyclic process, passing from onset to peak to offset as the vocal tract moves from a more closed to a more open to a more closed configuration. This process is realized in many different ways, depending on the phonetic choices made by the speaker, i.e., the values he assigns to an ensemble of phonetic features. Traditional phonetic segments have no formal status; both the difference between [pa] and [ba] and the differences between [pa] and [pla] depend upon a feature value selection.

STRUCTURE OF THE PROGRAM

The program is written in FORTRAN for the PDP 11/45 and calculates parameter values for a software simulation of OVE III. The input to the program is a phonetic transcription of an utterance in binary syllable features. Thus

LOW, BAC
IGL, IPL, HGH, FRO

represents a low back vowel in the first syllable and a high front vowel with an initial palatal glide in the second syllable, i.e., [ayr]. Ordered rules relate these feature values to the input variables of the algorithm for calculating parameter values.

In this algorithm the character of a syllable is considered to be determined by numerous influences; the vowels of the previous, current and following syllables, the final consonants of the previous and current syllables, and the initial consonants of the current and following syllables. With each such influence is associated a target value Tᵢ for each parameter and a curve Iᵢ that represents the extent of the influence over time. An influence curve is a modified exponential function of the form e⁻βt. The value of the function is restricted to the range 0...1. At a certain time after the notional beginning of the syllable cycle, β becomes negative, so that the influence will begin to diminish. Other variables determine the basic duration of the syllable and the incremental duration attributable to a particular influence.

The actual parameter values for a particular five-millisecond sample of speech are derived by an iterative calculation. The influences are regarded as ordered, from vowels to fricatives to stops. At each iteration, the value Vᵢ computed for a parameter is

\[ Vᵢ = Vᵢ₋₁ + Iᵢ(Tᵢ - Vᵢ₋₁) \]

(V₀ is the target value for the vowel of the previous syllable). Influences with values close to zero are ignored.

CONCLUSIONS

The scheme is in fact quite general, and could be implemented in terms of articulatory gestures, or vocal tract shapes, or formant movements, depending upon the parameters chosen. It is felt that this scheme will facilitate the incorporation into synthesis by rule of recent developments in phonetic theory.

REFERENCES

Most allophonic variation is conditioned within the bounds of a syllable. This fact provides the rationale for a speech synthesis-by-rule system in which the rules are generally restricted to those accounting for intersyllabic coarticulation (1). Furthermore, in the case of English, for example, we can reduce the 10,000-member syllabic inventory to around 1000 units by treating each syllable as a syllable core, decomposable into initial and final demisyllables, plus optional syllable affixes (2). In this treatment, each demisyllable may contain, as either an initial or a final syllable feature, only one consonantal place-specification (/sp, st, sk/ are considered as single units with a single place-specification). More complex syllable-final consonant clusters are analyzed as core-final consonantal features and one or more affixes, the apical consonants /tdsz0.../, if and only if the core-final and affixal voicing features agree. For example, the English syllable [strenk fut] would be analyzed as shown in Fig. 1.

We report on the results of studies of the feasibility of this approach to speech synthesis-by-rule. Concatenation of affixes to syllable cores has produced high quality synthetic speech. For example, in one experiment, phonetic segments corresponding to /s/ and /z/ were spliced as waveforms onto a set of CVC syllable cores in which the core-final consonant varied in terms of place and manner of articulation. The original and concatenated versions of each test syllable were presented to listeners who rated each version according to naturalness. The results indicated that, in general, there was no difference in rated naturalness between the concatenated syllables and the originals (Macchi and Ngro 1976, unpublished).

We have also investigated the problem of concatenating demisyllables to produce syllable cores. These investigations have shown that the initial demisyllable should be defined as a small, fixed-length portion of the beginning of the syllable, extending slightly past the CV transition, if any (the exact definition varies somewhat with the class of the initial consonant). The remainder of the syllable core comprises a final demisyllable. Using linear-prediction-coded parameters derived from isolated utterances of English monosyllables, we have found that such demisyllables can be successfully concatenated using a simple parametric smoothing algorithm to eliminate perceptible discontinuities. The nature of the cutting and smoothing procedures is such that context-sensitive variations in vowel duration or color (nasalization, etc.) determined by the final consonant are taken care of with no need for additional phonological rules within the syllable. We may freely combine, for instance, CV- and -VN, taken from CVC and CVN monosyllables, respectively (3).

REFERENCES

EXTENSION OF THE JSRU SPEECH SYNTHESIS BY RULE SYSTEM


INTRODUCTION

Since the Joint Speech Research Unit system for speech synthesis by rule was published in 1964 (1) machine-man communication has become increasingly important. To study further the use of synthesis by rule for voice output from machines this system has been re-implemented and extended to take advantage of a new hardware parallel-formant speech synthesizer.

SPEECH SYNTHESIZER

The new synthesizer is interfaced to a PDP-11/20 computer and is based on a software design previously described by Holmes (2). Improvements over the synthesizer previously used (1) include the provision of a low-frequency formant circuit for use in the modelling of nasalized sounds, improved representation of the glottal volume velocity waveform, and the ability to generate mixed voiced and unvoiced excitation.

STRUCTURE OF RULE SYSTEM

The rule system has been implemented in software and is divided into a number of separate programs, each operating at one of a hierarchy of linguistic levels; prosodic, upper-phonetic, and lower-phonetic (3). A modified version of the original JSRU system is implemented by the two phonetic level programs.

The programs are written in FORTRAN IV for running on the PDP-11 computer and data communication between them uses disc files that can be examined and edited interactively. This facility provides the user with a means of experimentally determining the effect of detailed changes at all levels of the system.

SYSTEM PROGRAMS

Input to the highest level (prosodic) program consists of a string of symbols representing a phonemic transcription of the text to be synthesized, and prosodic markers as required. The prosodic marking system currently implemented is based on that described by O'Connor and Arnold (4) and allows for both tone and stress markings. Output consists of the list of phoneme labels with associated numerical values for the perceptual correlates of pitch, length and stress, and additional syllable division markers.

The upper-phonetic level program converts this phonemic and prosodic information into phonetic information. A distinctive features table is used to identify phoneme characteristics and handle their expansion into corresponding phonetic segments. Durations of the segments are computed by rule from a mixture of prosodic input information, context, and the distinctive feature description of each segment. Output takes the form of phonetic segment labels with associated numerical values for the physical dimensions of fundamental frequency, duration, and amplitude.

The lower-phonetic level program converts these data to synthesizer control parameter values, expressed in terms of formant amplitudes and frequencies (in dB and Hz respectively) and excitation information specified every 10 ms. This program implements the JSRU rules for segmental synthesis, and requires tables of the synthesizer parameter values associated with each phonetic segment in the inventory, and of information controlling transitions between segments. A simple routine converts this non-synthesizer-specific output to the format required to drive the JSRU synthesizer.

Recordings will be played to illustrate the performance of the system.

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(3) R.D. Wright, Proc Institute of Acoustics (1976) 2-20-1
APPLICATION OF THE WAVE DIGITAL FILTER CONCEPT TO SPEECH SYNTHESIS

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It is known [1], that an acoustic tube of finite length can be described by an equivalent electric two-port which is the same for an electric transmission line of finite length. The human vocal tract can thus be modelled by an acoustic tube consisting of sections of equal length and constant area. For a digital model one has to find a two-port which describes a section of a transmission line. This two-port can be found among the elements used for wave digital filters [2]. Their theory is based on wave quantities rather than on currents and voltages. If we use voltage waves, the instantaneous values of the incident wave \( a = a(t) \) and the reflected wave \( b = b(t) \) are related to the instantaneous values of the voltage \( v = v(t) \) and the current \( i = i(t) \) at a port with port resistance \( R \) by

\[
a = v + R \cdot i, \quad b = v - R \cdot i \quad (1)
\]

The two-port one needs to solve the problem mentioned above is the unit element, known from microwave transmission line filters. The difference equations for the instantaneous waves describing the voltage-current relations of the unit element (Fig.1a) are

\[
b_1(t) = a_2(t-T/2), \quad b_2(t) = a_1(t-T/2) \quad (2)
\]

with \( T \) being the sampling period. The corresponding wave flow diagram is shown in Fig.1b. In order to connect an arbitrary number of ports with different port resistances we use adaptors. In our case we only need a two-port adaptor to implement the digital model of the human vocal tract. Its wave flow diagram is shown in Fig. 2 with

\[
a = \frac{R_1 - R_2}{R_1 + R_2} \quad (3)
\]

Various authors, mentioned in reference [3], came to similar results. But only by using the results of [4] one can implement digital filters with finite wordlength, which are completely free of parasitic oscillations for zero input conditions. By means of a three-port adaptor a nasal tract can be incorporated as well. Further work will be done, trying to utilize more of the theory of wave digital filters in the field of speech synthesis.

