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Institut für Technische Akustik
der Rheinisch-Westfälischen
Techn. Hochschule
Aachen
FOREWORD


The publications of the Congress include over 350 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
# Contents

## Volume I

<table>
<thead>
<tr>
<th>Environmental acoustics</th>
<th>page</th>
</tr>
</thead>
<tbody>
<tr>
<td>A. Acoustic criteria</td>
<td>1</td>
</tr>
<tr>
<td>B. Design and planning</td>
<td>35</td>
</tr>
<tr>
<td>C. Building acoustics</td>
<td>77</td>
</tr>
<tr>
<td>D. Acoustic Materials</td>
<td>129</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Noise and vibration</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>E. Noise sources</td>
<td>149</td>
</tr>
<tr>
<td>F. Noise control</td>
<td>219</td>
</tr>
<tr>
<td>G. Shock and vibration</td>
<td>289</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Psycho and physiological acoustics</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>H. Hearing</td>
<td>323</td>
</tr>
<tr>
<td>I. Speech and communication</td>
<td>405</td>
</tr>
<tr>
<td>J. Bioacoustics</td>
<td>527</td>
</tr>
</tbody>
</table>

## Volume II

<table>
<thead>
<tr>
<th>Physical acoustics</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>K. Ultrasound</td>
<td>551</td>
</tr>
<tr>
<td>L. Underwater acoustics</td>
<td>633</td>
</tr>
<tr>
<td>M. Molecular acoustics</td>
<td>677</td>
</tr>
<tr>
<td>N. Sound propagation</td>
<td>707</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Electroacoustics</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>O. Sound systems</td>
<td>759</td>
</tr>
<tr>
<td>P. Musical acoustics</td>
<td>781</td>
</tr>
<tr>
<td>Q. Instrumentation</td>
<td>807</td>
</tr>
<tr>
<td>R. Signal processing</td>
<td>845</td>
</tr>
</tbody>
</table>

Barcelona Symposium            | 867  |
Sevilla Symposium               | 875  |
ICA - SCOPE Working Groups     | 863  |
IV. Physical acoustics

K. ULTRASOUND
The objective of this work was to develop realistic models for quantitative flaw characterization using ultrasonic techniques. The flaws were introduced in diffusion bonded titanium alloy disks of ellipsoidal and disk-shaped cavities. The samples were immersed in water. A normally incident broadband (about 6 MHz wide with a center frequency of 5 MHz) longitudinal pulse after interaction with the cavity scattered and mode converted. The scattered longitudinal and shear waves were received by a second transducer and analyzed by a spectrum analyzer. The variation of power spectrum of the scattered waves were analyzed as a function of shape, size, and orientation and compared to two existing theories: (1) Born approximation [1] for the region of $ka \leq 1$ (where $k$ is the wave number and $a$ is the radius of the cavity), and (2) geometrical theory of diffraction introduced by Keller [2,3] for $ka \geq 1$. Typical experimental results on Figures 1 and 2 compare reasonably well with the corresponding theories.

Fig. 1. Power spectrum of an elastic wave scattered from an oblate spheroidal cavity in titanium.

Fig. 2. Power spectrum of an elastic wave scattered from an elliptical cavity in titanium.

REFERENCES

1. J. Gubernatis, 1975 Ultrasonics Symposium Proceedings, IEEE Cat. #75, CH0 994-45U.

*This work was supported by the Center for Advanced NDE operated by the Science Center, Rockwell International, for the Advanced Research Projects Agency and the Air Force Materials Laboratory under Contract F33615-74-C-5180.
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INTRODUCTION

The object of the present study is to analyze the angular dependence of the scattered field when a plane wave is incident upon an ellipsoidal cavity in an elastic medium. The present study has particular relevance in non-destructive evaluation of flaws by ultrasonic methods.

SOLUTION AND RESULTS

It is shown that when a plane wave given by

\[ u_0 = u e^{i k_0 r} \quad \text{or} \quad d_0 = 0 \]

is incident on an ellipsoidal cavity then the scattered field is given by

\[ u(s) = u_0 \sum_{n=0}^{\infty} \sum_{m=-n}^{n} [A_{mn} L(3) + \tau_{mn} M(3)] e^{i[k_0(n+1)+m]r} \]

where \( k = k_0 \), \( c \) being a characteristic length of the cavity. Using this result we have shown in the accompanying figure the relative backscattered power for ellipsoidal cavities of various aspect ratios.

INCIDENT LONGITUDINAL WAVE
Dans le cadre de l'étude de la diffusion des ondes ultrasonores par différents types de cibles, nous nous intéressons plus spécialement à la rétrodiffusion par les surfaces rugueuses présentant un profil aléatoire ou périodique.

Le transducteur émet des impulsions ultrasonores à bande étroite de fréquence centrale 2 ou 5 mégahertz. Nous mesurons, en fonction de l'angle d'incidence $0^\circ < \theta < 27^\circ$ du faisceau ultrasonore sur la surface, les variations de l'intensité rétrodiffusée $I(\theta)$. Le dispositif expérimental et le principe des mesures ont été décrits dans de précédents articles (1) (2). Les diagrammes de rétrodiffusion que nous obtenons expérimentalement pour un transducteur, une distance palpeur-obstacle et une fréquence de travail données dépendent spécifiquement de la rugosité $h$ (valeur quadratique moyenne) de la surface diffusante dans le cas de profils aléatoires. Une méthode d'interpolation nous permet d'obtenir une évaluation de la valeur de $h$ pour des surfaces aléatoirement rugueuses de profil inaccessible ou inconnu. Les premiers échantillons étudiés étaient de natures physiques assez diverses : plaques d'aluminium, plaques de dural, tôles épaisses recouvertes de peintures soumises ou non à la corrosion, monocristaux de tellure. Leur rugosité $h$ était comprise entre 5 et 100 microns et nous avons montré que l'évaluation indirecte par voie ultrasonore de $h$ présentait une précision de un micron pour les valeurs inférieures à 40 microns et de quelques microns pour les valeurs supérieures. Le déroulement du processus expérimental menant à cette évaluation a été rendu totalement automatique.

Dans le cas où le profil de surface possède une composante périodique les diagrammes de rétrodiffusion présentent des oscillations et il est généralement possible d'en déduire la valeur de la périodicité spatiale du diffuseur.

Nous présentons ici nos plus récents résultats dans ce domaine et plus particulièrement ceux obtenus dans le cas d'échantillon réfléchissant peu les ondes ultrasonores : éponges naturelles ou artificielles, tissus végétaux ou animaux. Nous montrons que la méthode appliquée précédemment avec succès au cas d'obstacles ayant un plus grand pouvoir réflecteur (métaux) donne aussi des résultats satisfaisants quelque moins précis pour ces différents types de diffuseurs. Il faut noter que les méthodes conventionnelles de mesure des profils de surface sont inapplicables à ces matériaux.

Nous présentons également quelques résultats préliminaires concernant, d'une part, la diffusion des ultrasons par la surface rugueuse d'une cavité interne à la cible, d'autre part, les modèles théoriques applicables à ce genre d'études.

(1) de Billy, 8ème Conférence Mondiale sur les Essais Non Destructifs, 1976, Communication 3 F 11.
(2) de Billy, Sonics and Ultrasonics (1976), SU 23, Nb 25, p. 356.

* Laboratoire associé au C.N.R.S.
FREQUENCY DEPENDENT ULTRASONIC REFLECTIVITY FROM PLATES IN A LIQUID

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INTRODUCTION

Recent theoretical investigations (1-2) of the reflectivity of a bounded ultrasonic beam from a solid plate in a liquid indicate that the nonspecular reflected beam effects are a function of the frequency, \( f \), of the incident ultrasound. This suggests that for a given product \( fd \) (frequency x plate thickness) one would expect to see different reflected beam profiles depending on the particular values of \( f \) and \( d \).

Furthermore, most of the experimental work on the so-called "beam displacement" has been done at small \( fd \) values, \( fd < 14 \) MHz.mm. This paper shows the nonspecular effects and their dependence on \( f \) for glass plates at \( fd = 35.5 \) MHz.mm.

THEORETICAL CONSIDERATIONS AND EXPERIMENTAL RESULTS

Nonspecular effects occur at angles of incidence, \( \theta_1 \), corresponding to Lamb angles of the plate. Lamb angles are related to the product \( fd \). Observation and profile of the nonspecular effects depend (1-3) on the value of a parameter, \( h \) defined as

\[
h = \left( \text{Im} \sin \theta_1 \right) \left( \frac{2\pi f W}{V} \right)
\]

where \( \text{Im} \sin \theta_1 \) — imaginary part of the pole location of the reflection coefficient; \( V \) — sonic velocity in the liquid; \( W = W/\cos \theta_1 \), and \( W \) is half the incident beam width.

Therefore, for a particular plate mode (fixed \( fd \) and Lamb angle) the reflected beam profile changes with \( f \) provided \( h \) is large to make the effects observable.

Schlieren techniques are used to observe the nonspecular reflected beam effects from glass plates in water at Lamb angle incidence.

Results (Figs. 1 and 2) clearly show that the nonspecular effects are present at relatively large values of \( fd \), and they change markedly with frequency.

Fig.1. \( fd = 35.5 \) MHz.mm; \( f = 2.9 \) MHz.  
Sound is incident from the left side

Fig.2. \( fd = 35.5 \) MHz.mm; \( f = 16 \) MHz.  
Sound is incident from the left side.

Acknowledgment: Supported in part by Phys. Dept. Georgetown University, U.S.A.

REFERENCES

(1) T.J. Plona, nop et al., JASA (1976) 59, 1324; (2) L.E. Pitts, nop et al., JASA (1976) 69, 374; (3) H.L. Bertoni and T. Tamir, Appl. Phys. (1973) 2, 157
INTRODUCTION

When an ultrasonic pulse is reflected from a plate, the pulse shape is deformed as a result of multiple reflection in the plate.(1)

In this paper, calculations of ultrasonic pulse reflected from a quartz transducer and its output waveforms are described. Also reported are reflection and transmission of ultrasonic pulse through a plate in water.

THEORETICAL BASES

The reflected pressure is calculated by superposing the impulse response of a quartz or a plate immersed in water. The incident pressure is assumed to be written in the form

\[ P = 4 \left( 1 - e^{-at} \right) e^{-at} \sin at \]

where \( a/\omega = 1/8m \) is taken.

The velocity at water-quartz interface is given by subtracting the reflected pressure from the incident pressure. The vibrational velocity of the quartz is proportional to the output current, whose envelope approximates the voltage envelope.

RESULTS

Some of the results for normal incidence are shown in figs(1-3). The similar calculation procedure is applicable for a laterally bound pulse obliquely incident on a plate. Fig(4) represents the reradiation from Lamb wave excited by the incident pulse. These results were confirmed by experiments.

REFERENCE

(1) H.E.Altman and R.T.Beyer, JASA (1976) 59, 545

Fig.1 Pressure waveforms reflected between two quartz transducers which are air-backed and confronted in water. Points are at \( \sin at = 1 \) in eq(1).

Fig.2 Velocity (or output) waveforms of the quartz corresponding to fig(1).

Fig.3 Reflection (R) and transmission (T) through Al plates of various thicknesses in water.

Fig.4 Reflection and transmission of 4.6 MHz pulse through an Al plate (1 mm thick) in water, \( \theta = 21^\circ \)
INTRODUCTION

The reflection sound field has been studied experimentally for very thin air-filled spheres in water, using disc radiators for transmission and very small PZT cylinders for reception. The pulse lengths used were of the order of 5 μsec and the cylinder diameters were roughly from 3 to 6 inches. The thinnest sphere studied has a shell thickness of only 1/100 inch. Both spheres and cylinders of other shell thickness-to-sphere diameter ratio have been studied for comparison, both with the thin sphere results and those of other experimenters. Experimental frequencies are one and two MHz, so that the ka ratios are fairly high. In the case of cylinders, low ka ratios were obtained at these two frequencies by the use of cylinders of 1 in., and ½ in. diameters.

PURPOSE

A major concern of the investigation was the explanation of the variation of the amplitude of the primary reflection with angle. The general behavior was found to agree with the predictions of specular reflection, probably because of the large ka values in most of the experimental cases, but marked differences were found between the very thinnest shells and those of somewhat greater thickness (though still quite thin). Materials used for the investigation were brass, aluminum and glass (blown-glass spheres were found to have exceedingly fine sphericity, variations in the outer diameter being commonly of the order of less than one part in 5000, as measured with a precision thickness gauge.)

EXPERIMENTAL

The experiments were performed in a tank 12 ft. long and 10 ft. wide, with the depth of the water about 1½ ft. The sides of the tank were slanted at an angle of about 45 degrees to minimize reflections. Because of the sharpness of the sound beam and the geometry of the emitter-scatterer-receiver combination protection was provided from any direct surface and bottom reflections. Later surface, bottom and side reflections were present but they could easily be time-separated from the signal under observation.

RESULTS

Direct evidence of circumferential wave trains was found in the case of aluminum cylinders, as in the experiment of K. J. Diercks, T. G. Goldsberry, and C. W. Horton,(1) but we found no similar result for very thin spheres. D. J. Shirley and K. J. Diercks,(2) have found such a train of echoes in spheres for particular ka ratios, but in an earlier work, K. J. Diercks and R. Hickling,(3) in a very similar experiment, found no such echo train. In any case, the contribution of circumferential waves to the scattering pattern must be considered. This was noted in the case of Diercks, Goldsberry and Horton for cylindrical shells, and there is no reason to suppose that the same is not true of spherical shells. Indirect evidence is found in the scattering pattern.

REFERENCES

La spectroscopie ultrasonore, appliquée aux essais non destructifs, peut fournir, dans certains cas, des informations quantitatives précieuses (géométrie des défauts, épaisseur et qualité des joints soudés ou collés...), mais une interprétation précise des spectres ultrasonores nécessite l'utilisation de modèles simplifiés. Nous présentons ici les résultats de l'étude par analyse spectrale des échos ultrasonores diffusés par des réseaux de même pas $A$ mais de profils différents.

Dans l'approximation de Kirchhoff l'amplitude de l'onde rétrodiffusée, pour une onde incidente plane, peut s'écrire

$$ A(f, \theta) = G_1(\theta) \int_{-L}^{+L} \exp i (k_1 - k_2) \cdot \hat{r} \, dx $$

où $k_1$ et $k_2$ sont respectivement les vecteurs d'onde de l'onde incidente et de l'onde diffusée, $\theta$ l'angle d'incidence et $L$ la longueur du réseau. Dans le cas où l'approximation basse-fréquence est applicable, on obtient pour la raie de diffraction d'ordre $m$

$$ A_m(\theta) = G_2(\theta) m F_m(\theta) $$

où $F_m$ est le coefficient d'ordre $m$ du développement en série de Fourier du profil d'un pas. Pour deux réseaux ($i$ et $j$) de même pas $A$ mais de profils différents, le rapport des intensités des pics de diffraction s'écrit alors simplement :

$$ \frac{I_{mi}(\theta)}{I_{mj}(\theta)} = \frac{m^2 |F_{mi}(\theta)|^2}{n^2 |F_{mj}(\theta)|^2} $$

A l'aide d'un appareillage classique en spectroscopie ultrasonore nous avons étudié l'intensité rétrodiffusée par des réseaux de pas $A = 400 \, \mu m$ dont la distance crête à vallée des traits variait selon l'échantillonnement de $4 \, \mu m$ à $20 \, \mu m$. La bande de fréquence ultrasonore analysée s'étend de 2 à 9 mégahertz. Dans ces cas l'approximation B.F. est limitée aux fortes incidences ($\theta > 60^\circ$) et aux deux premiers ordres. Nous avons représenté sur la figure ci-contre les résultats expérimentaux en fonction des résultats théoriques pour les ordres $m = 1$ (+) et $m = 2$ (x). Les valeurs théoriques sont déduites de la relation (3) où les coefficients de Fourier $F_{ij}(\theta)$ ont été remplacés par les transformées de Fourier discrètes des profils de surface relevés mécaniquement avec un intervalle d'échantillonnage de $20 \, \mu m$. L'un des échantillons ($j$) a servi de référence et ce sont les rapports $\frac{I_{mi}(\theta)}{I_{mj}(\theta)}$ qui sont portés en décibels. La droite de régression a une pente de 1,1 et la corrélation entre valeurs expérimentales et théoriques est de 0,96. Nous étudions également la dépendance de l'intensité des raies en fonction de l'angle $\theta$.  

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9 INTERNATIONAL CONGRESS ON ACOUSTICS
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SPECTROSCOPIE ULTRASONORE APPLIQUEE A L'ETUDE DES SURFACES RUGUEUSES PERIODIQUES

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PHASE TRANSFORMATIONS IN LIQUID BY RECTIFIED HEAT TRANSFER IN SOUND FIELD

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INTRODUCTION

Sound field influences on the phase transformations in liquid that are cavitation, boiling, and crystallization.

THEORETICAL BASES

The general theory that describes sound field effects on the formation of vapour bubbles in the liquid is given in ref. (1-2). By analogy the theory of sound field effects on the formation of solid spherical particles in the liquid may be developed.

RESULTS AND CONCLUSIONS

If the pressure in the liquid is changed during the time as $P(t) = P_o + P_m \exp(i\omega t)$ than the linear solution for the radius of sphere of new phase may be taken in the form $R(t) = R(t) + R_m \exp(i\omega t)$. The displacement amplitude $R_m$ is equal to

$$R_m = -P_m R(t)K/3Q,$$

where $K$ is the compressibility of the new phase sphere, $Q(R, \omega)$ is the resonant factor, $P_o$ is the static pressure in liquid, $P_m$ is the pressure amplitude of sound of the frequency $\omega/2\pi$. The average radius $R(t)$ is determined at the quadratic approximation

$$\frac{dR}{dt} = C \left( \pm \int \frac{P^2}{Q^2} + \Delta T \right)$$

where $J(R, \omega)$ is the function that defines the rectified heat transfer into the sphere due to acting of the sound field, $C$ is certain constant, connected with the parameters of liquid, $\Delta T$ is the superheating for the boiling liquid or the supercooling for the solidifying liquid. On the right hand of eq. (2) the upper sign is corresponded to the solid particles dynamics in liquid.