REFERENCES

MICROPROCESSOR CONTROLLED SPEECH SYNTHESISER

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INTRODUCTION

A digital speech synthesiser described here is based on synthesis of building blocks. The aim of this system is to operate with Yugoslav phonetic languages. These languages, being phonetic ones, require phonemes for building blocks stored in read-only memory, in terms of linear predictor coefficient parameters. Microprocessor is used as a system controller, which enables simple, inexpensive and autonomous real-time realization. The size of read-only memory (ROM) depends upon required speech quality. To obtain the real-time realization linear predictor (LP) is designed as a separate hardware for fast computing the sum of $a_n x^{n-k}$ (1).

DESCRIPTION

The program is stored in ROM as well as predictor coefficients for all existed phonemes. An excitation for LP is generated by the program, and depends upon phonemes which are to be synthesised. An excitation amplitude is determined by the program based on the simple grammatical rules (the beginning of the sentence, comma, full stop, question mark). Computed data for each phoneme (excitation and coefficients) are transformed in the form acceptable for LP, and stored in random-access memory (RAM). LP takes excitation data from RAM every 100 µs, and coefficients every 10 ms. Data transfer from RAM to LP is in direct memory access mode (DMA), independently to program, what makes real-time synthesis possible. The predictor is designed as an all-pole recursive filter (p=12). The coefficients are represented as 8-bit length words, and a sampling rate is 10 kHz.

APPLICATIONS OF THE SYSTEM

This system is acceptable for a wide consumer area due to low price and small dimensions. It can be used as a part of computer system equipment, but it can also operate autonomically. There is an input in ASCII-code serving for direct on-line connection to a teletype or any other keyboard device.

REFERENCES

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INTERACTIVE TEXT-TO-SPEECH SYSTEM FOR ITALIAN PROSODIC RULES DETERMINATION

INTRODUCTION

A speech synthesis method is presented, based upon joining sub-word segments, coded as parameters of an acoustic speech production model (1). The number of segments necessary to obtain all Italian nouns and words is about 150 (2). These segments can be easily coded to 4800 bit/s, for a standard speaking rate, and so they need about 9 kwords of storage.

SYSTEM DESCRIPTION

The synthesis system accepts as input any character string in normal orthography. The string is then converted into a sequence of elementary segment addresses by the use of a set of tabulated rules (3). The system allows to modify the duration, the intensity and the pitch period of each segment to obtain the desired prosodic contour. The example shown in fig. 1 puts into evidence the pitch and syllable duration contours for a simple sentence. This can be accomplished both manually and automatically according to those rules which can be considered quasi-definitive.

In the former case the operator modifies the pitch, the intensity and the duration by means of simple commands according to a prescribed syntax. The curves relative to the above mentioned parameters can be partially or totally changed by subsequent interventions.

In the latter case the preliminary implemented rules were derived by the study of simple sentences, but provisions are made for the handling of more complex ones.

For the application of the rules, some pre-processing is performed on the input string, such as syllable boundary location, stress assignment and breath-group division. This is done by the only aid of the punctuation and the knowledge of a few short functional words such as articles prepositions etc.

This first approach seems to indicate that it is possible to derive a more complete set of rules which permit to obtain a synthetic speech with a good degree of naturalness.

REFERENCES

By visual inspection, pitch synchronously computed short-time power spectra show a significant maximum-minimum structure and a smooth variation in time. So those spectra seem to be helpful in automatic, real-time formant extraction by peak-picking or similar methods in vocoder systems. It is shown, however, that the segmentation errors of the device used for pitch determination lead to spectral peaks, which are not closely related to the slightly damped poles of the vocal tract. Therefore, the short time spectrum of a fundamental period is computed by averaging the power spectra of gliding, Hanning weighted frames of fixed length over the duration of the current pitch period. The result is a smooth short-time power spectrum with peaks showing a significant relationship to the weakly damped poles of the vocal tract during the articulation of voiced sounds. The consideration of the time varying spectra of shifted frames also gives some insight into the articulation process. This method might also be used for pitch extraction of bandlimited signals whose spectra do not contain the fundamental frequency itself.

Combining both pitch extraction and spectral analysis in one procedure gives a new, easy to use tool for formant data extraction of bandlimited speech signals.
INTRODUCTION

As part of a study to characterize the integration of Fo production and pitch perception an experiment was conducted in which subjects attempted to match step changes in the frequency of a target sinusoid with the Fo of their own vowel production. Previous experiments have shown that subjects perform significantly better in pursuit auditory tracking tasks when a cursor tone controlled by a speech articulator is fed back to the right ear and a dynamic target tone is presented to the left ear over the opposite configuration (cf. 1). These same experiments did not find a right ear advantage (REA) when the cursor was controlled by non-speech-related movements. In a study of accuracy in matching the Fo of their own vowel production to a static target tone, better performances were generally characterized by an REA although not to a statistically significant extent. The present study tested the hypothesis that the task had to be dynamic in nature before a significant REA would occur, otherwise there would not be enough demand placed on the system to demonstrate a difference in performance as a function of different processing pathways.

METHODOLOGY

Ten males, screened to be dominant right-handers without previous vocal training, served as subjects. They performed two trial runs with 48 pitch matching trials in each run. The target was a sinusoid calibrated to 70 dB SPL, and the cursor was adjusted to an equal sensation level for each subject on each run. A trial consisted of presentation of a level initial target tone, then a random step up of 5, 10 or 15 Hz in target frequency. The target never went above 155 Hz or below 105 Hz, well within the subjects' range. Subjects were instructed to react as rapidly and as accurately as possible to the change which occurred in every trial. Subjects' Fo was measured with a Laryngograph and PDP-12 computer to the nearest 0.01 ms. (about 1 Hz) and measurements were made of the time from the target change until a directed change towards target frequency began and the time until the change was completed. The rate of Fo change during that interval was also measured.

RESULTS AND DISCUSSION

Despite only having a moderate amount of prior practice (about 10 trials), subjects performed quite reliably and accurately. The two main findings which characterize the subjects' ability to respond to sudden changes in target frequency are:

A) subjects reacted more quickly when the target was in the right ear; and

B) they completed the task more quickly when the cursor was in the right ear.

The latter finding corroborates the cursor/right ear advantage of earlier pursuit auditory teaching studies and confirms the hypothesis that a dynamic target is necessary to elicit an REA. The two findings together indicate that the advantage of the right ear/left hemisphere pathway results from the superiority of this pathway for auditory-speech sensorimotor integration rather than any other type of auditory processing.

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(2) Hanson, R.J. and MacNeilage, P.F., J. Acous. Soc. Amer. (1975) 55, S31-32.
ESTIMATION OF PITCH CONTOURS WITH THE AID OF DIGITAL FILTERS

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INTRODUCTION

During voiced segments of speech the vocal tract is excited by a nearly periodic pulse train. The result is a nearly periodic response at the output of the vocal tract. For a lot of applications the estimation of the time varying excitation periods and frequencies is of interest. Most of the numerous pitch estimation methods known till now [e. g. 1, 2] can be classified according to the following scheme: (1) frequency domain approach (2) time domain approach, (3) deconvolution approach. The first two methods are characterized by moderate expense but low accuracy whereas the deconvolution methods yield good results (although a certain degree of smoothing is inherent) at the cost of high expense. A recently proposed method [3] which combines the advantages (moderate expense) and avoids the disadvantages (low accuracy) of the first two methods is used as basis for the further proceeding.

PRINCIPLE OF PITCH ESTIMATION

The basic principle as shown in fig. 1 is derived from a peak picker preceded by a digital filter with the following properties: (1) multiple integrating magnitude response ensuring that all frequency components above the fundamental frequency are attenuated by a factor of $\nu^{-n}$, where $\nu$ is the ratio of a higher frequency component to the fundamental frequency and $n$ is the multiplicity of the integrating filter, (2) low group delay distortion minimizes pitch estimation errors caused by alterations of the fundamental frequency itself, (3) stability under operating conditions and optimum transient behaviour in the sense that rapid transients in the speech signal do not excite the filter's eigen-resonances too much. Detailed investigations [4] have shown that a stable recursive digital integrator whose transient behaviour is corrected by a notch filter at zero frequency with the transfer function

$$H(z) = \frac{\frac{1}{2}((z+1)(z-1))}{((z-1+\varepsilon_1)(z-1+\varepsilon_2))} \right)^n, \quad \varepsilon_1, \varepsilon_2 > 0,$$

fulfills the abovementioned specifications. A good compromise between the frequency response, stability requirement and transient behaviour is achieved for the following filter parameters: $n = 4$, $\varepsilon_1 = 0.01$, $\varepsilon_2 = 0.03$, sampling frequency $f_s = 16$ kHz. Great values of $\varepsilon_2$ lead to further improvement of the transient behaviour which can be utilized for pitch frequencies related to female and filial speakers. The pitch estimator's excellent performance has been tested with synthetic speech signals of known pitch and with natural speech.