From eq. (2) we have the existence of the threshold sound amplitude for given vapour bubble radius, above which the bubbles grow. On the other hand for given sound amplitude there are two critical radii determining the region of the bubble sizes, within which bubbles grow in underheated liquids.

By analogy in the case of crystallization of supercooled liquid from eq. (2) we have the existence of the critical sound amplitude and the critical sizes of the melting solid particles.

REFERENCES

(2) В. Н. Алексеев. Акустический журнал (1975) 21, 497; (1976) 22, 185.
ABSORPTION ULTRASONORE DANS LES MÉLANGES AQUEUX D'ALCOOLS

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INTRODUCTION

Le maximum d'absorption ultrasonore en fonction de la concentration dans les mélanges aqueux d'alcools a été étudié dans la gamme de fréquence 1-50 MHz (Sette 1949, Burton 1947...). Des développements plus récents ont permis d'étendre la fréquence d'étude jusqu'à 230 MHz (Blandamer 1970, Nishikawa 1976), et 500 MHz pour l'éthanol (Takagi 1975), mettant ainsi en évidence des processus à plusieurs temps de relaxation.

TECHNIQUES EXPERIMENTALES

L'étude présentée constitue grâce à l'utilisation d'un appareillage récent une extension du domaine de fréquence jusqu'à 1000 MHz. Deux interféromètres à deux transducteurs excités par des impulsions électriques ont été utilisés : le premier (de 5 à 75 MHz) est constitué de deux quartz de fréquence nominale 5 MHz ; le second (de 140 à 1000 MHz), de deux cristaux de Niobate de Lithium dont le déplacement relatif (sensibilité 0,1 μm) permet de mesurer à 3% près l'absorption pour des liquides dont l'atténuation est comprise entre 200 et 2000 dB/mm. La stabilité de température est assurée à 0,1°C près.

RESULTATS

L'absorption des mélanges aqueux de méthanol, d'éthanol, d'isopropanol et de n-propanol a été étudiée à deux températures différentes (0 et 15°C) et à diverses concentrations. Pour les 3 derniers alcools cette extension en fréquence a permis de déterminer avec précision la variation des fréquences de relaxation en fonction de la concentration à deux températures.

Les courbes d'absorption obtenues peuvent être décrites par l'expression suivante comportant deux fréquences de relaxation :

\[ \frac{\alpha}{f^2} = B + \frac{A_1}{1 + (\frac{f}{f_1})^2} + \frac{A_2}{1 + (\frac{f}{f_2})^2} \]

Les fréquences de relaxation \( f_1, f_2 \) passent par un minimum en fonction de la concentration alors que les paramètres \( A_1, A_2 \) présentent un maximum. Les valeurs de \( B \) sont supérieures à celles de \( \langle \alpha/f^2 \rangle \) classique et présentent un maximum pour une concentration supérieure à celle pour laquelle \( \alpha/f^2 \) passe par un maximum.

D'autre part il a été mis en évidence un processus de relaxation dans l'éthanol et les propanols purs : la précision des mesures sur l'absorption permet de situer la fréquence moyenne de relaxation à une valeur légèrement supérieure à 1000 MHz donc très inférieure à la fréquence de relaxation de viscosité.

Nous sommes reconnaisants à Monsieur le Professeur John Lamb de nous avoir accueillis dans son laboratoire permettant ainsi la réalisation de cette étude.

REFERENCES

MODIFICATIONS OF TRANSPORT PHENOMENA IN ULTRASONIC FIELD

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INTRODUCTION

The literature data (1,2) concerning electrical conductibility modifications in an ultrasonic field are sporadic and not systematized, the various authors giving different explanations, often contradictory, of the observed modifications. That is why a systematic study of the problem was initiated.

EXPERIMENTAL TECHNIQUES

By means of a Radelkis GK 102 conductometer the conductibility of aqueous solutions was measured, for Sr, Ba, Ca, Mg chlorides, for Na, Mg, Cu sulphates and for manite \( \text{CH}_2\text{OH - (CHOH)}_4 - \text{CH}_2\text{OH} \), an undissociable polyalcohol.

The measurements were carried out before and immediately after exposure to the ultrasounds. The variation of the conductivity (\( \gamma \)) and the equivalent conductibility (\( \Lambda \)) were observed as a function of concentration, temperature, ultrasonic intensity and irradiation time. The limits of the concentration range (high dilutions) were imposed by the precision of the instrument. The irradiation time ranged between 10 - 60 minutes, the intensity between 110 - 230 mA plate current and the temperature varied from 20°C to 40°C. The relaxation effect was also investigated, during 300 hours time.

RESULTS AND CONCLUSIONS

A conductibility increase was noted, after ultrasound irradiation: approximately independent of concentration for sulphates and slightly decreasing with increased concentration for chlorides. The intensity and the exposure time determined an increase of the \( \gamma \) (\( \Lambda \)) values, while temperature resulted in their reduction. The relaxation, more rapid at the beginning, remained incomplete. It is suggested that the conductibility modifications are caused by the action of ultrasounds on the solvent, due to the apparition of supplementary ions (\( \text{H}^+ \), \( \text{OH}^- \), \( \text{H}_2\text{O}^+ \)) and of hydrogen peroxide. The solute itself is implied only in the alteration of the solvent structure for the existing experimental conditions. A modification of the nature and concentration of solved gases (\( \text{O}_2 \), \( \text{CO}_2 \), \( \text{H}_2\text{CO}_3 \)) should also be taken into consideration.

REFERENCES

INTRODUCTION

Le coefficient de viscosité est un paramètre qui permet de faire la caractérisation d'une émulsion du point de vue de sa concentration et de son degré de dispersion.

La dépendance de la viscosité de l'émulsion diluée du rapport existent entre le volume des particules et le volume total de l'émulsion, conformément à la relation : \( \eta = \eta_0 (1 + 2,5 \phi) \), où : \( \eta \) la viscosité de l'émulsion, \( \eta_0 \) la viscosité de la phase externe, \( \phi \) le rapport entre le volume des particules et le volume de l'émulsion, peut nous renseigner sur les méthodes et les dimensions des particules. Vu que les méthodes visque-métriques sont peu précises, il y a la possibilité de corrélation de la viscosité de l'émulsion avec l'absorption de l'ultrason dans l'émulsion étudiée.

THÉORIE

Conformément à la théorie classique, l'absorption due à la viscosité du milieu est exprimée par la formule :

\[ \alpha_{vis} = \frac{6}{5} \cdot \frac{\eta^2 \nu^2}{\rho c^3} \]

où : \( \rho \) la densité du milieu, \( c \) la vitesse de propagation et \( \nu \) la fréquence des ultrasons. Cette interdépendance permet l'emploi d'une méthode ayant à sa base la mesure du coefficient d'absorption et qui facilite l'observation des émulsions, dans le temps.

La variation, dans le temps, du coefficient d'absorption dans l'émulsion étudiée renseigne sur la structure du milieu d'absorption.

RÉSULTATS ET CONCLUSIONS

Nous avons effectué le mesurage du coefficient d'absorption de l'ultrason dans les émulsions à la fréquence 10 MHz. Les données relatives à la variation du coefficient d'absorption, dans le temps, pour l'émulsion de 1,5 % huile de tournesol en eau et respectivement de 4 % huile de transformateur en eau, sont reflétées par le tableau ci-dessous :

<table>
<thead>
<tr>
<th>Emulsion</th>
<th>Conc.</th>
<th>( \alpha/1^2 )</th>
<th>( g^2/cm )</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>initial</td>
<td>après 24h</td>
</tr>
<tr>
<td>Huile de transformateur en eau</td>
<td>4</td>
<td>51</td>
<td>46</td>
</tr>
<tr>
<td>Huile de tournesol en eau</td>
<td>1,5</td>
<td>54</td>
<td>47,4</td>
</tr>
</tbody>
</table>

On remarque une baisse du coefficient d'absorption dans le temps, ce qui témoigne de la baisse de la viscosité des émulsions. Les données obtenues par les mesurages d'absorption renseignent donc sur la qualité et la stabilité des émulsions.

BIBLIOGRAPHIE

The acoustic levitation of a droplet of one liquid in another (in which an acoustic standing wave has been established) has been used to measure the mechanical properties of liquids, including biological samples. The basic format for the levitation apparatus is shown in Figure (1). Levitation is particularly useful when only small samples can be used; for instance, in measuring the properties of metastable (superheated, supercooled, or tensilely stressed) liquids, or for cases in which only a very limited amount of the liquid is available, as with a synthesized or extracted biological material.

We have reported measurements of the tensile strength of liquids, the dynamics of vapor bubble explosions, and the velocity of sound and adiabatic compressibility of levitated samples. Recently, we've extended our measurements of sound velocity and adiabatic compressibility from normal liquids to superheated liquids and to biological materials (Varanasi, et al - to be published). We have also been able to measure the interfacial tension associated with the interface between the levitated droplet and the surrounding host liquid by exciting the droplet into surface oscillations and noting at particular frequencies, the changes in the "rainbow effect" interference pattern produced when the droplet intercepts a laser beam. Moreover acoustic levitation shall be used to measure thresholds of neutron-induced cavitation by exposing levitated, superheated droplets to mono-energetic neutrons of known energy. Initially we will produce these neutrons by the fusion reaction in which accelerator produced deuterons are incident on a deuterated target.

Details of experimental procedures and recent results for normal, metastable, and biological liquids will be presented.

**ACKNOWLEDGEMENT**

This work is supported by the U.S. Office of Naval Research, the National Science Foundation, and the Petroleum Research Foundation, administered by the American Chemical Society.

INTRODUCTION

In the last few years acoustic emission analysis has received many applications to evaluate material properties with some applications to the phase transformations. The aim of the present research is the study of the phase transitions in liquid crystal GBOOA. A circular glass cell of 15 mm in diameter and 1 mm thick, was filled with melted GBOOA and acoustic emission was continuously detected in a range of temperatures from the isotropic phase down to the crystalline state.

RESULTS

Results (Fig. 1) show an increase in the acoustic emission near the transition temperatures, corresponding to isotropic-nematic-smectic-solid phase changes. A tentative explanation of such results can be given by taking into account the presence in the ordered mesophases of disclination lines or boundary surfaces between different oriented regions. The acoustic emission rate, shows the possibility of occurrence of a metastable smectic mesophase. Moreover, the occurrence of the crystallization is characterized by a strong acoustic emission.

![Diagram showing onset of crystallization and acoustic emission rate vs. temperature](image-url)
INTRODUCTION

Controlled experiments on the effects of ultrasonic compressional waves on the optical properties of a nematic liquid crystal with or without an applied electrical field have been carried out by the author, Dr. Chatro Sripaipan (now with Dept. of Elec. Engr., Chulalongkorn Univ., Bangkok, Thailand) and Dr. Charles Hayes (Phys. Dept., Univ. of Hawaii). Physical models based on acoustical streaming and mathematical analysis using hydrodynamic equations have been developed to explain the observed phenomena. Good agreements have been found between the theory and experimental results.

EXPERIMENTAL RESULTS

Patterns of lighted lines and circles parallel to the free boundaries of homeotropically aligned nematic liquid crystal cells and surrounding air bubbles in the cells were observed using a reflected-light polarizing microscope, when the cells were excited by ultrasonic compressional waves. These lines and circles were separated by about a quarter acoustic wave length. For increasing levels of excitation, the cell first went through the tunable birefringent phase where the center of the white areas show several orders of colors in the sequence of increasing phase difference between the ordinary and extra-ordinary rays. It then entered the turbulent phase in which the liquid crystal strongly scattered light and showed moving disclination lines and wavelike patterns. Measurements of the reflected light intensity shows the phase difference to be proportional to the fourth power of the excitation voltage applied to the piezoelectric transducer.

THEORY

We propose that, for the region of tunable birefringence, the lighting up of some areas and the colors seen may be interpreted as due to the tilt of the director from its normal position, where the director is the unit vector representing the local average direction of liquid crystal molecules. The tilt is caused by acoustic streaming flow induced by the standing waves formed by the transverse compressional waves reflected at the boundaries. In the turbulent region, the flow velocity is so high that the homeotropic ordering is disrupted and light is scattered by small domains of different director orientations.

The hydrodynamic equations of Leslie-Ericson(1) and their time averages are solved to the second order to find the director changes. The first order director change is to have a significant value only near the boundary. The second order director change caused by the streaming velocity is found to be $F \sin(2kx)$, where $F$ is a function which is (i) proportional to the square of the amplitude of the first order velocity and a viscosity coefficient, (ii) inversely proportional to the sound velocity and the bending elastic constant, and (iii) a function of the applied electric field, the thickness of the liquid crystal cell, and the boundary layer constant. $k$ is the ultrasonic wave number, and $x$ is the horizontal axis. The theory explains the observed lighted patterns and predicts the dependence of the light intensity on various physical parameters with reasonable accuracy.
ACOUSTO-OPTIC EXAMINATION OF PHASE TRANSITIONS IN LIQUID CRYSTALS

Kosmol, M. - Kwiek, P. Institute of Physics, University of Gdańsk, Sliwiński, A. Gdańsk, Poland
Witkowska-Borysewicz, M. Wojciechowska, I.

Introduction

Acoustical and optical properties of liquid crystals determine specific conditions for waves propagation in such a media where crystalical as well as hydrodynamical nature permeate one another. Many characteristic behaviours of elastic and optic constants have been observed /1-3/ near the phase transitions of liquid crystals. An explanation of physical mechanisms responsible for these behaviours is important in many l. c. applications for ultrasonic visualisation and for light modulation.

Experimental technique

In liquid crystal cell /see Fig. 1/ under temperature control an ultrasonic field is produced by a quartz transducer. The l. c. cell is situated in the optical arrangement for simultaneous schlieren and holographic observations. The He-Ne laser beam may be polarized and conoscopic vision is possible. Also, ultrasonic light diffraction phenomena of Raman-Neath and Bragg types can be realised. Photomultiplier or photographic technique is used for registration.

Results and conclusions

An ultrasonic wave passing through the liquid crystal changes its orientational structure what influences the optical properties. Simultaneous schlieren and holographic investigation of liquid crystal samples in monochromatic unpolarized and polarized light with ultrasonic field of 1 - 60 MHz and without it provided many details on growth of cybotatic groups against temperature and ultrasonic intensity near smectic A - nematic phase transition. Some results obtained for CBOOA /cyclyclobenzylidene-octyloxyaniline/ and for other liquid crystals can be interpreted in correlation with the results of other authors /1-4/ on abnormal behaviour of elastic constants, ultrasonic absorption, velocity etc.

References

ACOUSTO-OPTICAL EFFECTS IN A NEMATIC LIQUID CRYSTAL

Bertolotti, M. Universita' di Roma
Scudieri, F. C.N.R. - Italy
Sette, D.
Sturla, E.

INTRODUCTION

The effects of acoustic waves on optical properties of nematic liquid crystals have been considered. Regular stripe domain patterns in the form of rolls can be produced by cellular motion induced by shear component of ultrasonic field. Due to the optical anisotropy of the material, strong scattering of light is associated to the ultrasonic produced motion. The refractive index correlations can be studied by analyzing the spatial coherence properties of the scattered light. This method allows the determination of the refractive index correlation length which is connected to the orientation fluctuations correlation and to the velocity field.

RESULTS

The linear dimension $l_c$ of the coherence area as a function of the tension $V$ applied to the ultrasonic transducer is shown in Fig. 1. The behavior of the coherence area allows to determine the nature of the evolution of scattering centers at different hydrodynamic regimes. The considered effects are relevant for the construction of optoacoustical devices using liquid crystals.
INTRODUCTION
On applying a suitable voltage across a thin layer of a nematic liquid crystal, certain formations may be observed in a viewing microscope. When the applied voltage reaches a critical value $V_c$, a periodic distortion of the nematic alignment is clearly visible. These arrays of long parallel 'discontinuities' are referred to as 'Williams Domains' and are visible in unpolarized light, either by transmission or reflection. A.c. or d.c. fields may be used and the domains are obtained at any temperature within the nematic mesophase range. In the present experiments, similar patterns have been produced by coupling of surface waves with the liquid crystal.

EXPERIMENTAL
The simple cell (Fig.1) containing the liquid crystal was placed on the stage of a polarizing microscope and the phenomena were photographed, using a 35 mm Pentax camera attached to the microscope.

COMMENTS
The mechanical coupling between the surface waves and the liquid crystal arises from the periodic surface displacements of the solid, producing disturbances within the superposed crystal. Additionally, the consequent radiation pressure gradient in the liquid parallel to the interface will also stimulate a liquid flow which the authors have recorded on ciné film.

Fig. 1.
Diagram of Experimental system

Fig. 2.
Photograph of Domain Formation at 20MHz for Nematic 4-CYANO-4-n-PENTYL - BIPHENYL.

$E_t - E_r = +11.0$
Excitation voltage 4.2 volts r.m.s
$\leftarrow$ Propagation Direction.
Magnification 58x
5) 9 INTERNATIONAL CONGRESS ON ACOUSTICS
MADRID 4/9-VII-1977

ETUDE AU MOYEN D'ULTRASONS ET PAR DIFFUSION BRILLOUIN DES PHENOMENES CRITIQUES DANS DES CRISTAUX ANTIFERRODISTORSIFS

Berger, J.

Hauret, G.

Rousseau, M.

INTRODUCTION

La mesure des vitesses et de l'atténuation ultrasonores par les techniques de superposition d'échos et de diffusion BRILLOUIN, dans les composés isomorphes ACdF₃ (A = Tl, Rb, Cs) fournissent un ensemble cohérent de résultats complémentaires de 80 K à 500 K. Les différences de comportement critique des constantes élastiques et de l'atténuation de ces composés permettent de calculer les valeurs des différents exposants critiques et de développer un modèle microscopique du changement de phase.

MÉTHODES EXPERIMENTALES ET RÉSULTATS

Le mécanisme des changements de phase structuraux antiferrodistorsifs des composés AMF₃ est lié à l'instabilité d'un mode mou au point R de la zone de BRILLOUIN du système cubique. Le paramètre d'ordre associé à la transition est la rotation des octaèdres MF. Le couplage entre la déformation élastique et le paramètre d'ordre qui pilote la transition a été étudié à différentes fréquences.

De 1 à 100 MHz le couplage direct d'une onde ultrasonore permet de suivre l'évolution des constantes élastiques en fonction de la température. Les vitesses ultrasonores sont déterminées par une méthode de superposition d'échos pour des températures inférieures à 300 K. Cette technique nécessite des échantillons de grande dimension et la formation de domaine au voisinage de T° limite les possibilités de mesure.

A des fréquences de 10 à 200GHz le couplage d'un mode acoustique du cristal avec le paramètre d'ordre est détecté par diffusion BRILLOUIN. Cette technique réalise une sonde locale, autorisant des mesures sur de petits échantillons, peu influencée par la formation des domaines. Des mesures effectuées sur TICdF₃ permettent de suivre l'évolution des constantes élastiques au dessus et au dessous de la transition, ainsi que l'atténuation ultrasonore (fig. 1).

REFERENCES

ACOUSTIC INVESTIGATION OF DIBENZIL MONO- AND POLYCRYSTALS
IN PREMELTING RANGE

Adkhamov, A. Physical-technical Institute
Mukhtarov, N. Named after S. Umarov Academy of Science
Makhmudov, E. Tajic SSR

INTRODUCTION

The results of researched temperature-frequency dependences of wave-speed and the absorption coefficient of longitudinal ultrasonic waves near the melting temperature in dibenzil mono- and polycrystals---C_{14}H_{14} are represented in this paper.

EXPERIMENTAL TECHNIQUES

The monocrystals were grown from the melt by Bridgmen techniques.---Polycrystalline structures having the average sizes of crystallites-about 0.05 mm were obtained by the way of spontaneous crystallization under vacuum. The samples under investigation had following dimensions: diameter of 25 mm and the length from 10 to 15 mm.

The spreading of elastic waves inside monocrystals was studied along three basic crystallographic axes 100, 010, 001, and in polycrystals - along the direction of spontaneous crystallization in the sample.

Our measurements were carried out impulsly, namely by the alone fixed method with the application of immerse liquid(1) at the frequency band 2-35 MHz. The relative accuracy of sound velocity measurements was equal to 0.1-0.3 %, and the relative error of absorption coefficient measurements was equal to 3-8 %. In consequence of a strong anisotropy of dibenzil monocrystals and a history of their growth, the absolute values of the accuracy of sound velocity measurements in given directions can oscillate from 10 to 20 % according to the sound velocity 1-2 %.

RESULTS AND CONCLUSIONS

The absorption coefficient multiplied by wavelength in dependence on frequency in monocrystals for the temperature range 26-48°C is independent on temperature in the stability region of a crystal. Near the melting point, corresponding to instability condition, the temperature-time dependence is observed.

In polycrystals unlike from monocrystals the absorption peak multiplied by wavelength is arranged at low-frequency part of spectrum and is passed the maximum at the frequency range 3-5 MHz. As approaching to the melting point, the value M sharply increases, and the peak width remains the same.

By analyzing the obtained experimental data, we can suppose that acoustic properties are subjected to combinations of factors, such as the friability of structure, the changing of packing near the melting temperature, the formation of point defects in subgrains, the dislocations of different kinds, the appearance of fluctuation effects and others, which cause the additional changing of solid state.

REFERENCES

(1) U. Mezon, Physical acoustics, 1966, v.1, "World"
ULTRASONIC ATTENUATION IN CARBON STEELS AT HIGH TEMPERATURES

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INTRODUCTION

Acoustic attenuation studies provide a very sensitive tool for studies of steels at high temperatures. Although certain theories (2) and techniques (3) for determining the acoustic attenuation in polycrystalline samples are proposed, high temperature ultrasonic measurements are not very extensive.

THEORY

If we consider a steel sample, in cylindrical form, with a 'shoulder' at one end having a smaller diameter, and when ultrasonic waves propagate from larger crosssection S1 to the smaller crosssection S2, the reflection coefficient can be written as \( R/(1-R) = (S_1-S_2)/S_2 \) which can be approximated as \( (D^2-d^2)/D^2 \), D and d being the diameters of larger and smaller crosssections respectively. The pulse heights, \( h_1 \) and \( h_2 \) of the multiple pulse-echoes and the attenuation \( \alpha_2 \) in the 'shoulder' can be related as

\[
\frac{h_1}{h_2} = \frac{R}{1-R} \exp(2\alpha_2) \text{ (shoulder length)}.
\]

Here we consider the diffraction correction to be temperature independent.

EXPERIMENTAL TECHNIQUE

Measurements in EN steels are done by ultrasonic pulse technique at 5 MHz. The shoulder is kept in a furnace at the desired temperature which is measured by a sensitive thermocouple-millivoltmeter combination.

RESULTS AND CONCLUSIONS

Figure shows the variation of attenuation in the range 25 - 1000 C. Absorption peaks are evident at 170 and 750 C. First peak corresponds to martensitic transformation due to previous thermal cycling, and the second due to austenitic transformation. In the cooling run, peaks shift lower, possibly due to further structural changes. Ultrasonic velocities in the specimens show decrease with increasing temperature.

REFERENCES

ETUDE AU MOYEN D'ULTRA-SONS DE LA TRANSFORMATION STRUCTURALE ORDRE-DESORDRE DANS LES ALLIAGES DU TYPE Ni$_3$Fe

Turchi, P.
Zarembowitch, A.
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4, place Jussieu 75230 - Paris Cedex 05

INTRODUCTION

L'effet du degré d'ordre sur les propriétés physiques des solutions solides a fait l'objet de nombreuses études (1). Cependant, peu de résultats ont été publiés quant à l'incidence de l'ordre sur les constantes élastiques des alliages.

Les plus détaillés sont relatifs à Cu$_2$Au où les valeurs de constantes élastiques dans l'état ordonné et dans l'état désordonné ont été déterminées en fonction de la température (2, 3).

Nous présentons ici une étude de l'influence du paramètre d'ordre S, sur les vitesses de propagation d'ultrasons dans un monocristal de composition voisine de Ni$_3$Fe à différentes températures. Les résultats à température ambiante ont déjà été communiqués (4).

TECHNIQUE EXPERIMENTALE

La principale difficulté réside dans l'élaboration du cristal et la technique de mise en ordre. Ce problème a pu être résolu avec le concours de Y. CALVAYRAC (5). Ni$_3$Fe présente une transformation du type LI, ordre-désordre avec une température critique de 783°K. Par des recuits appropriés il est possible de maîtriser le paramètre S.

La mesure des constantes élastiques C$_{ij}$ en fonction de la température a été effectuée par la technique de pulse superposition à une fréquence voisine de 8 MHz, de la température ordinaire à 4° K.

RESULTATS

Les valeurs de C$_{ij}$ où S représente le paramètre d'ordre variant entre 0 et 1 ont été reportées dans le tableau ci-dessous. Les valeurs de $\theta_D$, température de DEBYE, ont été calculées pour les différents états d'ordre.

<table>
<thead>
<tr>
<th>État d'ordre</th>
<th>T° K</th>
<th>C$_{11}$</th>
<th>C$_{12}$</th>
<th>C$_{44}$</th>
<th>$\theta_D$° K</th>
</tr>
</thead>
<tbody>
<tr>
<td>S = 0</td>
<td>4,2</td>
<td>24,76</td>
<td>15,17</td>
<td>12,65</td>
<td>461 ± 2,0</td>
</tr>
<tr>
<td>OCD</td>
<td>300</td>
<td>23,59</td>
<td>15,10</td>
<td>11,92</td>
<td></td>
</tr>
<tr>
<td>S = 0,6</td>
<td>4,2</td>
<td>24,52</td>
<td>15,27</td>
<td>13,04</td>
<td>471,5 ± 2,0</td>
</tr>
<tr>
<td>OCD</td>
<td>300</td>
<td>24,51</td>
<td>15,95</td>
<td>12,28</td>
<td></td>
</tr>
<tr>
<td>S = 1</td>
<td>4,2</td>
<td>25,58</td>
<td>15,51</td>
<td>13,13</td>
<td>470,5 ± 2,0</td>
</tr>
<tr>
<td>OCD</td>
<td>300</td>
<td>24,59</td>
<td>15,29</td>
<td>12,41</td>
<td></td>
</tr>
</tbody>
</table>

O. C. D. : ordre à courte distance
O. G. D. : ordre à grande distance
(les C$_{ij}$ sont exprimées en 10$^6$ dynes cm$^{-2}$)

Comme on pouvait le prévoir théoriquement les valeurs de C$_{ij}$ sont sensibles à l'arrangement des atomes dans l'alliage. On note par ailleurs une variation de 10$^6$ K de $\theta_D$ entre l'état ordonné et l'état désordonné en accord avec la différence trouvée par FLINN et al (3) pour un alliage du type Cu$_2$Au.

REFERENCES

(2) S. SIEGEL, Phys. Rev. (1940) 57, 537
(4) P. TURCHI, F. PLICQUE et Y. CALVAYRAC, Script. Met, (1975) 9, 797
THE EFFECT OF ULTRASONIC DEFORMATION ON THE KINETICS OF AUSTENITE TRANSFORMATION IN STEELS

Abramov, O.V. Institute of Solid State Physics,
Kulemin, A.V. Academy of Sciences of the USSR
Manaenkov, V.P.
Nekrasova, S.Z.
Entin, R.I.

INTRODUCTION

The effect of ultrasonic on solid metals results in an increase of density of the structural defects (i.e., dislocations, vacancies) thus stimulating acceleration of phase transformations connected with diffusion.

EXPERIMENTAL TECHNIQUES

In this work, the effect of the ultrasonic deformations of amplitude $\varepsilon_m = 0-7 \times 10^{-4}$ (longitudinal vibration of frequency 19.5 kc) on the kinetics of perlite and bainite transformation of austenite of steels has been investigated. The ultrasonic influence on the perlite transformation of steel 504 (0.44% C, 0.25% Si, 4.16% Mn) at temperature 500-550°C has been studied. The ultrasonic treatment went on during 20-30 min.

The effect of ultrasonic on the bainite transformation of steel 30X16G2A (0.30% C, 1.02% Cr, 0.94% Mn, 0.92% Si, 2.1% Ni) has been investigated at temperature 350°C. The ultrasonic treatment carried out 1. Within the austenitizing of temperature 850°C during 10 min; 2. After austenitizing within the isothermal transformation during 1-4 min.

RESULTS AND CONCLUSIONS

The analysis of structure and properties of steels has shown that the ultrasonic increase the velocity of austenite transformation. It is established that there exists a certain threshold value of deformation, namely, $\varepsilon_m$, below which the effect of the ultrasonic does not result in a change of the kinetics of the isothermal transformation. At $\varepsilon_m > \varepsilon_m$, acceleration of the austenite transformation occurs, the degree being greater the higher amplitude $\varepsilon_m$ is. (see fig. I)

Dislocation internal friction $Q$ has been measured in these steels at the ultrasonic frequency 19.5 kc within the same range. Basing on these measurements and structural analysis one can assume that acceleration of the perlite and bainite transformation is connected with the peculiar features of the dislocation structure formed during the ultrasonic treatment.

![Fig. I. The effect of ultrasonic deformation on the kinetics of austenite transformation.](image-url)
We have developed a theory of nonlinear acoustical noise amplification by drift of the current carriers in piezoelectric semiconductors. The main purpose of our work is to investigate the noise spectrum evolution.

As long as a noise intensity is rather small a maximum in the noise spectrum arises at wave vector $q = q_m$ where the linear amplification coefficient $(q)$ has a maximum. The nonlinear equation for the phonon distribution function $N_q$ has the form $/1/:

$$\frac{w N_q}{3} - \gamma_c(q)N_q = \int P(q, q')N_q d^3q'/2 \tag{1}$$

where $w$ is the sound velocity, $\gamma_c(q)$ is the Peierls' term describing three-phonon collisions. The second term in the right-hand side describes the interaction of the phonons with overdamped electron density waves.

The ratio of the Peierls' collision term to the second term in the right-hand side of eq.(1) is of the order of

$$\lambda \equiv (V-w)\times T/\gamma \tag{2}$$

where $V$ is the drift velocity, $\gamma$ is the Debye length, $T$ is the Maxwell relaxation time, $\gamma$ is the electromechanical coupling constant squared.

In the case $\lambda \ll 1$ a new maximum in the spectrum at $q = q_m/2$ arises because of the splitting of the phonons at $q = q_m$. The whole intensity increases the intensities in both maxima become the same order of magnitude and the width of the maxima increase.

In the case $\lambda \gg 1$ spectrum evolution is determined by the behavior of the $P(q, q')$. A new maximum in the spectrum arises at $q = 0.77q_m$. The noise intensity in this maximum increases more rapidly than in the former one. The intensity increases, the position of the maximum shifts to $q = 0.67q_m$.

These results well describe the tendency of the noise spectrum development observed in many experiments (see review /2/).

REFERENCES


Les propriétés acoustiques non linéaires de composés ayant la structure cubique des pérovskites fluorées ont été étudiées à la température ambiante, sous pression hydrostatique, par la méthode ultrasonore de superpositions d'impulsions et celle de recouvrement d'échos d'impulsions (méthode de Papadakis). Les mesures ont été faites dans la gamme de pression suivante : 0 - 3000 bars. Les ondes ultrasonores engendrées par des quartz piezoélectriques de coupe X et Y avaient une fréquence fondamentale de 9 et 7 MHz.

**RÉSULTATS**

Alors que la plupart des cristaux ioniques de structure cubique étudiés au laboratoire (halogénure alcalins, chlorate de sodium (1)) présentent des variations linéaires de la vitesse des ondes ultrasonores en fonction de la pression dans la gamme 0 - 3000 bars, les pérovskites étudiées ici (TICdF₃ ; CsCdF₃) ont un comportement différent : la vitesse des ondes transversales ne varie linéairement avec la pression que jusqu'à 800 bars ; par contre, la vitesse des ondes longitudinales est quasiment linéaire jusqu'à 3000 bars cf. fig. 1 et 2.

**Fig. 1, 2 :** Temps de parcours d'une impulsion ultrasonore suivant un axe Aₗ dans un échantillon de CsCdF₃ en fonction de la pression.

En outre les mesures de vitesse en fonction de la pression nous ont permis de déterminer les dérivées par rapport à la pression des trois constantes élastiques du 2ème ordre, des combinaisons linéaires de constantes élastiques du 3ème ordre et certaines constantes micorscopique de Gruneisen. On peut remarquer :

1°) Les valeurs de dC_{44} / dP sont beaucoup plus grandes que celles de dC_{44} / dP. Ce comportement est tout à fait analogue à celui des pérovskites telles que RbMnF₃ (2) ou SrTiO₃ (3).  

2°) dC_{44} / dP est environ deux fois plus petit dans TICdF₃ que dans CsCdF₃. Il en résulte que la constante micorscopique γ (C_{44}) de TICdF₃ est quasiment nulle. Ce dernier résultat est à mettre en relation avec l'existence d'une transition de phase structurale pour TICdF₃ (transition à 191° K). Par contre CsCdF₃ ne présente pas de transition de phase.

**RÉFÉRENCES**

(2) E.R. NAIMON A.V. GRANATO Phys. Rev. B (1973) 7, 2091-2093  
The nonlinearity parameters of fused silica have been measured between room temperature and 3° Kelvin by the harmonic generation technique (1).

A 30 MHz ultrasonic pulse is generated in samples of fused silica which differ in hydroxyl ion content and in homogeneity. As the ultrasonic wave propagates through the sample it generates harmonics of the 30 MHz fundamental. By use of a capacitive receiving transducer, amplitudes of both the fundamental and the second harmonic are measured. Typical values of the fundamental and second harmonic amplitudes are, respectively, 2.2 Angstroms and 9.3 x 10^-3 Angstroms. These amplitudes are used to calculate the nonlinearity parameter (2)

\[ \beta = -\left( \frac{3C_{11}^{++}C_{11}^{-}}{3C_{11}^{+}} \right) \]

and hence the third-order elastic constant \( C_{11}^{-} \). Room temperature values of these quantities for four different samples are given in Table I.

As the temperature is lowered to 3 °K, \( C_{11}^{-} \) changes by 16% for Suprasil W1, and less than this for the other samples. The variation of \( \beta \) is 7% for Suprasil 2, and less than this for the other samples. Since \( \beta \) can be related to the longitudinal mode strain Gruneisen parameter \( \gamma_{11} \), this means that temperature dependence of the total Gruneisen parameter as great (3) as 800% must come from sources other than variations of the third-order elastic constant \( C_{11}^{-} \). [Research sponsored by the Office of Naval Research.]