REFERENCES

Speech signals can be modified in duration and fundamental frequency by analysis and resynthesis. The parameters (e.g., pitch and predictor coefficients) extracted from the original speech signal are modified (by "hand" or rule) according to the desired change. The quality of the resynthesized speech is, of course, limited by the distortions of the analysis-synthesis system employed.

These problems are avoided by our method in which the speech signal is inversely filtered and modifications are performed on the resulting "excitation" signal. In order to modify the excitation signal, it is divided into "working intervals" which correspond approximately to pitch periods during voiced speech segments. To effect pitch changes, each of these intervals is either shortened by cutting out a segment of low energy or stretched by inserting zeros at the point of lowest energy. The change in length of working intervals causes a change of duration of the whole speech signal which has to be corrected for. Moreover, an independent change of duration is often desired. Thus, in either case, a time adaption of the signal is necessary which can be realized by repetition or omission of entire "working intervals". For the resynthesis it is useful to label each sample of the "excitation" signal with the number of the inverse filter under which it was generated. This number remains attached to the sample during the whole process. In this manner the proper synthesis filter is used at all times.

The process proposed here is distinguished by high reliability (e.g., no voiced-unvoiced decisions are necessary) and modifications of pitch and duration are effected with excellent-speech quality.
CONTRIBUTION OF PITCH TO THE PERCEPTION OF SEGMENTAL QUALITY

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In a previous experiment (1) I demonstrated that synthetic vowels bearing a changing fundamental frequency are perceived as being longer than vowels of equal duration synthesized on a monotone. The present experiment was designed to test whether the effect of lengthening produced by changing $F_0$ is carried over into the perception of voicing in the final consonant of English monosyllabic words. It is well known that lengthening of the syllable nucleus suffices to signal that the consonant following the syllable nucleus is voiced; if perceptual lengthening is produced by changing pitch, syllable nuclei of the same duration may be perceived as being followed by a voiced consonant when the syllable nucleus is produced with pitch inflection, and as being followed by a voiceless consonant when produced on a monotone.

METHOD

Test stimuli were produced on the OVE synthesizer at Haskins Laboratories. The extreme duration values for one set of stimuli represented the word pair bad-bat, the second set corresponded to bead-beat. There were ten durations for each set, ranging in 24 msec steps from 396 msec to 180 msec. The initial and final consonants were synthesized by formant transitions alone; the final consonant was not released. Lengthening and shortening took place during the steady state of the vowel. All stimuli in both sets were synthesized on a monotone at 80 Hz and with a $F_0$ falling from 80 to 60 Hz. A randomized test tape was prepared, on which each stimulus occurred twice. The tape of 80 items was presented to 25 subjects, whose task was to identify the stimulus as either bad or bat in one case or as either bead or beat in the second case.

RESULTS AND DISCUSSION

The results of the listening test are presented in Table 1 for the bad-bat pair. Similar results were obtained with the bead-beat set. The perception was near-categorical in all cases. The crossover from the perception of voiced final consonant to the perception of voiceless final consonant occurred between stimuli 5 and 6 in both monotone sets; in the case of falling $F_0$, the crossover took place between stimuli 8 and 9 in the bad-bat pair and between stimuli 7 and 8 in the bead-beat pair. The difference between the results obtained with monotone stimuli and with falling pitch is significant at better than .005 level in both sets. There appears to exist no reason why changing fundamental frequency should contribute directly to the perception of voicing in the final consonant. Two steps are postulated to explain the observed relationship. The change in pitch produces the impression of greater length; greater length produces the impression of voicing in the final consonant.

ACKNOWLEDGMENT

The author expresses her appreciation to Dr. Linda Shockey, who produced the test stimuli.

REFERENCES

APPLICATION OF DISSIMILARITY AND APERIODICITY FUNCTIONS TO FUNDAMENTAL FREQUENCY MEASURE OF SPEECH AND VOICED-INVOICED DECISION

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INTRODUCTION

In this paper the theoretical study shown in an accompanying paper (1) is applied to process signals of a specific nature: speech. The quasiperiodicity of each segment has been measured and the segment was accordingly classified as voiced or unvoiced. In the first case the quasiperiod was estimated. The intensity is also computed, in any case.

THEORY

The speech signals are included in all the normed linear spaces described in (1), and also the numerical sequences (or "discrete" signals) obtained after sampling and quantifying the former signals.

Taking \( p=1 \) and being \( w \) the rectangular window (both chosen for its computing simplicity) the autodissimilarity function of a sequence \( f_n \) becomes:

\[
\text{ads}_m = \frac{\|f_{n-m'} - f_{n+m'}\|_p^\beta - \beta \|f_{n-m'}\|_p^\beta - \|f_{n+m'}\|_p^\beta}{\|f_{n-m'}\|_p^\beta + \|f_{n+m'}\|_p^\beta}
\]

where \( \|f_n\|_p = (1/2\ell' + 1) \sum |f_{n+i}| \), \( m' \) and \( \ell' \) are the integral halves of \( m \) and \( \ell = k|m| \). Note that the window length \( \ell \) is made proportional to the shift \( m \). See in (1) the obtainment of \( a_n, T_n, I_n \).

EXPERIMENTAL

Speech signals (see fig. 1) were sampled at 10 Kc, quantified at 64 levels and stored in the disk of the Speech System of the SES(CEA), Saclay, (France), through a minicomputer Multi-20. A program was written to process up to 1.1 seg of signals loaded in memory.

A sample of this process can be seen at the graphs: \( \text{ads}_m \) (with \( \beta = 0 \)) is computed at regular intervals of \( n \) (see in fig. 2, the \( \text{ads}_m \) in the points \( P \) and \( Q \) of the signal). According to the described -in (1)- strategy, \( T_n, A_n, I_n \) are obtained. They are displayed in fig. 3. The error marked with an \( \times \), and other details will be discussed at the ICA Congress. Also will be commented the relationship of this method with others.

REFERENCE

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THE DAWID SPEECH RECOGNITION SYSTEM

The DAWID speech recognition system at the present stage works as a syntax-guided isolated word recognition system. DAWID is built up as a hybrid-system consisting of an hardware analyzer and a software classifier (Device for Automatic Word Identifikation by Discrimination). After the hardware feature extraction the varying combinations of the indicated features are interpreted according to the algorithms of an ensuing software classification. The regularities of this classification have been developed on the basis of heuristic explorations of the behavior of the acoustic analyzers (Bierfert 1975, unpublished).

From each incoming speech signal a string of "DAWID-symbols" (i.e.: word differentiating elements) is derived, and each word of the vocabulary is represented by one or several strings. The matching process is based on the identity of any reference string with the actual string of each spoken word. The DAWID-symbols can - in a wider sense - be phonetically interpreted (Glave, Lancé, to be published). With respect to this relation a preselection of the application vocabulary is possible.

PERFORMANCE EVALUATION

The system performance was tested in a first step by 5000 spoken word items (25 sentences of four different syntactical categories, 88 different words, five male speakers) (Glave, to be published). The system was tuned up by 10 male speakers (10000 utterances).

The recognition accuracy is described in terms of the "mean identification performance" consisting of the "identification index" \( B \), and the "discrimination index" \( \eta \) (\( B \) is identical with the frequently used terms "recognition score", "success rate"). These indices range between 83 % and 93 %. The rejection rate was separately classified. In the mean the proportion of rejections is 7.9 %. According to the Spearman rank correlation coefficient there is no (or slight) association between the set of error scores and of rejection scores.

Similar results could be obtained with an increased vocabulary, namely of 103 german verbs (Lancé 1976, unpublished).

A more sophisticated recognition strategy is intended to operationalize the training phase of the system so that no speaker-dependent tuning up is required (Glave, to be published).

According to this schedule the system would be able to build up a sufficiently representative reference pattern from the lexical (orthographic or allophonic) information.
INTRODUCTION

Undoubtedly the use of synthetizers during the past years has contributed to attain further knowledge for speech studies. We decided to start with the synthesis of the vowels because they are the easiest sounds to obtain and the ones that allow a better control of their reproduction quality.