<table>
<thead>
<tr>
<th>Sample</th>
<th>( \beta )</th>
<th>( C_{11}^{-} ) (10^12 dynes/cm^2)</th>
<th>( C_{11}^{-} ) (10^12 dynes/cm^2)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Suprasil W1</td>
<td>-3.86 ± 0.071</td>
<td>7.8059 ± 0.0029</td>
<td>6.70 ± 0.124</td>
</tr>
<tr>
<td>Suprasil W2</td>
<td>-3.95 ± 0.034</td>
<td>7.8076 ± 0.0020</td>
<td>6.85 ± 0.058</td>
</tr>
<tr>
<td>Suprasil 1</td>
<td>-4.14 ± 0.041</td>
<td>7.7633 ± 0.0016</td>
<td>7.51 ± 0.073</td>
</tr>
<tr>
<td>Suprasil 2</td>
<td>-3.90 ± 0.039</td>
<td>7.7633 ± 0.0014</td>
<td>6.75 ± 0.057</td>
</tr>
</tbody>
</table>

REFERENCES
LEAKY SURFACE WAVES IN ISOTROPIC SOLIDS

Viktorov, I.A.
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Academy of Sciences of the USSR

INTRODUCTION

It is known than in isotropic homogeneous perfectly elastic half space with a plane free boundary there are Rayleigh surface waves. The purpose of this paper is to show the existence two unknown leaky surface waves (1 and 2) in the half space provided that boundary conditions on its surface are slightly changed (impedance boundary conditions or in the case when liquid or solid layer is over the half space). Till now it was known the existence of leaky surface waves in some complicated elastic media: crystals [1], interface solid-liquid [2].

STATE OF THE PROBLEM

Let us consider the leaky surface waves (1,2) for case of solid half space having a contact with liquid or solid layer of h thickness. We attempt to find a solution of the dynamical equation of the theory of elasticity in form of plane waves systems. The frequency of these harmonic waves is equal to \( \omega \) and every of these wave propagates along the boundary (along x axis) with the same velocity \( C \) (phase factor \( e^{i \omega t + i kx} \)). The waves systems must satisfy the corresponding boundary conditions on the planes \( Z=0 \) and \( Z=-h \).

RESULTS AND CONCLUSIONS

The waves 1,2 have the following complex wave numbers:

\[
\begin{align*}
K_1 &= K_1^{(o)} (1 + i \alpha) \\
K_2 &= K_2^{(o)} (1 + i \beta)
\end{align*}
\]

If the Poisson ratio \( \nu < 0.25 \) we obtain the following approximated expressions for \( K_1^{(o)} \) : 

\[
K_1^{(o)} \approx K_0 \quad K_2^{(o)} \approx 0.6 K_0
\]

(\( K_0 = \frac{1}{\omega} \) — wave number for longitudinal wave). Thus the waves 1 and 2 decay exponentially with distance along the boundary \( Z=0 \). The displacements in these waves consist of surface and volume components. In process of spreading the surface component is slowly transformed to the volume component. The depth of the layer localization and the attenuation coefficient of the wave varies in wide limits when the layer thickness \( h \) varies from \( 0 \) to \( K_0 \). Very interesting case for practice is possible when the coefficient of attenuation due to transformation the surface component to the volume component is less than the attenuation coefficient due to diffraction effects.

REFERENCES

Pour le cas de céramique piezoelectrique et des cristaux piezoelectriques solubles dans l'eau il est difficile de produire les transducteurs interdigités par la méthode photolitographique à cause de la porosité de la céramique ou la sensibilité des surfaces à l'action des produits chimiques. À cause de cela, on a étudié dans notre laboratoire des transducteurs de plaques pour les appliquer à l'exitation des ondes de surfaces.

Cette méthode de génération des ondes de surfaces possède une importance dans le domaine d'investigation de qualité des matériaux pour l'application dans l'acoustoélectronique. En outre l'exitation des ondes de surfaces type Rayleigh est possible sur les matériaux non-piezoelectriques.

Pour la céramique piezoélectrique avec la surface métallisée - la pénétration des ondes de surface se calcule selon la formule:

\[ h = \frac{\lambda}{2k'} \]

où \( \lambda \) - longueur d'onde, \( k' \) - coefficient de couplage électromécanique. La plupart de l'énergie se propage dans une couche d'épaisseur \( h = 0.6\lambda \) - ce qui peut faciliter la génération des ondes de surface à l'aide des transducteurs piezoelectriques en forme de plaques vibrantes transversalement. Le domaine de vibration du transducteur doit se trouver dans une couche d'épaisseur \( 0.6\lambda \). Les expériences montrents que les perturbations provoquées par les ondes de Rayleigh et de volume sont très basses /fig.1/.

L'application de transducteurs en plaques permet de déterminer les coefficients de couplage pour les ondes de surface et les coefficients de couplage / \( k_s \)/ pour le matériau.

Voici les résultats - céramique PZT \( k_s = 0.24 \) \( k_{15} = 0.56 \)

\[ \text{LiJ}O_3 - \text{cristal} \quad k_s = 0.30 \quad k_{15} = 0.65 \]

(fig. 1)
INTRODUCTION

There is a growing interest in using layered transducers for propagating surface acoustic waves and for using a layered substrate to give velocity-frequency dispersion. The objective of the present work was to determine the relation between coupling efficiency and the (thickness/wavelength) ratio. The results obtained should provide a basis for the design of layer-devices at ultra-high frequencies.

THEORETICAL BASIS

The presence of a conducting film near to a piezoelectric layered structure will produce a velocity change from $V$ in the absence of the film to $V_f$ in its presence. The ratio $\Delta V/V = (V - V_f)/V$ is taken as a measure of the piezoelectric coupling efficiency (1). The presence of the conducting film will modify the boundary conditions of the linear piezoelectricity theory (2).

RESULTS AND CONCLUSIONS

The main points are as follows:

1) The profile of the coupling efficiency is, in general, not uniform throughout the range of (thickness/wavelength) between 0.0 and 0.9, as shown in Figs. 1 and 2.

2) The presence of the conducting film appears to change the coupling efficiencies at certain values of (thickness/wavelength) and also to cause a shift of their peak values, as indicated in Fig. 1.

3) The profile of the coupling efficiency is dependent on the piezoelectric characteristics of the different crystal combinations, as is evident in Fig. 2.

REFERENCES

INTRODUCTION

Hardness is usually determined arbitrarily by the resistance of the surface material to indentation. The resistance is assessed from the geometry of the indentation produced under a particular load. Since indentation involves the processes in plastic deformation and post yielding, nondestructive determination of hardness would require an observable parameter which relates to these processes and consequently the microstructure. Other than perhaps acoustic emission, no such nondestructively determinable parameter has yet been discovered which relates directly to the relevant characteristics of microstructure. However, the wavelength limited depth of penetration of a surface wave offers potential as a tool to determine hardness and to probe the subsurface hardness profile via the wave velocity. The purpose of this paper is to show some results achieved.

HARDNESS RELATION TO VELOCITY

A study has been made of the variation in surface wave velocity measured at a frequency of 5 MHz on the surface of through hardened specimens of steels of various compositions. In the case of EN9 steel for example, the velocity changes by ~2% for a hardness change of 270 VPN. Whereas in the case of 18%W, 4%Cr, 1%Va tool steel the velocity change is undetectable, i.e., less than 0.1%. The conclusion to be drawn is that unless surface wave velocity can be determined to better than 0.1% not all steel compositions are amenable to hardness determination by surface velocity measurement.

HARDNESS - DEPTH PROFILE RELATION TO VELOCITY

Since the depth of penetration of a surface wave is one wavelength, it is in principle possible to assess a subsurface hardness profile by monitoring wave velocity as a function of frequency. Experiments with a steel EN32A, in which velocity was found to change with hardness, showed that the profile could indeed be determined, even in a subtle case where slight surface decarburisation occurred due to cooling the specimen in air.

REFERENCES

(1) G. Curtis, AERE internal report 1972 (see also London University Ph.D Thesis by W. Phothiphitchitr 1973)
(2) Flambard and Lambert "Determination nondestructive des profondeurs de cementation" Instruments et Laboratoires 1974
VISUALISATION DES SURFACES D'ONDE EN MILIEU LIQUIDE : ONDES DE SURFACE ULTRASONORES

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INTRODUCTION
La réflexion par une interface métal-liquide, d'un faisceau ultrasonore limité a été étudiée par visualisation des surfaces de phase constante du champ acoustique. Des interfaces planes et cylindriques ont été envisagées. Le déphasage entre le faisceau réfléchi et le faisceau dû à la réémission de l'onde de surface excitée sous incidence critique a été mis en évidence. Une étude mathématique confirme ce résultat et permet de déterminer la célérité de l'onde de surface dans le cas d'une interface cylindrique.

ASPECT THÉORIQUE
Le phénomène de diffraction de la lumière par les ultrasons [1] est utilisé. Le maximum d'énergie lumineuse est obtenu dans la direction faisant l'angle \( \theta \) avec celle de la propagation suivant :\( \sin \theta = n(v/\lambda) \) (1) où \( \lambda \) est la longueur d'onde lumineuse et \( v \) la longueur d'onde acoustique. L'amplitude lumineuse \( U_n(v,t) \) du \( n \)-ième ordre est :\( U_n(v,t) = A_n \eta_n(v) \exp[i(wt-\eta t)] \) (2), où \( w \) et \( \eta \) sont respectivement les pulsations lumineuse et ultrasonore, \( A_n \) est une constante, \( v \) est le paramètre de Raman-Nath [2]. L'effet stroboscopique permet la visualisation des surfaces de phase ultrasonores.

DISPOSITIF EXPERIMENTAL
L'ensemble de visualisation est constitué d'une cuve où sont immergés dans l'eau, l'échantillon et le transducteur (2 à 12 MHz). Un faisceau laser modulé par des ultrasons de même fréquence, dans une cuve auxiliaire, traverse le bac d'expérimentation. Les divers ordres de diffraction se recombinent sur un écran d'observation. Des conditions de réglage doivent être respectées pour obtenir un contraste optimal [3].

RESULTATS ET DISCUSSIONS
L'irradiation d'une interface plane métal-liquide sous incidence critique, fait apparaître, en plus de la réflexion spéculaire, un faisceau rayonné depuis la zone d'impact du faisceau incident. Ce faisceau est interprété par la réémission due à l'onde de surface créée dans ces conditions particulières, en accord avec les résultats de Neubauer [4]. L'angle critique \( \alpha_c \) est défini par \( \sin \alpha_c = c/c_0 \) (3), où \( c \) est la célérité dans le liquide, et \( c_0 \) la célérité de la pseudo-onde de surface, similaire à celle de Rayleigh.

La visualisation des surfaces d'onde montre un déphasage égal à \( \pi \) entre les deux faisceaux. Ce résultat est retrouvé par le calcul [5]. La figure 1 concerne une interface cylindrique : \( A \) est le faisceau incident, \( B \) la réflexion, et \( C \) la réémission attribuée à l'onde de surface cylindrique. La comparaison entre la forme de la surface de phase et le calcul théorique [C.R. Acad. des Sciences, en publication] permet de déterminer la célérité de l'onde de surface associée.

INTRODUCTION

Real time visualizations of acoustic field patterns produced by surface waves have been done in the past by several authors through the use of different optical probing techniques (1). Presently, detailed images of the acoustic field perturbation produced by surface waves propagating on a $y\nu$-LiNbO$_3$ crystal are obtained through a modified dark field image technique that employs a high efficiency acousto-optical geometry of interaction.

EXPERIMENTAL

The experiments were performed with a light beam impinging on the propagation surface from the inside of the sample at almost normal incidence. The phase modulation produced on the reflected light wavefronts by the surface corrugation and by internal strain distribution, is highly increased with respect to the modulation of light transmitted through or reflected from the free surface. Proper calculations may show that the intensity of the diffracted light in the interaction geometry used may amount as up as to 15% of the undiffracted one, at acoustic frequencies of tens of MHz(2). An all field image of the acoustic pattern was obtained by filtering out in the Fourier plane all but the first orders of the spectrum of the scattered light beam.

Fig.1 shows the track changing effect of the acoustic energy flow obtained through a multistrip coupler. The wide strip striations appearing in the figure are due to the shear waves that are generated by the transducer and propagate underneath the surface. The high signal-to-noise ratio that this kind of visualization offers is a valid performance that may ensure current use of it in designing acoustic surface waves devices.

REFERENCES

(2) A. Alippi et al., Appl.Optics, (1976), 15, 2400.
BANC DE MESURE ABSOLUE DE VITESSE D'ONDES DE SURFACE PAR SONDE OPTIQUE

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INTRODUCTION

La dispersion de vitesse des ondes acoustiques de surface (SAW) caractérise les modifications superficielles d'un matériau, provoquées par dépôt (1), trempe (2) ...

Nous décrivons ici un banc de mesure absolue de vitesse, en lumière cohérente [d'après (3)], de précision élevée, dans la gamme de fréquence acoustique 10-50 MHz. La production des ondes de surface sur divers matériaux non piezoélectriques (acier, verre ...) est assurée par transferts d'ondes (4) depuis des transducteurs à quartz munis de peignes interdigités attachés sur leurs fréquences fondamentales ou harmoniques (5).

DISPOSITIF EXPERIMENTAL

Une onde lumineuse spatialement cohérente (laser He-Ne) est réfléchie ponctuellement par la surface sur laquelle se propage l'onde acoustique de fréquence \( F \) puis reçue par un photomultiplicateur (P.M.) à demi obturé (Knife-Edge). La détection synchrone du courant anodique du P.M. fournit la phase \( \varphi = 2 \pi x/\lambda \) de l'onde acoustique de longueur d'onde \( \lambda \). En déplaçant, parallèlement à la direction de propagation, l'échantillon d'une longueur \( L \) égale à un nombre entier \( N \) de \( \lambda \), repéré par les passages à zéro de \( \varphi \), on obtient la vitesse de propagation ultrasonore \( v = LF/N \).

Le dispositif a été étudié pour rendre les diverses mesures commodes et fiables. Un multiplexeur analogique commandé par le flux moyen du P.M. minimise les défauts de réfraction de la surface étudiée et les variations du flux laser incident. Un filtre élimine le bruit résiduel du signal TBF \( \varphi \). Un compteur de nombre de périodes à deux étapes prépositionnées permet de laisser défile un nombre suffisant \( N \) de périodes \( \lambda \) pour attendre l'état stationnaire. La longueur \( L \), correspondant au nombre de périodes utiles \( N \), est ensuite mesurée et affichée au micromètre grâce à un codeur optique solidaire de la vis micrométrique rectifiée de commande du déplacement, le début et l'arrêt du comptage \( L \) étant commandés par des passages à zéro positifs de \( \varphi \).

RESULTATS ET CONCLUSIONS

La course maximale \( L \) est 25 mm à l'œil près, \( \Delta F \leq 0,1 \) Hz : les mesures de \( v \) sont faites alors à 2 ou 3.10^{-4} \% près. Lors de mesures de qualification faites sur quartz taille Y, propagation suivant X, nous avons atteint \( v = 3155 \text{ m/sec} \pm 1 \text{ m/sec}. \)

D'autres résultats (influence de couches minces, trempe, etc ...) seront donnés au Colloque.

REFERENCES

(1) S.V. Bogdanov, M.D. Levin and L.B. Yakoukin, Soviet Physics Acoustics (1969) 15, 10
(2) J.A. Bucaro and C.M. Davis, J. Appl. Phys., (1972) 43, 2151
(4) A. Defebvre and J. Pouliquen, UL 75 Conf. Proceedings, 1975, 129
A new ultrasonic imaging system which uses a rotating M-sequence random phase mask for the coding of the two-dimensional spatial distribution of the ultrasonic field into a random time series is proposed.

A random phase mask has holes according to the 1 of a M-sequence along circles. The same sequence is used for circles of different diameter in the disk with corresponding different initial positions as shown in Fig. 1. Hence in the region of ABCD of the disk different portions of the M-sequence are arranged and the thickness of the disk is chosen so that π phase difference is produced between the waves passed the disk and those passed through the holes directly in the medium.

The disk is rotated at a constant rate and the cross-correlation between the random phase mask and the original M-sequence is taken, then it is clear that only one point in the region ABCD has sharp correlation (δ-correlation) with the M-sequence for a fixed delay of the cross-correlation. Therefore, if this mask is placed just behind the object, that is, if the object wave field is coded by this random phase mask, the cross-correlation function between the resulting field and the original M-sequence gives the distribution of the wave field. The schematical construction of the ultrasonic imaging system based on this idea is shown in Fig. 2.

Only a fixed point receiver is required to detect the transformed field and the cross-correlation function between this detected signal and the M-sequence generated by logical circuits is calculated in a mini-computer. The result is redistributed according to the relation of Fig. 1 in the corresponding region ABCD on the image and displayed on a display device.

A complete system was constructed by using a mask of 25 cm diameter with M-sequence of 255 period. An image of $25 \times 20$ meshes is obtained in 5 seconds at 1 MHz and displayed on the color monitor.

Fig.1 M-sequence Random Phase Mask

Fig.2 Schematic Construction of the System
This work describes the general lines of an echosonographic visualization system for medical diagnostics which, by means of new techniques, aims at the presentation of a real time image with high lateral resolution.

Generally speaking the system uses a linear array of 96 elements of which only a group of 32 is in action at a time. Switching off the first element by means of a multiplexer and switching on the next to the last one it is possible to obtain a fast lateral beam scanning. In order also to have a high lateral resolution in the whole field of exploration a dept scanning is performed using three acoustic pulses for each line of the image. Applying a convenient phase shading to the 32 active elements the three pulses are focalized successively in three different regions which cover the whole field of exploration in dept. The echoes coming from the focalized regions where the ultrasonic beam has the minimum width are received by the transducer which in turn is focalized again by means of proper delay lines and separated from the others by means of a gate. All the functions of the system, that is phase distribution in transmission and reception, multiplexing etc. are programmed and controlled by a minicomputer in order that a high flexibility is obtained in use.

The above mentioned lateral scanning is more convenient than sector scanning because it makes it possible to obtain a constant angular width of the non focalized acoustic beam in the whole field of exploration. On the other hand the dept scanning makes it possible to reach practically the same result even with a focalized beam.

The first results obtained using dept scanning show that a 32 element array, 32 mm long, at 1.5 MHz gives an acoustic beam with angular width of 1.30° in transmission that is a lateral width contained within 2 and 4 mm in the whole exploration range (50–150 mm).

A special technique was used for the construction of the elements of the transducer using small piezoelectric ceramic bars oscillating with a contour extensional mode. This technique permits an accurate selection of the elements before final assembling in order that a high degree of uniformity of the electroacoustic characteristics is obtained.

The research program described above is sponsored by the "Programma Finalizzato Tecnologie Biomodiche" of the C.N.R., Italy.
In recent years, ultrasonic imaging based on a pulse-echo method has become an accepted and useful technique in medical diagnosis. However, the applications of the technique to detection of tumor in the brain or the deep region of the abdomen has been limited by poor azimuthal resolution, because in this cases the ultrasonic waves of low frequency must be used to maintain their penetration power. In order to obtain high azimuthal resolution over a wide range with a single transmission of pulse, the concept of dynamic focusing has been proposed (1-2) and several dynamic focusing transducers have been constructed so far (3-4). We also have constructed and developed the dynamic focusing transducer as a prospective tool for observing inside the brain through the skull.

The dynamic focusing transducer is composed of an annular array of transducers, a set of delay elements, electronic control circuits and a CRT display unit, as shown in Fig. 1. An element of the array is used to transmit ultrasonic pulses and all elements are used to receive pulse echoes. The set delay elements is used so as to delay the phases of received signals and make them coincide. The electronic circuits are used to control the phase angle of the delay elements as a function of time elapsed after an ultrasonic pulse is transmitted. The outer radii of each ring transducers which form the array are given by \( \alpha n (n=1,2,\ldots,N) \), where \( \alpha \) is a constant, \( n \) is a number labeled on the ring transducer from inner to outer one sequentially and \( N \) shows the number of the transducers which form the array.

Theoretical analysis of the directional pattern of dynamic focusing transducer is carried out to give some instructions for the design of it. One of the results of theoretical analysis is as follows: the widest range of the dynamic focusing transducer is obtained by setting the numerical value of \( \alpha \) to 0.92 \( \sqrt{\frac{r}{\lambda}} \), where \( r \) is the radius of the curvature of the transducer and \( \lambda \) is the wave length of ultrasonic waves.

The constructed dynamic focusing transducer is composed of four annular transducers whose outer radii are 12.5, 17.7, 21.7, and 25.0 mm respectively and the frequency of ultrasonic waves is set at 1.67 MHz to maintain the penetration power. The radius of curvature of the transducer is 120 mm and ultrasonic beam whose half amplitude is less than 4 mm is obtained over a range of 60 - 200 mm.

REFERENCES
PROGRES EN IMAGERIE ACOUSTIQUE PAR DIFFRACTION DE BRAGG ET APPLICATIONS

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INTRODUCTION - Un dispositif d’imagerie ultrasonore en temps réel par diffraction de Bragg a été développé [1]. Il exploite les propriétés de l’interaction de deux ondes optique et acoustique (monochromatiques et cohérentes) qui, dans les conditions de Bragg, construit une image optique diffusée topologiquement identique à l’objet testé.

EXPERIMENTATION - Le pouvoir de résolution, lié à la qualité des optiques, a été amélioré en corrigeant au mieux les composants optiques de leur aberration de sphéricité. Le pouvoir de résolution, selon les directions principales, est de 0,9 mm et 0,2 mm (à 15 MHz, l’eau constituant le milieu de couplage entre les deux ondes). La transparence acoustique d’échantillons métalliques a été notablement améliorée par le choix du mercure comme milieu de propagation ultrasonore [2] acoustiquement coupé par l’intermédiaire d’une lame adaptatrice d’impédances à la zone d’interaction. A 15 kHz, l’eau constitue un milieu de couplage acousto-optique compétitif avec d’autres liquides, dû, en particulier, à une absorption modérée des ultrasons (0,5 dB/cm). À plus haute fréquence (30-50 MHz) certains verres peuvent remplacer l’eau et constituent une chaîne acousto-optique mercure-matériau métallique-verre bien adaptée acoustiquement (coefficient d’atténuation de ces verres : 2 dB/cm à 50 MHz).

RÉSULTATS ET CONCLUSIONS - Des illustrations à l’aide d’échantillons-type métauxiques (titane TA 6V) à défauts internes calibrés (Fig. 1 et 2) démontrent l’aptitude du procédé au contrôle non destructif de matériaux. À la fréquence de 5 MHz, quelques résultats ont été obtenus dans le domaine du contrôle biomédical.

Echantillons de titane TA 6V immergés dans le mercure :

Fig. 1 - trou de Ø 0,5 mm épaisseur 5 mm.
Fig. 2 - stries carrées en surface de largeur 0,1 mm épaisseur 3 mm
Champ visualisé : Ø 50 mm.


* Cette étude bénéficie du soutien de la DNM.
INTRODUCTION

In conventional pulse-echo method or acoustical holography, it is difficult to obtain high resolutions in lateral and longitudinal directions simultaneously. We have developed a new acoustic imaging method to improve the both resolutions by using an impulsive sound.

PROCEDURE OF RECONSTRUCTION AND COMPUTER SIMULATION

When a point source projects an impulsive sound \( f(t) \),

\[
f(t) = \begin{cases} \frac{P \exp\left(\frac{T_p}{2-t}/t\right) \sin 2\pi ft}{0} & \text{if } 0 < t < T_p \\ 0 & \text{if } t > T_p \end{cases}
\]

the magnitude of a reconstructed image \( I(x,y,z) \) is given by

\[
I(x,y,z) = K \int_{0}^{T_p} \left[ \int R(x',y',0,t+(r+r')/c) \, dr' \right] dt
\]

where \( R(x',y',0,t) \) is the received sound wave at the point on the receiving plane \( H \), \( T_p \) is pulse width and \( c \) is sound velocity.

The image of the source was constructed on the \( x-z \) plane by a numerical calculation. Here, the point source \( P \) was placed on the \( z \)-axis at a distance \( L \) from the origin, and a small number \( N \) of receivers were arranged on the \( x \)-axis of the receiving plane in a span \( 2A \). When \( N=11 \), \( 2A=L=100\lambda \), \( T_p=1.5/f \) and \( \tau=0.36/f \), a high resolution of \( 1.5\lambda \) was obtained for both directions.

EXPERIMENTAL RESULTS

Experiments were made in air using small speakers as a object. When a sound pulse of about one cycle wave duration (wave length \( \lambda=34\text{mm} \)) was used, both of \( 2A \) and \( L \) were about \( 100\lambda (3.5\text{m}) \) and \( N \) was 11, a clear image of the speaker was obtained as shown in Fig.3.
INTRODUCTION

Mechanical C scanning is an established non-destructive test procedure with numerous variants in existence. There are, however, three basic restrictions which can, in some applications, render the technique difficult or inappropriate:

a) the transducer scan must follow the specimen surface contours with the acoustic beam axis normal to the local tangent to the surface. b) liquid coupling is required. c) unless a large pitch scan is appropriate, the duration of a complete specimen scan could be considerable.

The purpose of this paper is to describe the first results of a new concept in through-transmission C-scanning which does not have these restrictions.

PRINCIPLE OF OPERATION

If a thin flexible polymer film is metallised upon one side only and a dc voltage is applied to the metallising, then the film will adhere by electrostatic attraction to any conducting surface which is at earth potential relative to the dc voltage supply. The flexible film traps a thin layer of air. If an ultra-sound pulse moves through the substrate, then the thickness of the air gap will vary and as a consequence if the metallising is further connected to an amplifier the device will produce a pulse which relates directly to displacement in the acoustic waveform. This is the action of the transducer as a receiver, but since it is a reciprocal device it functions equally well as a transmitter. (A transmitter-receiver pair is approximately 30 dB less sensitive than a conventional PZT pair).

Using photo-etch techniques it is possible to produce arrays of strip transducers in a single sheet of film. In the transmission mode, the transducer strips are applied to opposing faces of a flat or curved parallel sided object with the receiver array at 90° to the transmitter array. To form a transmission C-scan a control system is devised to sequentially select transmitter strips, pulse excite them, and for each one scan the receiver strips storing the transmitted pulse amplitudes ready for display as elements of the overall C-scan.

PRELIMINARY RESULTS

In order to test the basic capability of POLYSCAN, arrays of 32 transducer strips 5mm wide with 2mm spacing were formed and applied to the walls of a narrow water immersion tank into which simple objects could be placed. Working at a frequency of 1.6MHz, C scans of objects up to 200mm square are formed in 12.8 seconds indicating the intrinsic high speed of POLYSCAN.

Current work with POLYSCAN concerns the imaging of adhesive bonding defects, where its relatively coarse resolution is readily acceptable.

REFERENCES

(2) G. Curtis and A.B. Joinson, "Polyscan - A rapid ultrasonic through-transmission C scanning system" Proc. 8th World Conf. on NDT Cannes 1976.
An ultrasonic imaging system which combines aperture synthesis with spectral synthesis is proposed.

Generally speaking, the aperture synthesis is used to get high azimuth resolution as in the case of conventional synthetic aperture sonar systems, while the spectral synthesis, that is the utilization of waves which cover certain frequency range, is used to get high range resolution. Several methods to reconstruct the images from the data obtained by scanning the aperture and changing the frequency are considered.

The first method consists of the following processes; i) complex image for each frequency of waves in the frequency range is obtained by completely synthesizing the aperture, ii) they are superposed, iii) the intensity of the resulting image is displayed. The relation between the range of the aperture synthesis and the resulting azimuth resolution and the relation between the frequency range and the range resolution are derived. The required interval of frequency step to eliminate the ghost image and the suitable weighting function in frequency domain to suppress the side lobes are also discussed.

A variation is as follows; i) complex image for certain image region are obtained by partly synthesizing the aperture, ii) intensity image are obtained for each subregion, iii) the subimages are connected patchwisely and the whole image is obtained. The special features of each method are discussed from the viewpoint of the coherency controlled ultrasonic imaging systems.

A concrete system was constructed. In this system the following parameters and methods were adopted; i) ultrasonic waves between 1.0 MHz and 1.5 MHz were used in water, ii) aperture synthesis was carried out along a circle of 20 cm diameter, iii) image reconstruction is performed in a mini-computer, iv) results are displayed on a color CRT. The schematic block diagram is shown in Fig. 1.

The results obtained for several objects with different surface conditions by using the proposed methods show the special features and effectiveness of each method and imply the need of the selection of suitable reconstruction method depending on the condition of the object to be imaged.

Fig. 1: Schematic Construction of the System.
APPLICATION OF PIEZOELECTRIC POLYMER TRANSDUCERS FOR ACOUSTIC IMAGING

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INTRODUCTION

A piezoelectric polymer, polyvinylidene fluoride (PVF₂), is used to form an underwater acoustic imaging array. The material is cheap and easily polarized, and can have electrodes evaporated onto its surface in any desired configuration. 200 parallel strip electrodes are used in this application.

THEORETICAL BASES

PVF₂ can be drawn into very thin sheets, typically 6, 12, and 25 microns in thickness, and is suitable for use as a transducer with a flat response up to at least 100 MHz. The material has an acoustic impedance very close to that of water, ensuring good matching for underwater applications. Its piezoelectric strain constant $d_{33}$ has been measured, and although found to be less than that for conventional transducers, the resulting low receiving sensitivity is partly compensated for by the improved matching to water. (The constant $d_{33}$ relates strain and electric field in the thickness direction.)

The method of imaging used is based on the production of sum and difference frequencies (1), and is one of the first underwater applications of PVF₂.

EXPERIMENTAL TECHNIQUES

The array consists of a 12 micron thick sheet measuring 20 x 20 cm. Aluminium is evaporated on one side of the PVF₂ to form an earth electrode. On the other side a photoconductive layer of zinc telluride and 200 parallel electrodes of indium oxide are evaporated. These electrodes are transparent and allow light from an incoherent source to reach the photoconductive layer, which behaves as an optical switch when an electric field is set up in the PVF₂ by incident acoustic waves. The light source has a fan-shaped beam which spans all 200 electrodes orthogonally, sweeping across their entire length by means of a rotating mirror. By modulating the light intensity at frequency $f_0$ ('pump' frequency), a signal is received from each strip with sum and difference frequencies ($f_0 \pm f_p$), where $f_0$ is the 'source' frequency of the acoustic waves from an underwater object being visualised. A time-varying voltage is obtained from each of the strips since spatial variations in pressure across the acoustic beam cause corresponding variations in the electric field in the transducer. These voltages are applied to a column of light emitting diodes (one for each strip) which are viewed in another rotating mirror as a two-dimensional array having a spatial variation of intensity.

RESULTS AND CONCLUSIONS

The system is still under development but the various principles of operation have been tested and its optimum frequency range for imaging is 100-300 kHz. At frequencies above this the photoconductive layer does not respond quickly enough to the modulation frequency of the light beam to produce an image. Area for area the PVF₂ is about 30 dB less sensitive than PZT transducers off resonance but is ideally suited for underwater applications.

REFERENCES

(1) C. Sabet and C.W. Turner, Electronics Letters (1976) 12, 44
Some New Ultrasonic Lenses

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Introduction

Acoustic imaging was one of the goals of NDT, very soon after ultrasound has been proved to be able to find flaws in a non-transparent object. When Pohlman succeeded in producing a very simple and cheap image converter in 1937, the famous Pohlman cell, he was quick to introduce an acoustic lens with it, made of perspex. However, for various reasons his system was not popular.

Then, for a time, lenses were intended for a purpose other than imaging, concentrating high acoustic power. Tarndczy was a pioneer in this field.

After a period of neglect efforts put into acoustic imaging - and with it, lenses - are again on the increase, not only for NDT, but also for medical diagnostics and short-range underwater applications. This tendency has prompted our activity in this field.

The problems are these: most solid lens materials, plastics or rubbers, show high attenuation, while with liquid lenses the shape is more of a problem.

For both types the sound velocity generally goes down with increasing temperature, while in water it goes up, raising the problem of compensating the lens for the effect of temperature changes.

Folds was one of the most successful lens makers for short range sonar applications. He investigated a large number of possible lens materials and made some quite good temperature-corrected lenses with low aberrations.

We wanted to work mostly at frequencies up to several MHz, about five or ten times higher than Folds, where his lenses would be too lossy.

The Lenses

As a first step, we wanted to make a lens performing well at room temperature only, so from the data of Folds, we selected a silicone rubber, RTV 602 and a liquid, FC 75, purely on the basis of refractive index, impedance match and low absorption, ignoring any temperature effect for the time being. Our first lens consisted of a plano-concave aspherical rubber member supported by a taut Melinex membrane in contact with a liquid part with another Melinex membrane. Two such units formed a symmetrical doublet. This system gave an insertion loss of only 2 dB at 2 MHz with a diameter of 19 cm and an aperture of F1. A resolution of 2λ/line was achieved. Aberrations were not noticeable.

Then we measured the temperature coefficient of our lens materials and found that the refractive index of rubber w.r.t. liquid changed by 0.00033 or 0.0584/°C. Looking for further lens materials we found a new plastic on the market under the trade name TPX. Its impedance ratio to water is 1.25, ultrasound absorption similar to perspex and its refractive index 0.68, nearly independent of temperature between 16 to 23°C. The advantage of a TPX lens over the above described rubber-liquid combination lies in its mechanical stability. A combination of two TPX lenses with FC 75 in between has the remarkably strong refractive power of 3.7, which is worth having, even if losses may thus be slightly higher than with TPX alone and the temperature effect is not negligible. The latter problem can be overcome anyway by enclosing the lens and image space into thermostatically controlled water in a housing with a Melinex window.

New results will be reported.

References

(1) Pohlman, R.: German Patent DRP.710413 (1937)
(2) Tarndczy, T.: Ultrasonics 3 (1965) 115-127
THE CALCULATION OF MOLECULAR PARAMETERS FROM ACOUSTICALLY
INDUCED OPTICAL DIFFRACTION DATA

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INTRODUCTION

A solution of the difference-differential equations, describing the optical
diffraction by an ultrasonic wave is given. As a result, general formulae for the
molecular orientational relaxation time \( \tau \) and for the parameter \( G \) describing the
molecular assymmetry are found in function of the phase angle and the rotation of
the polarization plane of the incident light beam.

BASIC EQUATIONS

The electric field components \( E_x \) and \( E_y \) of a light beam entering the liquid along
the \( z \)-axis, are solutions of

\[
\frac{\partial^2 E_y}{\partial t^2} = \frac{\mu_y^2(x,t)}{\varepsilon_0 c^2} - \frac{\partial^2 E_x}{\partial t^2} \tag{1}
\]

When the liquid is disturbed by an ultrasonic wave with angular frequency \( \omega^* \) the
refractive index can be written as:

\[
u_x(x,t) = u_x^0 + u_x \sin (\omega^* t - k^* x + \delta^*)
\]

\[
u_y(x,t) = u_y^0 + u_y \sin (\omega^* t - k^* x + \delta_y)
\]

Solving (1) in the Raman-Nath region \( (\rho << 1) \), the amplitudes \( c_{nx} \) and \( c_{ny} \) of the
various diffraction orders are:

\[
\begin{align*}
\phi_{nx}(z) &= J_n(\gamma_n) + \frac{ie}{\rho} \left[ \frac{n^2 \tau \gamma_n}{\varphi_n^0} J_n(\gamma_n) + \frac{e_n}{\varphi_n} \frac{dJ_n}{d\gamma_n} \right] \\
\phi_{ny}(z) &= J_n(\gamma_n) - \frac{ie}{\rho} \left[ \frac{n^2 \tau \gamma_n}{\varphi_n^0} J_n(\gamma_n) + \frac{e_n}{\varphi_n} \frac{dJ_n}{d\gamma_n} \right]
\end{align*}
\]

Reasoning on the same manner as Klein and Fitts (1), \( G \) and \( \tau \) can be expressed
in function of the phase angle \( \gamma_n \) and the rotation \( \beta_n \) of the polarization plane:

\[
\begin{align*}
\tau &= \frac{2\beta_n}{\omega^* \left[ e_n - \frac{\rho g n \beta_n}{2} \right]} \\
G &= \left[ \gamma_n - \frac{n \rho g n \beta_n}{2} \right]^2 + 4\beta_n^2
\end{align*}
\]

In the Bragg region \( (\rho >> 1) \), non zero amplitudes exist only for the zeroth
and first diffraction order. That leads to:

\[
\begin{align*}
\tau &= \frac{2\beta_1}{\omega^* \left[ \gamma - \frac{g_1}{2} \right]} \\
G &= \left[ \gamma - \frac{g_1}{2} \right]^2 + 4\beta_1^2
\end{align*}
\]

CONCLUSIONS

Although the various intensities of a diffracted light wave are described in a
totally different way in the Raman-Nath and in the Bragg-region, the final results
for the molecular parameters \( G \) and \( \tau \) are identical.
Furthermore, we found in both the cases the same correction term containing \( \rho \). This
term is not negligible for the first diffraction order in the Bragg region and for
higher diffraction orders in the Raman-Nath region.