THEORETICAL BASES

The acoustic characteristics of the vowels we used as start point of this synthesis work were taken from the work GUIRAO-MANRIQUE "Identification of Argentine Spanish Vowels", Journal of Psycholinguistic Research, Vol. 4, n° 1, 1975. On the basis of the acoustic characteristics proposed in this work, we introduced variations in the formants values, its band width and intensity and we varied the glottal tone and its extent until we found the spectral values which seemed to be more effective.

EXPERIMENTAL TECHNIQUES

To synthetize the spanish vowels we used a synthetizer simulated in a digital on-line computer and peripherical equipment specially designed to perform this type of research. The peripheric equipment consisted of a teletype, a perforator and reading, four transporters of magnetic tape, an analogous-digital converter, three digital-analogous converting and a tube of cathodic rays. The software included a system on line, on line editor program, a macroassembler, a FORTRAN IV compiler, several transfers lines and a debugging on-line system. Work was performed with a total of 120 vocalic sounds. First we synthetized ten groups of vowels, in five groups we used six different values of F1 and F2 for each vowel and in the other five to the six values mentioned we added the average value of F3 for each vowel. Four identification tests were prepared with a total presentation of 272 stimuli. These test were generated in the computer and recorded in a professional recorder. To present the sounds we used a high fidelity audio system which reproduced them at a comfortable level in a sound proof room.

RESULTS AND CONCLUSIONS

With the parameters of the vowels choosen we obtain a 100% correct identification of the sounds. We believed that other values can be found within those areas that can produce an identification percentage as high as the one we found, taking into account the considerable distance between the areas of the Spanish vowels.
III. Psycho and physiological acoustics

J. BIOACOUSTICS
The methods of actography and bioelectrical skin reactivity / BSR /, developed in our laboratories were used to investigate the annoyance of the natural sleep by noise of 25 experimental subjects. Ss were selected by psychodiagnostic methods and their properties of nervous system were identified by measures of BSR.

The results showed the relations between the exposure to the noise of different levels and annoyance of natural sleep as so as the inter- and intraindividual differences and the hygienic limits of noise in buildings and flats. It was also found that the BSR is more sensitive indicator of annoyance of the sleep by noise in compare to actography and EEG method.
UNTERSUCHUNGEN ÜBER DAS DURCH DEN ÜBERSCHALL HERVORGERUFENE AUFSCHRECKVERHALTEN

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THEORETISCHE BASIS
Es wird von der Annahme ausgegangen, dass Erschrecken dann eintritt, wenn der Informationsfluss größer ist als die Kanalkapazität der biologischen "Nachrichtensenke". Das bedeutet: Erschrecken tritt dann auf, wenn einem Individuum mehr Information dargeboten wird, als dieser verarbeiten (erkennen) kann.

Als wesentliche Parameter ergeben sich dabei:
b) Die Wahrscheinlichkeit (P) des eintreffens eines bestimmten Reizmusters. Mit einer steigenden Wahrscheinlichkeit, wird das Erschrecken abnehmen.

EXPERIMENTIERTECHNIK

Während den Versuchen wurden drei Versuchspersonen gleichzeitig beschallt. Pro Versuchssequenz (Dauer 48 Minuten) wurden 12 Knalle dargeboten.

RESULTATE
Die Versuche ergaben folgende Resultate:
a) bei einer ersten Darbietung eines Knalls erschrecken sämtliche Versuchspersonen, unabhängig von dessen Spitzendruck und Anstiegszeit.
b) nach einer grösseren Anzahl von Darbietungen, stoebt die Häufigkeit des Erschreckens einem Konstanten Wert zu, welcher nur noch von den physikalischen Parametern des Knalls abhängt.

Diese beiden Ergebnisse zeigen die Schwierigkeit auf, die sich ergibt, wenn Normen über die belastbarkeit in Bezug auf das Erschrecken gefordert werden. Denn, tritt ein Knallphänomen nur sehr selten auf, so wird sicher ein sehr grosser Teil der Bevölkerung, schon bei relativ geringen Lautstärken erschrecken, während dieser Anteil bei häufiger Belastung wesentlich geringer sein wird.

REFERENZ
INTRODUCTION:

A study has been made on the hearing acuity of the tribals of the hilly areas of South India as these people are residing in comparatively noise free zones. These tribals are good hunters and they do so with the help of their sharp ears and eyes. Our investigation was motivated by the desire to compare their hearing levels with those of people living in urban areas.

THEORETICAL BASES:

For the first time in 1962 Dr. Rosen of United States pointed out that Mabaans of Sudan, a closed community in North Africa were noticed to possess hearing levels different from the conventionally expected ones. There was in fact very little disparity in hearing acuity between the young and old people amongst them and this has been pointed out by Dr. Rosen as being the result of freedom from Noise Pollution. Our tribals have a similar noise free environment like the Mabaans and hence it is we decided to investigate their hearing sensitivity for comparison.

EXPERIMENTAL TECHNIQUES:

The assessment was done in the interior of the forest. Environmental noise levels were measured with sound pressure level meter with 1/3 Octave band filter set up. The instrument was calibrated using piston phone. Tripod stand and wind screen were used at necessity. The condenser microphone used in the SPL meter for this study was one inch in diameter having a frequency response of 10 Hz to 20,000 Hz with 1 dB variation.

The hearing testing was done using Arphi portable Audiometer calibrated to ISO 1964 standards.

The representative population of the tribes Todas, Kotas, Irulas and Kurumbas who were free from otological and other illness was selected and threshold audimetry was administered in a less noisy place of known ambient noise level.

The hearing levels obtained are compared with that of the normals in urban areas (Kameswaran 1976, to be published) belonging to the same age group. The results are analysed and discussed.

RESULTS AND CONCLUSIONS:

Tribal population of Nilgiris (Blue mountain) live in different altitudes in the same hill range and do not intermix with each other. They retain their ethnic identity and live exclusively staying away from the rest of the world. Even the older age groups amongst them have hearing within normal limits. Interestingly enough we find no significant difference between the hearing levels of the different tribes living at varying altitudes. Also the tribes possess significantly better hearing than the urban population of the same age group.
INTRODUCTION: In order to investigate the dynamic range of the neural response to cosine noise (noise plus the same noise delayed having a cosinoidal power spectrum) and the frequency selectivity of the units of the Dorsal Cochlear Nucleus (DCN) of the cat we recorded their activity at several intensity levels of the noise and at different modulation depths of its power spectrum.

METHODS: In the present experiment (set-up described in 1,2) the spike rate of a unit was registered as a function of the delay time $\tau$ (between the noise and the delayed noise), which was swept linearly from zero to about $20/cF$ ($cF$ being the characteristic frequency of the unit). Before starting a registration ($\tau$-diagram), the neuron was adapted to the specific intensity level of the continuous noise.

RESULTS: Data are presented in fig.1 from a pauser with $cF=9$ kHz with a non-monotonic spike-intensity relation to continuous pure tone stimulation and one high-frequency inhibitory sideband. The spike rate of the neuron in response to cosine noise is plotted against the delay time $\tau$ at two intensity levels $L$. $g$ is relative level of the delayed noise with respect to original noise. Two facts are visible:

a) For large $\tau$ the spike rate depends on intensity level $L$. From similar recordings at intensity levels over 100 dB the slope of the spike-intensity relation measured remains .9 sp/dB/sec. This value is small in comparison to 7 sp/dB/sec of the neural response to noise over 40 dB in the non-adaptive state.

b) The number $n$ of peaks decreases with increasing intensity level $L$. (see also fig.2).

DISCUSSION: As this number $n$ (comparable to a spectral harmonic number) is a quality index for the frequency resolving power of the auditory periphery and for the internal preservation of the cosine comb spectrum (2), a deterioration of the internal spectrum with increasing intensity is evident. This is in agreement with the widening of iso-intensity curves (e.g.3). Nevertheless, the modulation in the internal spectrum is still discernible at the highest intensity levels, even for values of $g$ down to -20 dB. This result, and the constant spike-intensity relation in a 100 dB-range (see also 4), indicate that, although eight-nerve units have a limited dynamic range, the spectral structure of a complex sound and its intensity are well coded in the DCN over a large intensity range.

REFERENCES:
2) F.A. Bilsen et al J.A.S.A.(1975)58,4,858
EINTEILUNG

Dieser Artikel schlägt eine bioakustische Methode für Quantitätsbewertung der isolierten und kombinierten Lärm- und Schwingungseinwirkung auf den Menschen vor.

GRUNDPRINZIPIEN


THEORETISCHE RESULTATE

Für die akustische $W_a$ und mechanische $W_m$ Energie bei isolierten Lärm- und Schwingungseinwirkungen kann man die folgenden Formeln bekommen:

$$W_a = \int_0^T \int_0^T \frac{1}{2} \left( L(f) + L_m(f) \right) \cdot \frac{Z_o}{Z_m} \cdot \frac{Z_m}{S} \cdot \frac{1}{S} \cdot dt \cdot df,$$

wo $f$ - die Frequenz; $t$ - die Zeit; $T$ - die Einwirkungsperiode; $Z_o$ - die akustische Eingangsimpedanz der Membrane des Mittelohres; $Z_m$ - die mechanische Eingangsimpedanz des Menschkorps an der Kontaktstelle mit der Quelle; $U$ - die mittlere quadratische Bedeutung der Schwingungsgeschwindigkeit in gleichen Punkt; $L(f)$ - die Frequenzspektrum des gemessenen Schalldruckpegels (in dB); $L_m(f)$ - die Korrektur (in dB), die die Schalldiffraktion auf den Schalldämpfer und die Schallenergieumwandlung bei ihrem Durchgang in außerhalbkanal ablegt; $S$ - die Membranfläche. Die Funktionen $K(f)$ und $L_m(f)$ sind von Experimentsergebnissen abhängig.

Die Energie $W_g$ der gesamtwirkenden Lärm und Schwingungen auf das biologische Objekt wird mit folgender Formel gegeben:

$$W_g = W_a + \left( \frac{W^{(n)}_a}{W^{(n)}_g} \right) W_2 + \left( \frac{W^{(n)}_m}{W^{(n)}_g} \right) W_3,$$

wo $W_a$ und $W_g$ - die Energie der lokalen Schwingungen und Körperseinschwingungen; $W^{(n)}_g$, $W^{(n)}_m$, $W^{(n)}_a$ haben die analogische Bedeutung bei den Lärm- und Schwingungspanparametern, die gleich der Normativwerten des betrachteten Systems sind.
INTRODUCTION

Laboratory experiments were conducted to investigate the process of adaptation to a high intensity industrial noise and the individual's behavioral consequences due to the exposure to noise.

THEORETICAL BASES

The effort entailed in adapting to aversive events may be achieved at some expense to the individual's behavioral functioning. As a consequence deleterious aftereffects may be the direct result of cumulative experience of stress, despite the fact that the individual has adapted to the aversive event. It is possible to expect task degradations and other impairments of behavior following cessation of the stressor. (1)

The present study is based on the research of Glass, Singer et al.

EXPERIMENTAL TECHNIQUES

Subjects were 120 university students divided into two samples: Experimental and Control. Each experiment was individual. In the first part of the experiment both samples of subjects performed same task during 60 minutes, and while the Control Sample worked in normal environmental conditions, the Experimental Sample worked under the influence of an industrial noise of 90 dBA with peaks of 95 dBA.

In the second part of the experiment both samples were divided into three groups, each of which did a different task under normal environmental conditions.

The purpose of the first part of the experiment was to investigate the possible differences of the mental performance of the two studied samples in a task requiring concentration.

The purpose of the second part was to investigate the possible aftereffects to exposure to a high intensity noise in the following aspects:

a) The capacity for tolerance of frustration.

b) The capacity for abstract reasoning tasks.

RESULTS AND CONCLUSIONS

The collected data ratifies the hypothesis used to design the survey and seems to have proved that high intensity noises would have as consequence a delay in the higher logical processes of thinking and a lesser capacity for tolerance for frustration.

REFERENCES

INTRODUCTION
The injurious effects of noise and vibration have become an international problem of utmost importance.

THEORETICAL BASES
In contrast to the rather extensive research that has been done on the separate non-specific effects of noise and vibration, their effects in combination have received less attention. As noise has become ubiquitous and the use of machines generating vibration increases, changes in the organism for their interaction are expected to be greater. This study was aimed to investigate certain parts of the metabolism of rats liver, heart, and suprarenals.

EXPERIMENTAL TECHNIQUES
Forty albino male rats were divided into 4 groups as follows: group ISt - exposed to combined noise (105 dB SPL) and vibration (100 Hz ampl. 0.1 min.) action, group IIInd - exposed only to vibration, group IIIrd - exposed only to noise and group IVth - non treated animals, which served as a control. The exposure lasted 2 hours daily for 10 days. The animals were decapitated immediately after the last exposure. Homogenates from liver, heart and suprarenals were prepared using physiological solution (1:10). In the supernatant (after centrifugation) the activity of the following enzymes was determined: GLDH, EC. 1.4.1.2; SDH, EC. 1.3.99.1; LDH, EC. 1.1.1.27; MDH, EC. 1.1.1.37; iCDH, EC. 1.1.1.49; GL6PDH, EC. 1.1.1.49; Cyto, EC. 1.9.3.1.; ATP, EC. 3.6.1.3.; AP and AcP, EC. 3.1.3.1. and EC. 3.1.3.2.; Gl-6-P-ase, EC. 3.1.3.9. as well as the quantity of -SH groups and the soluble proteins in the homogenates.

RESULTS AND CONCLUSIONS
As far as the changes in the activity of GLDH, SDH, iCDH, AP, AcP and G6P-ase in the myocardium, AP in the liver and AcP and G6P-ase in the suprarenals are concerned a definite synergistic effect of the two factors was observed. The effect of combined noise and vibration action on the activity of MDH and the quantity of SP in the heart, iCDH, ATP-ase and Cyto in the suprarenals and ATP-ase and the quantity of the SP in the liver could be classified as an additive one. An antagonistic effect of the factors was observed when comparing the effect of noise and vibration on the activity of GLDH, SDH, LDH, GL6PDH and the quantity of SP in the suprarenals, as well as SH in the liver and heart, and Cyto in the heart. Predominantly a prevalence of the effect of noise was observed. On the basis of the experimental data indicative for a definite disturbances in the metabolism of the organs investigated one could assume that under the effect of noise and vibration certain reduction of the adaptation ability and reliability of the physiological functions might occur. It could be taken into consideration when estimating noise and vibration effect.
UNE MODELISATION DE LA PROPAGATION DES ONDES ACOUSTIQUES DANS LES MILIEUX BIOLOGIQUES

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Les techniques ultrasonores d’investigation des milieux biologiques (échographie, vélocimétrie sanguine) posent un problème fondamental d’acoustique du fait de l’importance de phénomènes habituellement négligés, tels l’absorption et la dispersion. La nature essentiellement longitudinale des vibrations ultrasonores dans les milieux biologiques, ainsi que la constitution essentiellement aqueuse de ces derniers, induisent un modèle grossier de fluide de Navier-Stokes. Cette même nature ainsi que la relative lenteur des mouvements et écoulements biologiques, autorisent une hypothèse de quasi-homogénéité et de quasi-stationnarité du milieu vis à vis des phénomènes acoustiques.

Une méthode classique de perturbation permet d’obtenir une équation décrivant la propagation avec amortissement et effet de convection par le milieu, de la forme:

\[ \left\{ \begin{array}{l}
[dt - KA] \left[- \frac{1}{c^2} dtt + (1 + K \frac{PN + 1}{c^2} dt) \Delta \right] - (\gamma - 1) (PN - 1) \frac{k^2}{c^2} dt \Delta \right\} \delta p = 0
\]

où \( \delta p \) est la pression acoustique, \( c \) la célérité adiabatique, \( K \) la diffusivité thermique, \( P \) le nombre de Prandt, \( N \) le nombre de viscosité, et \( dt = \frac{\partial}{\partial t} + V \cdot \text{grad} \), la dérivative convective.

En fait, pour les milieux biologiques, l’absorption reste faible, et on peut prendre une équation locale de type:

\[ \left[ - \frac{1}{c^2} dtt + (1 + K \frac{PN + 1}{c^2} dt) \Delta \right] \delta p = 0 \]

qui est très proche du d’Alemberien classique ; équation qui demeure valide pour les signaux à large bande qu’on peut être amenés à utiliser.

Pour des signaux à bande étroite, on aboutit alors à une formulation voisine de l'opérateur de Helmholtz :

\[ \left\{ k^2 + [1 + i \frac{PN + 1}{c^2} - \frac{1}{c^2}] \Delta \right\} \delta p = 0 \]

où \( k \) est le nombre d’onde, \( s \) le nombre de stokes.

Le cas d’ondes planes-monochromatiques conduit à des relations d’absorption et de dispersion faisant intervenir à la fois la nature du milieu et son mouvement.

L’étude débouche sur un résultat original qui concerne en fait l’acoustique aérodynamique : on montre, en effet que l’absorption réelle \( \alpha \) et l’absorption au repos \( \alpha_0 \) sont reliées par l’expression:

\[ \alpha = \alpha_0 (1 + m)^{-3} \]

où \( m \) est le nombre de Mach de l’écoulement.
Taking into consideration the numerous industrial workers and drivers who are prone to hazards related to alcoholic states, we have investigated the synergistic effect of noise and alcohol in an experimental test system.