REFERENCES

If an intense laser wave is passing through a liquid, the relation between $D$ and $E$ is given by

$$D = \varepsilon_0 (1 + X) E,$$  \hspace{1cm} (1)

in which $X$ is generally field dependent.

Substituting the field-dependent

$$X = X_1 + X_2 E^2, \quad (X_2 < X_1)$$  \hspace{1cm} (2)

into (1), we have

$$D = \varepsilon_0 (1 + X_2) E + \varepsilon_0 X_2 E^3$$ \hspace{1cm} (3)

If the considered medium is disturbed by an ultrasonic wave, with frequency $\nu^u$ and wavelength $\lambda^u$, we deduce the following nonlinear relation

$$D = [\varepsilon + \varepsilon_1 \sin 2\pi (\nu^u t - \frac{X}{\lambda^u})] E + \varepsilon_2 E^3$$ \hspace{1cm} (4)

which is responsible for the generation of a third harmonic lightwave. From the Maxwell equations, the wave-equation is deduced for the electric field. Looking for a solution of the form

$$E = E_1 + E_2 E^2,$$ \hspace{1cm} (5)

the scalar wave-equation is equivalent with a system of two partial differential equations, for $E_1$ and $E_2$ respectively. The first equation is the usual wave-equation while the second one is of the same form as this found by Jozefowska (1). In the Raman-Nath region, the intensities of the diffracted lightwaves can be calculated by means of the generating function method of Kulasisko, Mertens, Leroy (2). The influence of the nonlinear part is calculated for some liquids.

REFERENCES

INTRODUCTION

Slowly but surely, coherent optical methods are gaining an important place in the field of optical systems design. Ultrasonic waves are in general coherent too, nevertheless, practically no attempts have yet been made to use them for data processing in the acoustic domain.

THEORETICAL BASES

Fundamentals of the analogy are based on the concept that light and sound waves are disturbances, and both may be accurately described by the hypothesis that they are scalar, monochromatic waves. Coherent optics is not of itself a new subject, but these ideas have only recently been applied to acoustic image forming systems. The topic of this presentation is to demonstrate that coherent ultrasonic data processing is feasible and may be used to differentiate textures of anorganic as well as of organic origin.

EXPERIMENTAL TECHNIQUES

The technique we wish to present does not want to reproduce an image, it merely wants to answer some simple questions about it: what are the absorption and/or scattering properties of the tissue it represents. The intensity distribution that is formed in the focal plane of an acoustic lens placed behind the transilluminated tissue sample may look complicated but it consists of simple diffraction patterns which are characterized by symmetry around the center point. Due to the symmetry of the pattern, it can be analysed by a special detector system in measuring the energy distribution as a function of angle with wedge-shaped elements, and as a function of radius with ringshaped elements.

In this diffraction pattern analysis leading to specimen recognition, we do not calculate what the observed pattern should be. Instead we observe the patterns of a large number of samples and then test - by some predetermined logic - selected features of our measurements to determine how well they describe the sample under investigation. Such a feature can be the spatial frequency $K$ of the structure and can be determined by measuring the radial distance $r$ from the center of the pattern in which the diffraction spot appears:

$$r = \frac{F}{K}$$

where $F$ is the focal length of the acoustic lens, and $K$ is the wavelength.

CONCLUSIONS

A short research film will demonstrate how these ideas can be put into practice.

REFERENCES

INTRODUCTION

Piezoelectric transducers are commonly used to monitor acoustic emission signals. Although such transducers can detect acoustic emission events, the information they yield is modified by the response characteristics of the transducers themselves, as well as by those of the medium which is required to acoustically couple the transducers to the test material. Optical detection methods offer many advantages over piezoelectric transducers for monitoring acoustic emission waves. Among these advantages are: direct contact with the test material is unnecessary; no couplant is required; the waveform of the acoustic emission event is not modified by the optical detector; optical detectors have high sensitivity and broad bandwidths; they can be absolutely calibrated; and they can give the instantaneous surface displacement due to the acoustic emission waves.

OPTICAL METHODS

The various optical methods which have proven to be ideally suited for probing the detailed characteristics of ultrasonic waves vary in their suitability for detection of acoustic emission signals (1). The knife-edge technique lacks sensitivity at frequencies in the 100 kHz range and will not sense waves moving in directions at right-angles to the knife-edge. The diffraction technique is not applicable to the study of waves having a wide range of frequencies and a short wave train because the phase grating in such cases is not periodic. Optical heterodyning, as with any Doppler frequency shift sensing method, suffers from the fact that the detected signal contains information about the velocity of the test material surface, not the displacement. A further difficulty with this method is that both undesired mechanical vibrations and acoustic emissions produce Doppler shifts proportional to the velocity of the surface. Since optical interferometers measure optical path differences, they can be used to directly measure instantaneous surface displacements as a function of time or position. Interferometers have very high sensitivity over an extremely broad frequency range and may be designed to detect acoustic emission signals in the presence of undesired mechanical vibrations and other background disturbances.

EXPERIMENTAL RESULTS

Several optical techniques were used to monitor acoustic emission signals from both simulated and real sources and the detected signals compared with signals obtained using conventional piezoelectric transducers. The predicted advantage of the optical detectors over the piezoelectric transducers were realized in all cases. These results suggest that optical detectors can be beneficially used to detect acoustic emission signals in high temperature and radioactive environments inaccessible by other techniques.

ACKNOWLEDGMENTS

This research was supported by the U.S. Army Research Office. A special note of appreciation in this regard is due Dr. George Mayer.

REFERENCES

INTRODUCTION

Les techniques utilisant l'interaction acousto-optique permettent de prélever l'information "in situ" dans les matériaux transparents. Les expériences que nous avons mises au point autorisent notamment la mesure absolue de la vitesse et de l'atténuation des ultrasons se propageant dans ces matériaux avec une très bonne précision ceci dans la gamme de fréquence 100 MHz - quelques GHz.

MESURE DE LA VITESSE

L'analyse par hétérodynage optique d'un faisceau lumineux diffracté dans les conditions de BRAGG par des ondes élastiques permet de recueillir l'amplitude et la phase de celles-ci (1). Par déplacement du volume d'interaction acousto-optique dans l'échantillon on peut réaliser un étalement des longueurs d'onde acoustiques en longueur d'onde optiques. Cette méthode nouvelle qui peut être exploitée en vue d'une étude locale (échelle 10 μm) des propriétés élastiques locales des matériaux permet également de mesurer la valeur absolue de la vitesse de phase des ondes élastiques avec une précision inégalée (1 ms⁻¹ près).

MESURE DE L'ATTÉNUATION

La mesure de l'intensité d'un faisceau lumineux diffracté par les ondes élastiques dans les conditions de BRAGG en fonction de la distance au transducteur permet d'atteindre leur atténuation intrinsèque. L'extension de cette technique vers les basses températures réalisée au laboratoire autorise l'étude de changements de phase ainsi que la mise en evidence d'un changement de régime de propagation des ondes élastiques dans des composés anharmoniques.

REFERENCES

(1) F. SIMONDET, F. MICHAUD, R. TORQUET Optics Commun (1976) 16, n°3
(2) F. MICHAUD, F. SIMONDET, L. ROYER, R. VACHER Phonon scattering in solids International Conference Nottingham (1976)
ANOMALOUS REFLECTION OF ACOUSTIC SHOCK PULSE AT A SOLID-SOLID INTERFACE

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INTRODUCTION

The experimental studies have been made on the generation of acoustic shock wave of single pulsive form in solids (1). In this paper, the acoustic shock pulse is used to measure anomalous reflection phenomenon at an interface between specimens before and after annealing of the same specimens. The measurements are performed at room temperature using a polycrystalline aluminum rod of 1 cm in diameter. The experimental results are compared with the analytical results obtained by means of stress-strain relationship of a material deformed by high intensity shock pulse.

EXPERIMENTAL METHOD

An interface is formed by specimen I before annealing and the specimen II after annealing as shown in Fig. 1. The annealing temperature are 350°C, 450°C and 500°C. PZT transducer of 1 mm in thickness is used for measurement of incident pulse and reflected pulse from an interface. The acoustic shock pulse used in the experiment is 15 to 150 bar in stress range and 20 microseconds in pulse duration.

RESULTS

The reflection phenomenon is not observed for low and moderate intensity pulses. However, for high intensity pulses, it is observed (2). The sign of the reflected pulse is altered and the pulse length is shorter than the incident pulse. The waveform distortion of shock pulse due to velocity dispersion is negligible small for incident pulse but must be considered for reflected pulse including the components higher than incident pulse. In Fig. 2, the peak values of reflected pulses are shown as a function of the peak values of incident pulse. The open circles are the values measured by the PZT transducer, the closed circles are the values at an interface corrected by the Pochhammer solution for the velocity dispersion and solid line is the value at an interface analyzed by using the stress-strain relationship of the specimen II deformed plastically by high intensity shock pulses. The reflection coefficients increase with increasing of stress amplitude also of annealing temperature. The details are described in a reference (2).

REFERENCES

(1) Y. Yasumoto et al., Acustica 30 (1974), 261.
(2) Y. Yasumoto et al., Acustica 36 No. 5 (1977) in press.
COUPLED VIBRATION OF A CYLINDRICAL SHELL FOR RADIATING HIGH INTENSITY ULTRASOUND

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INTRODUCTION

The coupled vibration of longitudinal and radial modes in a cylindrical shell is useful to radiate high-intensity ultrasound for continuous processing of liquids and gases.

In the present paper, the solutions to the coupled vibration obtained by the "apparent-elasticity" method are discussed, and its usefulness is shown experimentally.

THEORETICAL ANALYSIS AND EXPERIMENTAL RESULTS

Following frequency equations are obtained by the "apparent-elasticity" method in the boundary conditions that both sides are stress free and the circumferential length is one wave length.

\[ \frac{\omega L}{\sqrt{1 - \nu^2}} - \frac{n}{C} = \frac{\pi}{2} \] (1)
\[ \frac{\omega R}{\sqrt{1 - \nu^2}} - \frac{n}{C} = 1 \] (2)

\( \omega \): circular frequency
\( L \): length
\( v \): Poisson's ratio
\( R \): mean radius
\( C \): bar velocity

Where, \( n \) is the ratio of the stress in the axial direction to the stress in the circumferential direction. The case \( n < 0 \) corresponds to "antiphase" and \( n > 0 \) corresponds to "in-phase", and the lower curve and the upper curve in Fig.1 respectively.

The natural frequencies in various cylindrical shells made of structural steel are plotted in Fig.1. These frequencies are measured by the method reported in the preceding paper (1).

CONCLUSION

Eliminating \( n \) in equations (1) and (2) shows easily that this method is equivalent to the frequency equation to a thin cylindrical shell analyzed by A.E.H. Love (2). But in this method, the axial propagation constant is obtained at first, and then the natural frequency is obtained in the boundary condition. So this method makes it possible to apply the method of distributed systems to the analysis of vibrations in a cylindrical shell.

Then, by this method, the system for processing liquids with electro-acoustic efficiency of about 78% was made.

Now, the system for processing gases using the transverse and longitudinal coupled mode is being made.

REFERENCES

(1) E. Mori, K. Itoh, 7th ICA, 25-D-7

Fig.1 Natural frequency of a finite-length cylindrical shell as a function of the inner radius to outer radius ratio (\( y_d \)) in the coupled region.
MEASUREMENT METHOD OF ACOUSTIC POWER IN ULTRASONIC CLEANER BY METHOD OF CALORIMETRY

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INTRODUCTION

For evaluation of ultrasonic cleaner, it is necessary to know its acoustical power. A measurement method of acoustic power of ultrasonic cleaner by method of calorimetry is described(1).

METHOD OF MEASUREMENT

As shown in Fig. 1, cleaner is set in standard working condition and in the tank of which proper quantity of water is filled. From the value of water temperature rise, acoustic power W in cleaning tank can be calculated as shown eq. (1).

\[
W = \frac{3186 \cdot M \cdot AT}{t} \text{ (watt)} \quad (1)
\]

where M: Quantity of water including water equivalent of cleaning tank, t: working time, AT: Temperature rise during t seconds.

Before this measurement, water temperature in cleaning tank and atmospheric temperature must be in the equilibrium state. And measurements must be carried out during short time before the heat of the transducers caused by being driven flows into cleaning tank. This heat is major important cause of external heat flowing into the water. Consequently it is necessary to use high sensitivity and quick response instrument in temperature measurement.

RESULTS AND CONCLUSIONS

Measurement of acoustic power of an ultrasonic cleaner on market was made. Twelve ferrite transducers are bonded at the bottom of cleaning tank (bottom area: 180mm x 200mm, depth 200mm) of the cleaner. In case of water level 110mm and being driven by 150 watt electrical power, measured acoustic power by this method was 110 watt as shown in Fig. 2. Acoustic power can be measured easily even though in work shop, this measuring method may be convenient in such case as quality control of mass production cleaners.

REFERENCE

(1)M.Ide, Reports of 1966 Spring Meeting of Acoustical Soc. of Japan (1966), 185
ULTRASONIC ASSIST IN DEWATERING AND FILTRATION OF FINE PARTICLES

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INTRODUCTION

This paper reports two studies in which ultrasound was used to aid both vacuum filtration and the drying of washed pulverized coal. The ultrasonic frequency used was 20kHz with sound intensities up to 165 dB. Particle size used ranged from 20-400 microns with approximately 65% being less than 150 microns in size.

THEORETICAL BASES

The primary basis for applying ultrasound to these processes was to increase the liquid flow rate from the interior of the porous mass to both the filtering surface and to the drying surface. This is in accordance to Biot's equation:

\[ f > \frac{\mu}{4\pi d^2} \]

where \( f \) is sound frequency in cycles per second, \( \mu \) is kinematic viscosity in square centimeters per second, and \( d \) is the channel diameter in centimeters.

When the sound frequency and intensity applied to the system are correct, the normal viscous liquid flow of the liquid through the channels becomes a plug type of liquid flow.

In the drying process, a second basis for applying ultrasound was to increase the evaporation rate at the liquid-air interface by producing more air turbulence by the sound pulsations.

EXPERIMENTAL TECHNIQUES

With vacuum filtration, airborne ultrasonic radiation was introduced onto the top surface of the filter cake being dewatered.

With drying, airborne ultrasonic radiation was introduced onto the drying surface of a tray drier type of set-up using a cross-flow stream of warm air.

RESULTS AND CONCLUSIONS

Fig (1) shows the results obtained by adding ultrasound to the vacuum filtration process. Fig (2) shows the results obtained by adding ultrasound to the drying process. For both systems, the introduction of ultrasound improved the efficiency of the process.

Fig. 1 Curve showing percent increase in filter cake dryness versus ultrasonic intensity.

Fig. 2 Curve showing percent increase in drying rate versus water content.

REFERENCES

UNTERSUCHUNGEN ÜBER DIE ULTRASCHALLBEWIRKUNG AUF DEN VAKUUM-FILTRATIONSPROZESS VON INDUSTRIEABWÄSSERN

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INTRODUKTION


EXPERIMENTALER TEIL


SCHLUSSEFOLGERUNGEN

- Die Beschallung der untersuchten Suspension vergrößert den eigenen Widerstand und vermindert die Hydratation.
- Es wurde die Änderung der Makrostruktur von beschallten Filtrationsplättchen, die man leichter von Filterstoff beseitigen kann, als die unbeschallte, festgestellt.
- Die veränderte Makrostruktur beeinflusst auf die Erhaltung des konstanten Unterdruckswert während des Filtrationsverlaufs.
- Die Ultraschalleffekte sind günstiger bei der Temperatur von 291° und 298° als von 283°K.

LITERATUR

/1/E.Kowalska,A.Słoczka,J.Zajęczowska, Przegląd Górniczy/1967/12,563
Matematyka-Fizyka/1972/21,121
/3/A.Słoczka, Roczniki Chemii /1968/ 42,1075
THE AGGLOMERATION OF AEROSOLS BY AIRBORNE INTENSE ULTRASOUND WAVE

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ABSTRACT

On the agglomeration of aerosols by sound wave has been investigated in detail since Brandt, Hiedemann and others. In this paper we describe an experimental result of reduced exhaust smog by intense airborne ultrasound wave as an industrial application of the agglomeration. The combined using with a bolt-clamped vibrator and vibration plates in flexural mode generated intense sound field in the space of box. The aerosol of exhausted smog caused the agglomeration, and the result of this fact, the density of black smog obviously dropped under a half (1)(2) This method is able to apply to industrial using because of a quantity of smog is exhausted smog in the incinerator which can easily buy at the market.