Our test is based on the swimming performance of mice. This test gives a good evaluation of their fatigue resistance.

To separate the auditory and non auditory effects of sound and infrasounds, three strains of mice were used, one of these being totally deaf.

An aqueous solution (10 % v/v) of ethanol was given orally two hours prior to the swimming test at a dose of 0.15 ml per mouse which does not affect the swimming time. During these two hours mice were exposed to sinusoidal frequencies from 6 to 10 000 Hz with different S.P.L. varying from 60 to 118 dB.

At threshold sound intensities, which do not decrease swimming time, the combination of noise and alcohol provoked an additive effect for high frequencies in normal hearing mice. At frequencies below 50 Hz for all strains of mice, we found a significant synergical interaction between ethanol and sound.

Since this effect was obtained with infrasounds on both deaf and mice with normal hearing, it is not the auditory system alone that is involved. We have confirmed these experimental results with complex noise such as that existing inside a typical french automobile running on a highway.

These results lead to a more precise evaluation of the noxious effect of alcohol on individuals already exposed to the stress of a noisy environment.
EXPERIMENTAL EFFECTS OF NOISE ON PREGNANT MICE AND ON THEIR OFFSPRINGS

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The purpose of this work was to determine whether there was an effect of a noise, or a noise associated with another stress, either on the mother, or on the offspring in mice, and whether an eventual effect was due to a stress response of the mother or of the young themselves.

EXPERIMENTAL TECHNIQUES

To this effect different groups of mice were subjected either to
a) four hours a day, 7 days/week, recorded subway noise (105 dB) or
b) during two of these four hours, the mice were also shaken on a vibrating table.

Duration of experiment: 6 litters or about 6 months to one year.

Two strains of mice were used:
1. normal hearing albino (Rb).
2. Qq deaf mutant (dn/dn) from GFF strain, were coupled with hearing hybrid (dn/+). The offspring are thus made of about 1/2 deaf and 1/2 hearing pups.

RESULTS AND CONCLUSIONS

On all experimental groups we were unable to find any differences between controls and treated animals for the following physiological data: weight of the mothers, number of young at birth and at weaning, sex ratio of the offspring.

Differences on the other hand were found in particular on the following measures:
A. Mean weight gain of litters from birth to weaning: (decrease of 25 to 30 % in treated pups) this however only for the 3 first litters of hearing mothers (Rb).

Since the young continue to be subjected to the noise until weaning, it could have two implications: a) that a stressed pregnant mother influences the weight of the young even well after birth. Or, that the influence is due to a poorer parental care during lactation. b) that habituation renders the situation less stressful. (30 % decrease at the 1st litter, 25 % for the 2nd, 20 % for the 3rd).

These differences are no greater when sound is associated with shaking.

B. The interval between litters are quite regular in the control groups and very irregular in the treated ones, and the number of possible miscarriages or uterine resorptions is far greater.

C. Foetuses. The 6th litters of each group were delivered by cesarean.

Eventual cannibalism was checked by comparison with prior litter size and no difference in number of foetuses was found.

The number of malformations, on the other hand (mostly cranial and spinal), is much greater among offspring of treated mothers than untreated ones. (6 % for controls, 15 % sound alone, 30 % sound + shaking for the Rb strain and respectively 7, 20 and 37 % in the deaf strain).

Researches are under course to elucidate the divergence in results between pups weights and malformations. Hybrid hearing (dn+/+) mothers are being coupled with deaf fathers. The type of pups should be similar to those of previous deaf mothers and hybrid fathers, but mothers should be more noise stressable.

The difference between these groups should give us an interesting outlook towards understanding the mechanism of action of noise on pregnant mice.
In an attempt to find an acoustic substitute for chemical insecticides in stored flour, the effect of high intensity sound waves was studied on various insect pests (Ephestia, Sitophilus). Stimuli applied for 15 seconds were:

1. High intensity low frequency sound (10-30 Hz at 135 dB) produced by a 46 cm diameter loudspeaker in a 1 m³ sound proof box.
2. 40 kHz pure tone produced by a Lansing compression chamber (Type 375) with a paraboloid lens giving 155 dB at the focal point.

High intensity sound waves had no effect on any of the Coleoptera or Lepidoptera eggs, larvae or imago, even though the plastic boxes containing insects and flour were in some instances shattered to pieces. Longer exposures obviously would not be feasible for industrial use.
INTRODUCTION

L'effet lésionnel des différents paramètres physiques qui caractérisent un bruit impulsif (bruit d'arme à profil triangulaire produit en champ libre) a été étudié chez le cobaye.

TECHNIQUES EXPERIMENTALES

Les techniques audiométriques utilisées ont été:
- l'observation du seuil d'apparition du réflexe de PREYER,
- l'électrocochléographie par voie externe.(I).

RESULTATS

Les déficits auditifs (TTS), exprimés en décibels, augmentent comme $40 \log_1 \Delta P$ ($\Delta P$: surpression de crête) lorsqu'ils sont mesurés un jour après l'exposition, et comme $20 \log_1 \Delta P$ lorsque l'audiométrie intervient quatorze jours après l'exposition.

Les déficits auditifs augmentent comme $20 \log_1 N$ ($N$: nombre d'expositions) et sont maximaux lorsque la durée de l'intervalle entre deux expositions consécutives est de 3 secondes. Lorsque la durée est inférieure à 3 secondes, l'action du réflexe de protection acoustique diminue l'amplitude des pertes auditives. Pour des durées supérieures à 3 secondes il semble qu'il existe une certaine récupération entre deux expositions consécutives.(2).

L'étude de l'influence de la durée de la première phase positive du bruit impulsif (A-duration), n'a pu mettre en évidence une proportionnalité entre déficits auditifs et énergie sonore. Ce résultat qui va à l'encontre de l'application du principe d'isoénergie aux bruits impulsifs du type étudié, a pu trouver un commencement d'explication grâce à l'enregistrement de pressions acoustiques dans les rampes cochléaires. On a constaté notamment un renforcement de la pression acoustique dans la rampe vestibulaire du tour basal pour une durée de la première phase positive de l'onde de 60 microsecondes. Cette observation pourrait expliquer le fait que l'amplitude des déficits auditifs observée chez un groupe de cobayes exposés à un nombre donné de bruits impulsifs de 45 microsecondes (A-duration) était supérieure à celle observée chez un autre groupe qui avait été exposé au même nombre de bruits mais d'une durée de première phase positive (A-duration) de 1000 microsecondes et ceci bien que l'énergie acoustique contenue dans les bruits de 1000 microsecondes soit 20 fois plus importante que celle contenue dans les bruits de 45 microsecondes.

La récupération des déficits auditifs, exprimés en décibels, s'effectue selon $10 \log_1 T$ ($T$: temps de récupération).

REFERENCES

INTRODUCTION
Les déplacements de la chaîne tympano-ossiculaire produisent des variations de pression acoustique dans les liquides cochléaires. Ces variations de pression, agissant de part et d'autre de la cloison cochléaire, provoquent le déplacement de cette dernière.

L'enregistrement de la pression acoustique dans la rampe tympanique et dans la rampe vestibulaire, des différents tours de la cochlée, permet de mieux comprendre la mécanique cochléaire ainsi que la genèse des traumatismes sonores.

Les premiers enregistrements de ce type ont été réalisés par BURGEAT (1) chez le cobaye et, plus récemment, par NEDZELNITSKY chez le chat (2).

Nos mesures ont été faites chez le cobaye soit dans la rampe vestibulaire du premier, deuxième et troisième tour, soit dans la rampe tympanique du premier tour.

Les résultats obtenus furent comparés à ceux fournis par l'enregistrement du potentiel microphonique cochléaire par électrodes différentielles dans les différents tours cochléaires.

TECHNIQUES EXPERIMENTALES
Les capteurs de pression utilisés étaient du type piézo-résistif, de diamètre environ 1 mm et étaient munis d'une sonde remplie d'huile de paraffine. Le diamètre de l'extrémité de la sonde était de 0,5 mm, sa longueur de 1,2 mm et son volume intérieur était inférieur à 1 mm3. Les capteurs étaient implantés de façon étanche dans la cochlée, la bulle tympanique refermée et l'animal placé sous respiration artificielle après curarisation.

Les stimuli acoustiques utilisés furent soit des bruits impulsifs.

RESULTATS
Lorsque le stimulus est un son pur, on observe dans la rampe vestibulaire une pression de 30 dB supérieure à celle du stimulus extérieur pour les fréquences comprises entre 6 et 8 kHz. L'amplitude de la pression intracochléaire augmente de 6 dB/octave jusqu'à 1 kHz avec un déphasage de 90 degrés par rapport au stimulus extérieur.

La linéarité semble conservée lorsque le niveau de stimulation passe de 75 à 120 décibels.

L'allure des courbes de réponse est semblable à celle du potentiel microphonique cochléaire enregistré au niveau du premier tour par électrodes différentielles.

Lorsque le stimulus est un bruit impulsif, on n'observe pas de latence significative d'apparition de la pression dans les tours supérieurs. Cette constatation peut être faite également au cours de l'étude du potentiel microphonique des tours 2 et 3.

CONCLUSION
Les résultats obtenus ont permis la mise en évidence d'une onde de compression longitudinale qui précède la propagation de la traveling wave et ont permis de mieux comprendre la genèse des traumatismes sonores produits par l'exposition aux bruits impulsifs.

REFERENCES
NON LINEARITE COCHLEAIRE CHEZ LE COBAYE ET TAUX DE VARIATION DE L'INTENSITE ACOUSTIQUE

Burgeat, M. 
Loth, D. 
Grall, Y. 
Menguy, C. 
Toupet, M.

Des électrodes mises en place dans la cochlée du cobaye permettent d’enregistrer la réponse de l’oreille interne à une stimulation acoustique.

On peut ainsi étudier les variations de l’amplitude du Potentiel Microphonique Cochléaire (PMC) en fonction de l’intensité du son stimulant. Grâce à ces enregistrements, on a depuis longtemps mis en évidence l’existence de phénomènes non linéaires dans les réponses électrophysiologiques de la cochlée (Hystérésis cochléaire HUGSON & WITTING 1935).

Lorsqu’on étudie l’amplitude du PMC pour un son d’une intensité donnée, on constate que celle-ci diminue transitoirement, immédiatement après l’émission d’un son de forte intensité.

Les auteurs proposent que la période de récupération d’un PMC après un son fort, soit sous la dépendance de 2 phénomènes :
- l’un, diminuant l’amplitude de la réponse, qui pourrait être, au moins en partie, dû à un phénomène de fatigue (Fatigue Cochléaire),
- l’autre, tendant au contraire à augmenter l’amplitude de la réponse (Sensibilisation Cochléaire Post-stimulatoire).

Ces 2 phénomènes, qui paraissent évoluer séparément en fonction du temps, entrent en concurrence, l’un pouvant, dans certaines conditions expérimentales, prédominer momentanément sur l’autre.

Afin de préciser l’influence du facteur "temps" dans l’apparition respective de ces phénomènes, les auteurs utilisent un stimulateur à sons forts dont la vitesse de variation de niveaux peut être programmée.

Grâce à ce stimulateur et en traitant les résultats sur un Ordinateur DEC PDP-12, il a été possible de mettre en évidence plusieurs types de variation d’amplitude du PMC en fonction du taux de variation de l’intensité du son de stimulation.

Voitages du signal acoustique (s.a) du signal électrique (s.e) et du PMC en fonction du temps et de la variation exponentielle de la pression acoustique de 70 dB à 120 dB et de 120 dB à 70 dB.

REFERENCES :
INTRODUCTION

Les processus physiologiques déterminés par l'action des ultrasons sur les cellules sont des phénomènes très complexes. La littérature de spécialité traitant de l'action des ultrasons sur les suspensions cellulaires, quoi que assez richement représentée, est loin de l'élucidation et de la résolution satisfaisante de tous les problèmes en la matière. La hémolyse des érythrocytes dans un champ ultrasonique crée la possibilité d'effectuer des déterminations précises dans ce domaine. Nous avons employé comme matériel expérimental des érythrocytes de carpe (Cyprinus carpio), de grenouille (Rana temporaria), de tortue (Emys orbicularis), de poulet (Gallus domesticus) et de rat (Rattus rattus).

PARTIE EXPÉRIMENTALE

Nous avons réparti des suspensions d'érythrocytes à 1 ml, ayant une concentration de 1%, dans des récipients d'expérimentation et nous les avons soumis à l'action du champ ultrasonique à intensité (0,2 - 1,2 W/cm²) variant leurs durées de traitement, après quoi nous avons déterminé le degré de hémolyse en employant la méthode photocolorimétrique, par dosage de la quantité de hémoglobine (1).

RESULTATS ET CONCLUSIONS

La figure ci-contre (Nr.1) relève les durées du traitement ultrasonique nécessitées par la hémolyse, en fonction de l'intensité du champ. Ce qu'on remarque c'est la grande résistance à la hémolyse des érythrocytes de poulet, la valeur de l'intensité du champ ultrasonique y étant de 1,2 W/cm², après 60 secondes de radiation. Les autres intensités, au-dessous des valeurs minima de cavitation (valeurs minima représentées par les intensités 0,6 W/cm² pour les érythrocytes de tortue et celle de 0,2 W/cm² pour ceux de grenouille), ne sont pas suffisamment élevées pour provoquer la destruction des érythrocytes. A cette même occasion, on peut relever la faible résistance des érythrocytes de carpe et de rat.

Un des paramètres acoustiques pouvant influer sur la résistance des érythrocytes en champ ultrasonique est l'impédance acoustique spécifique. Les érythrocytes à impédance acoustique spécifique plus élevée (le cas de Gallus) se montrent être plus résistants à l'action des ultrasons, ce qui n'est pas le cas pour les érythrocytes à basse impédance acoustique spécifique.

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INTRODUCTION

This paper will discuss a method by which focused, pulsed, ultrasound can be used to detect and to monitor continuously microparticles in a flowing fluid.

EXPERIMENTAL TECHNIQUES

The system uses a commercially available reflectoscope whose output is connected directly to a PDP 11/40 DELLAB Computer. The system has been calibrated to detect microparticles in the size range of 10 to 400 microns in diameter. The calibration studies were done using glass beads, gas bubbles, and 3-M microspheres. The unique feature of the system is the design and construction of the plexiglass ultrasonic chamber. The chamber is designed such that the flat 5-MHz transducer is positioned externally to the fluid flow. The curvature of the cylindrical chamber acts as a plane-concave lens to focus the beam at a distance 1.36 cm. The focal point is very sharp and well defined and has been mapped with a 0.625 cm steel reflecting ball.

RESULTS

The chamber has been applied experimentally to such flowing fluids as distilled water, plasma, plasma with packed and washed red blood cells, and to various extracorporeal circuits used in open heart surgery procedures. It has been found in the last two groups of experiments that even the minimum length extracorporeal circuit has an irreducible background count with the average size range between 35 to 160 microns in diameter. The identity of this background, which is characteristic of the biological system, has not yet been determined. Current investigations are directed toward the identification of the nature of the particles responsible for the background.
INVESTIGATIONS ON ALBINO MICE IN INFRASOUND FIELD

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INTRODUCTION

Beside some effects of infrasound on man e.g. the similarity to the effect of alcohol /1/, there are a lot of references showing conflicting facts in this matter because of its subjectivity and complexity.

A big problem in carrying out investigations by means of infrasound of high levels, especially measuring the hearing threshold, is the distortion /2,3/ which may call in question whether the effect of the infrasound or of its harmonics have been observed.

We intended to make experiments with small animals exposed to really pure tones under 20 Hz with the aim to observe their behaviour in infrasound of high level and long duration.

EXPERIMENTAL TECHNIQUES

According to our earlier works /4/ we produced infrasound of 5.7 Hz in a Helmholtz-type resonator of 20 litres. This was our infrasonic chamber. The distortion in audible frequency range was measured by special technique using A-weighted and highpass of 22.5 Hz filters, and condenser microphon Brüel-Kjaer type 4132 with the pressure equalization tube opened. The measured results according to this way are shown in Figure 1.

RESULTS AND CONCLUSIONS

Investigations on albino mice in the above mentioned infrasonic chamber have been carried out by different methods. We tried to observe the behaviour of the mice first in infrasonic field switched on and off periodically, and later on, giving a task for them to choose between being in infrasound of high level or silence.

The main conclusions are:
- the sleep of the albino mice seems to be disturbed by the infrasound energy at level of about 150 dB;
- they choose the silence rather than the infrasound but after some days we could observe the opposite behaviour, too.

The measurements are to be continued in order to confirm our preliminary results and to get closer informations about the behaviour of the animals depending on the infrasonic level and frequency.