EXPERIMENT

The equipment which was shown in Fig.1 inserted three vibrating plates as a sound source in a box which was made by iron. By the operating on a sound source can generate the sound pressure of 700 N/m² (151 dB) within a box. In addition to this phenomenon, water mist is generated to help the agglomeration of smog aerosol by vibrating plates, in addition to this fact, the water 750 cc/minute supply to make cool down three vibrating plates. The aerosol of exhausted smog caused the agglomeration, and the result of this fact, the density of black smog obviously dropped under a half. The curve 1(2) when aerosol flow at an exhaust pipe was 0.7 m/second, the rate of contamination by smog was 25 % in the case of no action of ultrasound wave and no water mist. And also the curve 1 (2) directly recorded so that drew off black smog from an incinerator. The curve 2(3) using two equipments in a series. In this case, the rate of black smog reduced 2.5 X, it became almost 1/10.

CONCLUSION

We could make an intense sound field by using the vibrating plate as a sound source. This is one of the industrial applications for agglomeration of aerosols. By using a small exhaust pipe successfully reduced and corrected exhausted aerosols. However, about the other various aerosols as materials do not try to study yet. We shall improve these experimental conclusions.

REFERENCES

AN EXPERIMENTAL CHAMBER FOR THE ULTRASONIC COAGULATION OF SMOKES

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INTRODUCTION

In the previous Congress on Acoustics (London 1974) we presented the first results of the laboratory experiments on ultrasonic precipitation of smoke particles under static conditions, using a piezoelectric emitter (1). During the last years we have worked at developing and extending these experiments. As a consequence we have developed a new coagulation chamber, made for dynamic operation, in which the general characteristics have been considerably improved.

DESCRIPTION OF THE CHAMBER

The coagulation chamber is a cylindrical tube, made of polyvinylchloride, of 220 mm in diameter with a reflecting end and a high-directional piezoelectric emitter on the other. The reflecting surface, opposite the emitter, is parallel to its radiating face and can be adjusted to make the chamber length a multiple of half waves to obtain a good standing wave field. The emitter, made for an operating frequency of about 20 kHz, consists of a flexural vibrating stepped plate (200 mm in diameter) driven at its center by a piezoelectric vibrator (2). The chamber was designed with the aim of studying experimentally the acoustic conditions for the treatment of the aerosol. Therefore, the length can be varied from about 500 mm to 2000 mm in order to select the most adequate dimension, bearing in mind that the degree of purification of the smoke depends on the treatment time and the energy density.

EXPERIMENTAL

Acoustic tests have essentially consisted of measuring the ultrasonic radiation inside the chamber. The sound pressure distribution on the axis and on parallel lines at different radii have been measured. Figure 1 shows the axial pressure distribution for an input power of 150 W (tube length = 1800 mm); the maximum sound pressure level is 173 dB and the average level is about 161 dB.

On the other hand, the chamber was tested for the precipitation, in dynamic operation, of a finely dispersed carbon black smoke (average particle radius = 0.2 μm) with a concentration of from 6 to 11 gr/m³ and flows of the order of 8000 cm³/s. The clearing of the smoke was measured by changes in light transmission during the process. Collection efficiencies in the acoustic chamber of up to 90% have been obtained for treatment times of about 10 seconds.

REFERENCES

(2) L. Gaete, J.A. Gallego and G. R. Corral, Proc. 9ICA (1977)
DEPLACEMENTS D'ÉQUILIBRE DANS LES SUSPENSIONS ULTRASONORISEES

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INTRODUCTION

À la suite de nos mesurages concernant l'action du champ ultrasonique de cavitation sur les suspensions de ZnO en eau, nous relevons l'instabilité dans le temps de ces systèmes hétérogènes quant à la distribution de dimensions des particules solides. L'objet de notre recherche a été l'étude du déplacement, dans le temps, de l'équilibre de la distribution aussi bien dans les échantillons-témoin que dans les suspensions irradiées ci-dessus mentionnées.

BASES THÉORIQUES

On suppose qu'à la base du phénomène d'agglomération relevé par nos expériences il y a un mécanisme de heurts dus au mouvement brownien. Pour la fréquence de ces heurts dans l'unité de volume, nous avons obtenu la relation :
\[ \dot{\gamma} \sim r^{3/2} = \alpha \cdot (\alpha'/\eta)^{2/3} \cdot (T/\eta)^{1/2} \] (1)

où \( r \) = le rayon moyen des particules, \( \alpha' \) = constante, \( c' \) = la concentration des particules browniennes; \( \gamma \) = la densité de la phase solide; \( T \) = la température absolue; \( \eta \) = la viscosité du milieu de dispersion.

La croissance, dans le temps, de la dimension moyenne des particules du système dispersé résulte de la formule suivante :
\[ (\Delta r)^{1/2} = \beta \cdot B \cdot (\alpha'/\eta)^{2/3} \cdot (T/\eta)^{1/2} \cdot \Delta t \] (2)

où \( \beta \) = une constante; \( B \) = un coefficient d'adhérence; \( \Delta t \) = l'intervalle de temps à partir du moment de la préparation.

RÉSULTATS EXPERIMENTAUX

Les suspensions de ZnO en eau, ayant une concentration de 0,33 \( \cdot 10^3 \) g/cm³, ont été préparées par agitation magnétique et respectivement par irradiation en champ ultrasonique de cavitation. Nous avons déterminé la distribution de dimensions par une technique de sédimentation en champ de gravitationnel /1/.

Nous avons calculé les distributions intégrales et différentielles, tout comme les surfaces spécifiques moyennes (\( S_m \)), après la préparation et à diverses intervalles. Dans les deux cas, on constate une baisse dans le temps du paramètre \( S_m \), dont la variation est plus grande dans la suspension irradiée. La différence entre la surface spécifique moyenne de l'échantillon et celle lui correspondant dans la préparation irradiée diminue dans le temps, sans disparaître complètement. Cette variation dans le temps de \( S_m \) n'est que l'expression d'une croissance des dimensions moyennes des particules, par la suite du processus d'agglomération dans le temps, qui correspond au mécanisme que nous sommes proposé d'employer.

BIBLIOGRAPHIE

ACOUSTIC CAVITATION ON MELTS OF LOW-MELTING METALS

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INTRODUCTION

The investigations of the cavitation processes were made on the melts of metals with low temperature of crystallization (Sn,Bi,Pb,In, Ga,Zn) at the temperature t~1,2 T₀ (T₀—the melting point of the metal) under the action of ultrasound at frequency 20 kc.

THEORETICAL BASES

The analysis of the character of the motion of the cavitation bubble in the low-melting metals and the assessment of the values of the finite amplitude pressure wave was made by means of a computer using the Keerwood-Bethe-Gilmor equation for the bubbles with the initial radius \( R_0 = 10^{-7} - 10^{-6} \) cm. For melts the equation of state was used in the form:

\[
P = A \left[ \left( \frac{\rho_0}{\rho} \right)^n - 1 \right]
\]

here \( P \) — pressure; \( A, n \) —parameters; \( \rho_0, \rho \) —densities in nondisturbance and disturbance melts correspondingly.

EXPERIMENTAL TECHNIQUES

The experiments were made using the cylindrical focusing and rod transducers with titanium radiators. The threshold of cavitation was measured by the change of the vibrations shape due to the cavitation noise spectrum appearance. These measurements were made by means of the waveguide metallic and spherical piezoceramic (in Ga) probes. The eddy-current pick up was used for the control of displacement.

RESULTS AND CONCLUSIONS

Acoustic cavitation on melts occurred at the displacement value 1-3\( \mu m \) which corresponded to the effective mean pressure 1-3 atm and the resistance of the radiation \( 10^5 \) g/cm² sec.

The computer analysis has shown that the subharmonics in Ga have appeared at the pressure 2,7 atm for the bubbles with the initial radius \( R_0 = 5 - 10^{-7} \) cm.

The influence of the properties of the melt on the character of the motion of the cavitation bubble was estimated in the calculation where the meaning of the velocity \( c \), surface tension \( \gamma_{ls} \), viscosity \( \eta \) and density \( \rho \) changed over the range such as might occur in real metals. The calculations have shown that the changes of the values of \( \rho, \eta \) did not affect greatly the character of the motion of the cavitation bubble. The pressure in the shock wave reduces as the surface tension increases and the velocity decreases. (see fig. 1).

The results of the calculation are confirmed by the experiments.
EFFETS ELECTRIQUES DES BULLES DE CAVITATION. INFLUENCE DE DIVERS PARAMETRES

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INTRODUCTION
Au cours du Bème ICA à Londres nous avons exposé des résultats relatifs aux effets électriques liés à l'évolution rapide d'une bulle de gaz dans un liquide. Nous sommes toujours guidés par le même modèle physique suivant lequel une bulle qui implose prend une vitesse importante et entraîne la formation d'un microjet capable de transporter des charges électriques. Nous avons détecté celles-ci à l'aide d'une sonde disposée à l'intérieur d'un liquide soumis à un champ ultrasonore.

ACTION D'UN CHAMP MAGNETIQUE
Dans des conditions bien définies, l'action d'un champ magnétique peut réduire ou augmenter le potentiel mesuré d'une centaine de millivolts. D'autre part, en général une inversion du sens du champ appliqué ne change pas les résultats comme le montre l'enregistrement ci-contre. Bien entendu, si l'on arrête les ultrasons, le champ magnétique est sans effet. Les résultats s'interprètent en faisant intervenir l'action d'un champ magnétique sur des charges liées au mouvement des bulles de cavitation qui forment le nuage de cavitation. Le champ magnétique déplace celui-ci par rapport à la sonde, ce qui modifie l'intensité du courant mesuré.

ACTION DES GAZ DISSOUS
Ayant établi un régime de relaxation entre cavitation de gaz et cavitation de vapeur, si au cours du régime de cavitation de gaz on injecte du xénon, ce régime cesse aussi-tôt, la cavitation est réduite et le potentiel mesuré varie considérablement. Il suffit alors d'injecter de l'azote pour retrouver le régime de relaxation initial. L'Injection de gaz perturbe également la relaxation de température correspondante.

CHANGEMENT DE SIGNE DES COURANTS MESURES
Si on modifie progressivement le régime de cavitation, quelle que soit la méthode utilisée, on observe, pour le courant, le passage progressif d'une valeur négative à une valeur positive. À chaque instant, l'arrêt des ultrasons ramène le courant à la même valeur moyenne d'équilibre. Des variations brusques et légèrement oscillantes du courant correspondent, à notre avis, à un apport direct de charges par les bulles tandis qu'une variation continue de type exponentiel, traduit plutôt un courant de diffusion dû au mouvement des charges libres dans le liquide.


Die Kavitationskevernenkomplexe bilden ein akustisch nichthomogenes Milieu (Diskontinuitäten) im flüssigen Kontinuum. Es ist deshalb verständlich, daß an der Grenzfläche der bestehenden oder entstehenden und schwindenden (kollabierenden) Kavitierungskeverne im flüssigen Medium eine Reflexion der US-Energie nach den bekannten phys. Gesetzen entsteht. Wenn man aus diesem Grunde ins flüssige Medium US-Wellen z.B. mit der Frequenz von 10 MHz einführt, dann werden diese Wellen vom Kavitierungskevernenkomplex reflektiert. Es ist verständlich, daß man die der Bewegungsgeschwindigkeit des Kevenenkomplexes proportionale Dopplerfrequenz \( \Delta f = f_0 \frac{2v}{c} \cos \gamma \) verwenden kann, wo \( f_0 \) die Frequenz des ausgesandten US, \( v \) die Bewegungsgeschwindigkeit der Kevenen, \( c \) die Geschw. der Fortpflanzung der US-Wellen im flüssigen Medium und \( \gamma \) der Neigungswinkel der Sonde in der Richtung der Kevenenbewegung ist. Falls eine Kaverne entsteht und kollabiert, ist \( v \) die Geschw., mit der die Kaverne entsteht oder kollabiert.

Literatur:
INTRODUCTION

The fatigue mechanisms in copper, brass, iron and titanium at ultrasonic frequencies, ~20kHz, have been shown to differ from those at the lower frequencies commonly used in engineering tests, i.e. ~30 Hz.

EXPERIMENTAL TECHNIQUES

The high frequency tests were carried out using a piezoelectric driver, mechanical transformer system capable of producing large uniform strains in the center section of resonant dumbbell specimens (1). The ultrasonic mode of stressing was axial push/pull. The low frequency tests were done in reversed torsion on thin-walled tube specimens. All specimens were machined, annealed and electropolished before testing. The micrographs are mounted with the horizontal direction parallel to the specimen axis.

RESULTS AND CONCLUSIONS

At high frequencies localized slip occurs in isolated large grains with slip planes oriented in the maximum shear stress direction. An example of this "ultrasonic" slip (2) is shown in Fig. 1. The importance of this ultrasonic slip is that it is an isolated area of persistent slip that readily develops into a fissure capable of cracking neighboring grains to form a microcrack. The ultrasonic stressing rapidly develops the microcrack into the propagating macrocrack that parts the specimen. The ultrasonic frequency fatigue process is strikingly efficient in that very little micro-damage is observed that is extraneous to the development of the eventual macro-crack.

This is in contrast to the fatigue mechanisms operating at low frequencies. An example from copper is shown in Fig. 2 where now most grains develop multiple slip zone fissures producing numerous microcracks. At low frequencies fatigue occurs by a random linking of various microcracks until finally one becomes the propagating macrocrack.

Fig. 1 Scanning electron micrograph of an "ultrasonic" persistent slipband of the surface of a copper specimen. The micrograph is 36 μm wide.

Fig. 2 Scanning electron micrograph of the intrusion-extrusion fissures on the surface of a low frequency copper specimen. The micrograph is 15 μm wide.

REFERENCES

(1) D.E. MacDonald, Engineering Fracture Mechanics (1976) 8, 17.
APPLICATION OF SONIC AND ULTRASONIC VIBRATIONS TO ACCELERATED FATIGUE TESTS OF MATERIALS

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INTRODUCTION

Development of new structural materials and preparation of recommendations for their use is associated with a large extent of fatigue testing to evaluate effects of various manufacture factors. Fatigue tests conducted with conventional loading frequency (up to 100 Hz) require much time that may lead to substantial delay in a wider utilization of new materials.

A promising testing procedure appears to be the accelerated fatigue test employing increased frequency of loading. In this case, a question arises, however, as to how the loading frequency affects the fatigue strength of materials. Investigations into this effect were conducted on alloys with different levels of strength and ductility in a large range of both loading frequency and temperature.

EXPERIMENTAL TECHNIQUES

Fatigue tests were conducted using various testing machines (1) including electro-hydraulic pulsators with loading frequency of up to 30 Hz, vibration tables with load frequency ranging from hundreds of hertz to 2 kHz and magnetostrictive set-ups with loading frequency of 3 kHz and above.

RESULTS AND CONCLUSIONS

It is shown that fatigue strength decreases as the strength level of metals grows and ductility level diminishes. Analysis of the obtained dependence of fatigue limit and load frequency made it possible to evaluate load frequency ranges within which the fatigue limit change due to increased loading frequency does not exceed 10%, which is well within the bounds of scatter of results obtained in experimental fatigue limit determination. This finding enables to raise the loading frequency up to 500-1000 Hz for most metallic materials and up to 10-20 kHz for high-strength ones without the chance to distort fatigue results.

The results of investigating frequency effect on the reduced factor of stress concentration showed that the latter did not depend on the cyclic load rate for a wide frequency range.

Using test procedures with sonic and ultrasonic loading frequencies, the effects of a number of manufacture factors such as heat and surface treatment modes, alloying elements content and welding technique on fatigue strength were investigated for titanium alloys. It was shown that high frequency loading method can be readily applied to quick and reliable evaluation of the above effects on the fatigue strength of metallic materials.

REFERENCES

INTRODUCTION

Carbon fibre reinforced plastics are being used for many high-technology applications particularly in aerospace where their structural integrity has to be ensured. Acoustic emission is one of the best prospects for nondestructive testing of CFRP. It is the purpose of this paper to show the usefulness of spectrum analysis for acoustic emissions and to show that this analysis can identify different emission source mechanisms and therefore different composite failure mechanisms.

THEORETICAL BASIS

An observed acoustic emission spectrum is a product of a number of transfer functions. This can be written to a first approximation as:

\[ O(f) = S(f) \cdot T(f) \cdot N(f) \]

where \( O(f) \) is the observed emission spectrum, \( S(f) \) is the source spectrum, \( T(f) \) is the transducer spectrum and \( N(f) \) the normal modes of the specimen and mounts. Thus if \( T(f) \) and \( N(f) \) remain constant then \( O(f) \) will reflect changes in \( S(f) \).

EXPERIMENTAL TECHNIQUES

Different acoustic emission source mechanisms were produced by stressing specimens of identical geometry in either tension or torsion. All CFRP specimens contained unidirectional arrays of either type 1 or type 2 fibre. A PZT transducer with a centre frequency of 2 MHz was used as a detector covering a frequency range from 100 Hz to 2 MHz. After amplification the transducer signals were digitised and stored on magnetic tape. FFT's were generated using a PDP8/I small computer and displayed on a facsimile recorder. A moments analysis was also performed on the data to enable small changes in the observed power spectrum to be quantified.

RESULTS AND CONCLUSIONS

It was found that the spectra from different sources were very similar in terms of the range of frequencies observed but that the distribution of energy amongst all the available system modes was source dependent. These differences were often quite small and a statistical moments analysis was used to describe quantitatively the shape of this distribution. It was found that the differences in spectra arising from torsional and tensile failure were most clearly shown by the second and third moments. Results have been plotted in moment space allowing confidence limits to be set for a particular source mechanism and its associated power spectrum moments.

REFERENCES

Résumé

Dans le but de définir les phénomènes physiques et mécaniques gouvernant la rupture des matériaux, deux objectifs ont été déterminés : d'une part, l'analyse de la configuration des fissures au sein d'une éprouvette de minerai de fer soumise à des contraintes de compression uniaxiales ; d'autre part, l'étude d'un mécanisme de propagation de fissures dans des échantillons de graphite.