REFERENCES

When the wavelength of a sound is greater than the dimensions of the body, there results an approximately uniform pressure acting over the surfaces of the body. The pressure gives a compressional force, which may cause resonant vibrations controlled by the stiffness of the structure and effective mass of the body parts involved. The most prominent resonance is of the chest at frequencies between 30 Hz and 70 Hz, depending on body stature. Other resonant effects have been noted at lower frequencies in the stomach and calf.

Measurements were made in a low frequency noise chamber with the subject standing on a vibration isolation platform. Detection of resonances was by an accelerometer attached to an elastic belt which was strapped round the body whilst the subject was exposed to a frequency sweep from 10 Hz to 100 Hz, of duration about four minutes, at constant sound levels in the range 100 dB to 107 dB. The magnitude of the resonance peak was also noted at levels from 90 dB to 110 dB. The inherent chest vibrations were recorded in the absence of external sound in order to give a background vibration level.

Fourteen male subjects and ten female subjects took part in the investigations although only six males and four females were included in the measurements over the range of levels. There were marked differences in the response of male and female subjects, females showing non linear effects. A typical chest resonance is given in Fig. 1 and typical level dependence in Fig. 2. Repetition with a subject breathing a 90% Helium - 10% Oxygen mixture did not change the chest resonant frequency, indicating that the effect is dependent on structure, rather than on an air cavity in the chest.

A simple model considering the system as a hollow cylinder gives a stiffness at resonance of $1 \sim 1.5 \times 10^6 \text{Nm}^{-1}$ with a quality factor of $2.5 \sim 3.5$, females generally having lower Q factors than males. The proportion of body weight involved in the chest vibration is $15 \sim 20\%$ which agrees with anatomical data.

Chest vibrations were also detected in subjects exposed to traffic noise, especially in the presence of heavy vehicles.
Un modèle mécanique du cœur permettant la description du mécanisme d'émission des bruits cardiaques, fournit une relation qualitative entre le phonocardiogramme et les fluctuations de la pression ventriculaire gauche (P.V.G.) : l'accélération de la paroi thoracique est égale au produit de la (P.V.G.) par une fonction du temps dépendant des propriétés mécaniques du muscle. Cette relation s'écrit :

\[ \text{Phono}(t) = K_i \left| \text{P.V.G.}(t) - \text{P.V.G.}(i-1) \right| + \text{Phono}(i-1) \]

A chaque indice \( i \) (\( i = 1, 2, ..., 6 \)) correspond une phase de la révolution cardiaque (remplissage, contraction pure, contraction avec éjection, éjection pure, éjection avec relaxation, relaxation pure). Par phono \( (i-1) \) et P.V.G. \( (i-1) \) on désigne les valeurs des fonctions phono et P.V.G. à la fin de la \( (i-1) \)ième phase. Les paramètres \( K_i \) sont identifiés en minimisant la norme au carré de la différence entre la fonction d'intercorrélation de l'accélération thoracique théorique donnée par (1) et du phonocardiogramme expérimental, et la fonction d'auto-corrélation de ce dernier signal. Ce procédé, qui donne certains renseignements sur les propriétés mécaniques du muscle cardiaque, permet en outre de justifier le modèle théorique proposé : le phono théorique est en effet en très bon accord avec le phono expérimental (voir figures 1 et 2).
Vibrations influence the well-being of man. We have good vibrations and bad vibrations. The baby is comforted and enjoys the movement of the cradle, the rythmic movements when carried, and many enjoy the more violent movements in dancing. - Vibrations are as associated with life as sound.

The measurement of vibrations in the frequency range and dynamic range of interest for whole-body and hand-arm vibrations presents special transducer and instrumentation problems.

The transducer shall give a correct presentation of the vibratory signal at the point of entrance into the body in a well defined direction. The instrumentation shall analyze the signal in both the time and frequency domain.

The use of a special transducer and the result of a statistical analysis of a ride in a car will be reported relating the statistical results obtained to the exposure levels standardized in ISO 2631, "A Guide for the Evaluation of Human Exposure to Whole-body Vibration". The standard prescribes the use of one third-octave filter analysis. However, the use of a frequency weighted band is also taken into account. The exposure, in a practical working situation, requires the use of a statistical analysis technique. The combination of a weighting curve and a statistical analyzer makes it possible to apply the standardized exposure levels to practical work situations.

In order to test the exposure experienced over a long trip by car, a standard automobile was equipped with a portable set of instruments as shown.

A seat transducer using B&K delta shear elements was cast into a neoprene rubber disc to obtain good coupling to the buttocks. A whole-body weighing filter was included. An integration time constant of 1000 m/sec. was used and a sampling rate of 0.5 sec. The 7200 samples obtained during each hour were read out onto a Thermal Printer 2312 and the data shown obtained during 12 x 1 hour periods driving from Copenhagen, Denmark, to Aosta, Italy.
GEFÄHRDUNGSFAKTOREN DES OPERATEURS BEI DER EINSCHÄTZUNG VON AKUSTISCHEN UND VIBRATIONSKRAFTWIRKUNGEN

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Die Gefährdungsfaktoren, die den Charakter der Anpassungsfähigkeit des Operateurs an akustischen und Vibrations-Kraftbelastungen bestimmen, sind individuelle typologische Eigenschaften des Organismus.


BIBLIOGRAPHIE


EVALUATION DE LA NOCIVITE DES BRUITS IMPULSIFS PAR COMPARAISON, SUR ANIMAUX, DE LEURS EFFETS AVEC CEUX DE BRUITS STATIONNAIRES

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INTRODUCTION

à l'heure actuelle, il n'existe pas encore de mode d'évaluation de la nocivité des bruits impulsifs faisant l'unanimité sur le plan international. Les critères dont on dispose vont d'une simple majoration arbitraire du niveau sonore à des évaluations plus élaborées obtenues à partir d'oscillogrammes, tels les critères de COLES et RICE, de PFANDER ou encore à partir de l'énergie sonore. En tout état de cause, les résultats varient selon le critère utilisé, d'où la nécessité de rechercher à valider un critère simple convenant, au moins dans un premier temps, aux bruits industriels.

PRINCIPAUX AXES DE RECHERCHE

Les principaux axes de recherche utilisés à ce jour sont les suivants :
- Enquêtes épidémiologiques cherchant à relier la surdité observée chez des travailleurs avec la nature du bruit auquel ils sont exposés.
- Études basées sur la fatigue auditive (TTS).
- Études utilisant l'expérimentation sur animaux.

CHOIX DE LA MÉTHODE

La méthode décrite ici est basée sur l'hypothèse suivante : "Deux bruits, l'un stationnaire (dont on connaît assez bien les effets sur l'homme), l'autre étant celui à étudier, qui produisent à long terme la même surdité sur l'animal, seront jugés identiques du point de vue nocivité".

Les animaux sont des cobayes, de souche DUNKIN-HARTLEY, utilisés par groupes de 10 pour obtenir une bonne information statistique. Les animaux sont des cobayes, de souche DUNKIN-HARTLEY, utilisés par groupes de 10 pour obtenir une bonne information statistique.

Le bruit est généré par quatre baffles à trois voies de forte puissance et haute fidélité, alimentés par un générateur de bruit rose pour les bruits stationnaires par des enregistreurs magnétiques longue durée, lui également, pour les bruits industriels à étudier.

Les audiogrammes sont réalisés en utilisant le réflexe de MEYER selon une technique rigoureuse mise au point par DALCER à l'Institut franco-allemand de St Louis.

DEROULEMENT DES ESSAIS

L'exposition au bruit et à la lumière reproduisent le rythme journalier d'un travailleur-type. Les premiers résultats d'essais laissent apparaître un défi permanente après 2 semaines d'exposition à un bruit rose de 100 dB(A).

L'expérimentation prévoit le tracé de l'évolution de la perte auditive à différents niveaux de bruit rose, à titre de référence. Viendront ensuite les essais avec des bruits industriels de plus en plus complexes : bruit de tissage à navettes batantes, de presses à découper rapides, de pilons...

CONCLUSION

L'expérimentation décrite ici n'en est qu'à ses débuts à l'heure où est rédigé ce texte. On peut espérer que quelques résultats seront déjà dis plus en plus complexes : bruit de tissage à navettes batantes, de presses à découper rapides, de pilons...

Conclusion

L'expérimentation décrite ici n'en est qu'à ses débuts à l'heure où est rédigé ce texte. On peut espérer que quelques résultats seront déjà dis plus en plus complexes : bruit de tissage à navettes batantes, de presses à découper rapides, de pilons...

* Outre les données déjà disponibles, une enquête est en cours à l'I.N.R.S. portant sur 10 000 audiogrammes et les relevés sonométriques correspondants.

** Ce chiffre pourra être modifié ultérieurement.
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