Des contraintes de compression uniaxiales ont été appliquées à des échantillons de minerai de fer, secs et humides, jusqu'à rupture. Deux techniques ont été utilisées : l'extensométrie et l'émission acoustique dans les bandes basses, moyennes et hautes fréquences, de façon à suivre simultanément l'évolution de différents paramètres : le module d'Young, le coefficient de Poisson, l'amplitude des microbruits, l'énergie acoustique et le spectre de fréquences du matériau étudié.

Un mécanisme de fissuration est proposé. Une loi approchée, reliant les déformations à l'émission acoustique, a été déterminée par la méthode des différences finies. Des bandes de fréquences caractéristiques du minerai ont été définies. L'influence très nette de l'humidité de l'échantillon a été mise en évidence.

Des échantillons de graphite de section carrée, dont la région médiane a été réduite à un triangle isocèle, ont été soumis à des contraintes de flexion. La détection de l'émission acoustique a été réalisée par une chaîne de comptage à seuil AUDIMAT à hautes fréquences. Des procédures expérimentales ont été adoptées pour mettre en évidence l'influence de divers paramètres. Des éprouvettes, taillées parallèlement et perpendiculairement à l'axe de filage, ont été étudiées. L'énergie d'initiation et l'énergie de propagation de fissures ont été calculées à partir de l'enregistrement de la courbe force-fléche. Les variations de la contrainte d'initiation de fissures et de la charge de rupture de l'échantillon, en fonction de la section réduite de l'éprouvette, ont été établies (courbe de WÖHLER). L'influence de la vitesse de mise en charge et de la variation des paramètres géométriques a été étudiée.
INTRODUCTION

Design of modern instruments and apparatus incorporating sources of powerful sound and ultrasound is associated with the development of acoustically active materials for electroacoustic transducers, enabling to obtain high specific levels of emitted energy. Performance and efficiency of such instrumentation involving acoustically active materials is greatly determined by fatigue strength of their individual elements. Hence, fatigue strength of piezoelectric ceramics and ferrites is of great interest, in particular, the effect of various manufacture and design factors on fatigue strength.

EXPERIMENTAL EQUIPMENT

Using magnetostrictive fatigue machines (1) with frequency of 20 - 40 kHz, fatigue strength was investigated on specimens subjected to fully-reversed cycles of uniform tension-compression produced by inertial forces of specimen's mass. Since brittle materials were fatigue tested, a special technique was developed for connecting the specimen to the vibration exciter (2). Cyclic stress amplitude was calculated using the specimen's end face vibration amplitude.

RESULTS AND CONCLUSIONS

Fatigue strength characteristics were obtained for piezocrystals of various composition. These made it possible to estimate the effects of polarization conditions on fatigue limit of the materials under investigation. A quantitative evaluation was made as to how fatigue strength is influenced by acoustically active materials of the glass phase that is present in the piezocrystal and ferrite specimens.

The results obtained in the present study enable to assess the effect of the direction of piezoelements' polarization on fatigue strength of piezoceramics.

Along with determining fatigue properties, amplitude dependences of energy dissipation were studied for piezoceramics. The study revealed monotone rise of specific value of energy dissipation with increasing cyclic stress amplitude as well as the existence of correlation between energy dissipation and cyclic strength levels of piezoceramics.

REFERENCES

INTRODUCTION

Il y a en principe une possibilité de réaliser un détecteur des défauts sur la surface lisse par le moyen acoustique ou optique afin de contrôler mécaniquement des produits industriels. Cependant le contrôle des défauts est actuellement effectué visuellement par inspection humaine au procédé de fabrication industrielle des carreaux, bien que la plupart des procédés de fabrication aient été automatisés. Il se trouve que la surface du carreau émaillé est lisse, mais légèrement convexé, concave ou tordue. Par conséquent, la direction de l’onde réfléchie vacille. Il en résulte que l’amplitude du signal réfléchi fluctue au récepteur. Dans ces conditions, si l’on émet l’onde à faisceau étroit, la réception du signal réfléchi sera incertaine. Nous avons résolu cette situation difficile par un moyen acoustique tel que de balayer la surface à contrôler avec un rayonnement ultrasonore entraîné à large faisceau et le traitement différentiel du signal. D’autre part, le moyen optique est influencé par la couleur (foncé ou pâle) ou par le dessin en couleur de la surface à contrôler.

PRINCIPE DE MÉTHODE ET RÉSULTATS EXPERIMENTAUX

Une surface à contrôler (par exemple, un échantillon de carreau émaillé) passe à l’aide d’une courroie transporteuse sous le dispositif d’émiteurs et récepteurs en étant rayonné l’onde ultrasonore entrainée à large faisceau. Les combinaisons d’émiteur et récepteur sont disposées en parallèle de façon à couvrir toute la surface avec balayages ultrasonores. Une combinaison d’émiteur et récepteur balaye l’aire du ruban de la surface. Supposons qu’une surface lisse de carreau émaillé est légèrement convexe et qu’elle a une défaut de glaçure. L’onde réfléchie captée par le récepteur est modulée à cause de la convexité et également décrue à cause de la diffusion par le défaut. On peut obtenir le signal provenant du défaut par le traitement différentiel, en éliminant le signal de convexité.

CONCLUSION

Dans nos études, nous avons vérifié en laboratoire que le système décrit ici a une détectabilité comparable à l’inspection humaine; mais il faudra encore perfectionner quelques problèmes pour mettre ce système dans le procédé de fabrication industrielle, notamment la détection des défauts à l’encoignure de surfaces.

Figure 1 - Représentation schématique du dispositif expérimental.
Figure 2 - Schéma de principe de fonctionnement de la détection d’un défaut.
Figure 3 - Un exemple de résultat typique.
Ein wichtiger wirtschaftlicher Teil des Betriebes großer Maschinen ist die rechtzeitige und automatisierte Identifizierung der Defekte, die in der Anlage entstehen, ohne daβ diese zu demonstrieren und zu kontrollieren ist. Unsere Aufmerksamkeit wird auf die Kontrolle dynamisch beanspruchter Maschinen mittels einer Analyse des stochastischen durch einen dynamisch beanspruchten Maschinenteil emittierten Ultraschallsignals gerichtet.

Bisher haben wir die Diagnostik z.B. an folgenden Maschinensteilen realisiert: Gleit- und Wälzlager jeglicher Art und Größe, Maschinenteile - Schaltgetriebe, Verbrennungsvorgang in Naphtha- und Benzinmotoren, Kompressoren jeglicher Größe, Gleitlager, in denen die Kurbelwellen der Verbrennungsmotorwellen gelagert sind; tribologische Prozesse in Kolbengruppen von Kompressoren und Verbrennungsmotoren und Kavitationsvorgänge in hydraulischen Systemen (Wasserturbinen, 100 MW, Pumpenanlagen u.s.w.).

Für diese Zwecke haben wir die diagnostischen Geräte Diagnost 01 FEL, Diagnost 014 FEL, Diagnost 020 FEL, Diagnost 030 FEL und den Wahrscheinlichkeitsanalysator des stochastischen Ultraschallsignals - Diagnost PA 040 FEL entwickelt.

Die durch die Maschinenteile hervorgerufene Ultraschallfeldemission verfolgen wir im Frequenzbereich von 100 kHz bis 1 MHz nach der Detektion, d.h. am Ausgang unserer Diagnosten registrieren wir das Ultraschallsignal mit Hilfe von Analogmethoden, bzw. messen wir die Dichte der Wahrscheinlichkeitsverteilung, in der die Pegel des analysierten Signals im gewählten Zeitraum auftreten.

Literatur:
(1) O. Taraba: Le testing des défauts des cousinets glissants par une analyse spectrale de l’émission du champ d’ultracon généré par la perturbation du coussinet. The sevent International Conference on nondestructive testing, Wreszawa, Poland, 4 - 8 June 1973
(2) O. Taraba: Spektralanalyse des Ultraschallfeldes als Hilfsmittel bei der Diagnostik von Maschinen- und Maschinenteildefekten. Defektologie, 1972, DT Prag, S. 137 - 142 (Im Tschechischen)
INTRODUCTION

The piezoelectric transducer having a frequency response suitable for an ultrasonic device has been strongly required. This paper presents a synthesis of the acoustic matching networks for the transducer.

SYNTHESIS THEORY

Considering the acoustic terminal $T_s$ and applying distributed constant circuit theory to our synthesis, it is reduced to the determination of the driving point impedance $Z_r$ of the acoustic matching networks for the mechanical internal impedance $Z_m$ looking into the transducer from the terminal $T_s$.

$$Z_r = \frac{A_m p^{m+1} A_{m-1} p^{m-1} + \ldots + A_1 p + 1}{p = j \tan(\pi m/20_p)}$$

(1)

where $m$ is the number of layers, $S$ is the active area of the transducer, $0=f/f_p$ ($f_p$ is the half-wave resonant frequency of a piezoelectric layer), and $A_p$ (at which the imaginary part of $Z_m$ becomes zero) is the normalized center frequency of the transducer.

At the terminal $T_s$, the operating attenuation of a transducer is given as follows.

$$L = 10 \log \left( \frac{|Z_m + Z_r|^2}{4 \text{Re}(Z_m) \text{Re}(Z_r)} \right) \text{ (dB)}$$

(2)

When the prescribed values of $L$ at frequency points the number of which is equal to $m$ are given, we can solve Eq. (1) and (2) with respect to $A_m$ and $B_m$ and find the value of acoustic matching impedance for each layer by means of Richard's key theorem.

Fig. 2 shows an example of the transducers designed by the use of this theory.

REFERENCES

Afin d'étudier le rendement électroacoustique des transducteurs piezoelectriques dans leur condition normale d'utilisation, c'est-à-dire excités par de fortes puissances (500 W) entre 10 et 300 kHz, nous avons réalisé un impédancemètre automatisé. Celui-ci permet de tracer les cercles de Kennedy à partir desquels se déduit le rendement.

PRINCIPE DE L'IMPÉDANCÉMETRE

Le schéma équivalent d'un transducteur piezoelectrique peut se ramener à une résistance \( r(\omega) \) en série avec une réactance \( x(\omega) \), qui sont fonction de la pulsation \( \omega \). L'appareil permet de déterminer \( r \) et \( x \). D'une série de ces couples de valeurs, fonction de \( \omega \), nous pouvons tracer le cercle de Kennely des admittances

\[
G(\omega) = \frac{r}{r^2 + x^2} \quad \text{et} \quad B(\omega) = \frac{-x}{r^2 + x^2}
\]

Le secondaire d'un transformateur de puissance fournit une tension \( e \) au transducteur d'impédance inconnue \( Z \) et à une résistance ajustable \( R \) (fig. 1).

L'analyse du circuit donne (fig. 2) :

\[
\begin{align*}
\frac{e}{V} &= \frac{(R + Z)}{(R + r + jx)} \\
\angle AB &= \angle AM + MC + CB
\end{align*}
\]

Lorsque \( R \) est ajusté pour que \( |Z| = R \), M est alors sur 00'.

Nous en déduisons

\[
\begin{align*}
\frac{e}{V} &= R \left( \frac{1}{2} \cdot \frac{e^2}{V^2} - 1 \right) \\
x &= R \left( \frac{e}{V} \left[ \frac{1}{4} - \frac{e^2}{V^2} \right] \right)
\end{align*}
\]

où \( V \) est la tension aux bornes de \( Z \).

Donc \( r \) et \( x \) se calculent grâce à la connaissance de \( R \) et de \( e/V \).

L'appareil a été automatisé, la résistance \( R \) étant constituée d'une série de résistances progressivement décourcircuitées jusqu'à l'égalisation des tensions \( AM \) et \( MB \). L'appareil affiche alors la valeur de \( R \).

Les précisions relatives sur \( G \) et \( B \) sont de \( 5 \times 10^{-3} \). Elles sont suffisantes pour tracer le cercle de Kennedy.
INTRODUCTION

In the field of the generation of airborne ultrasound, we studied in a previous paper (1) the low power characteristics of a piezoelectric transducer based on a novel flexural vibrating stepped plate (2). Now this transducer has been improved for high-power applications, obtaining, with a better efficiency and directivity, a remarkable increase in the power capability and sound pressure levels in free field higher than 160 dB.

DESIGN CONSIDERATIONS

The new transducer has been designed on the basis of similar criteria to those of the precedent model, i.e., to achieve a good impedance matching to the propagation medium and high amplitudes of vibration and directivity.

The structure of the transducer is shown in Fig. 1. The points of maximum stress, in the radiating plate and in the mechanical amplifier, are water cooled in order to avoid fatigue failure and local overheating of the material. Also the piezoelectric ceramics in the sandwich are cooled by air.

EXPERIMENTAL

We have constructed two kinds of transducers: one has the radiating plate in aluminium alloy and the other in titanium alloy. In both cases, the vibrator is made of steel. The experimental results are summarized as: (1) the resonant frequency is of about 20 kHz, (2) the efficiency, measured with small amplitudes, is of the order of 80% for the aluminium plate transducer and 75% for the titanium plate transducer, and (3) the 3 dB beamwidth is of 5° in both cases. These transducers are capable of handling 100 W and 200 W of input power respectively. Figure 2 shows the axial pressure distribution of one titanium plate transducer for an input power of 200 W.

REFERENCES

(1) J.A. Gallego Juárez and G.R. Corral, Acustica (1973) 29, 234
(2) A. Barone and J.A. Gallego Juárez, J. Acoust. Soc. Am. (1972) 51, 953
INTRODUCTION

In ultrasonic measurement of materials and in other fields, circular flat transducers have been widely used. The nature of sound wave field generated by these in usual apparatus presents a complicated near-field pattern. One of the important approaches to investigate into the quantitative nature of this field is in computing derivative of the retarded potential of particle velocity of a system.

MODEL OF THE ACOUSTIC FIELD

Model of the acoustic field used in the computation, stands
1). Energy of the acoustic wave conserves.
2). Propagation of the acoustic wave is isotropical.
3). The circular flat transmitter vibrate coherently.

FUNDAMENTAL EQUATIONS

Fundamental equation for the particle velocity potential of a sinusoidal source $S$ can be represented as

$$\Phi(r,0,z,t) = \frac{\rho c}{2\pi} \int \frac{1}{r'} e^{i k r'} ds'$$

Particle velocity of the system is represented as

$$v_z(r,0,z,t) = -\frac{\partial}{\partial z} \Phi = R \varphi e^{i(\omega t - \psi)}$$

RESULT

Computational result of the system with two coaxial circular transducers is shown in Table 1. In previous article(1) $z\lambda/a^2$ is set to be a normalized figure for the axial distance of the transducers. But it can be realized by this table that the particle velocity (mean) of different $a/\lambda$ with same $z\lambda/a^2$ system has each $R$ value.

CONCLUSION

Numerical integration of the acoustic field by circular flat transducers is being carried out with the result that the distance scale of $z\lambda/a^2$ is not complete. Computation with these numerical method will give a solution to such a problem.

REFERENCE


<table>
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<td>13.7</td>
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</table>
INTRODUCTION

The sound fields of pulsed radiators of various types were calculated numerically and represented graphically. The analysis was made in the same way as in ref(1). In this resume, the results for a circular concave radiator are shown.

THEORETICAL

When a concave spherical radiator is vibrating with uniform velocity \( w_0 \), \( f(t) \) being the pulse shape, in a rigid plane wall, the sound pressure is approximately given by

\[
p = \frac{\rho c}{a-b} \left[ e^{-i\omega(t-b/c)} f(t-b/c) - \frac{1}{\pi} \int_{b}^{\infty} e^{-i\omega(t-r/c)} f(t-r/c) dr \right],
\]

where \( \rho \) and \( c \) are respectively the density and the sound velocity of the medium, \( a \) the radius of curvature of the radiator, and \( b, r_1 \), and \( \theta \) are shown in Fig(1). Eq(1) was derived by modifying the Schoch's method and applying the method in ref(1), under the assumption that \( R \), the radius of the radiator, is large relative to the wave length and to the depth of the concave surface.

NUMERICAL RESULTS

When the concave radiator with \( a = 4.5 \text{ cm}, \ b = 1.5 \text{ cm} \) is vibrating at 500 kHz in water, the pressure amplitude at \( t = 30 \mu \text{sec} \) calculated with eq(1) is illustrated in Fig(3) as a function of \( x \) and \( \theta \), the coordinates shown in Fig(1). The pulse shape was assumed to be as given in Fig(2). Fig(4) represents the same field in contour lines. These figures show clearly the peripheral waves from the outline of the radiator superposed upon the converging waves from the radiator surface.

REFERENCES

ULTRASONIC WAVE PROPAGATION ALONG A SOLID CYLINDRICAL WAVEGUIDE IMMERSED IN WATER

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ULTRASONIC WAVE PROPAGATION ALONG A SOLID CYLINDRICAL WAVEGUIDE IMMERSED IN WATER

In ultrasonic diagnostics puncture needles are used for guiding ultrasonic waves in the human body. In this paper wave propagation is investigated along a solid cylinder which in the first approximation is assumed to be a needle.

THEORY

Wave equations in solid and liquid can be fulfilled in cylindrical coordinates r, z by the following scalar functions /for axial symmetry/:

\[ \psi = I_n(\alpha r)e^{i\omega t} \]

where \( I_n \) and \( K_n \) are modified Bessel functions of first and second kind, respectively, \( \alpha = \frac{c_s}{c_w} \), \( c_s, c_w \) are the wave velocities in solid and liquid, \( \omega = \frac{2\pi}{\lambda} \), \( \omega_p \) is the phase velocity of the wave under consideration.

Equality of normal stress and acoustic pressure, zero of tangential stress and equality of radial displacements on the rod surface \( r = a \) are assumed as boundary conditions. Inserting the solutions /1/ in the boundary conditions the characteristic equation was obtained to be

\[ \frac{c_s^2}{c_w} \frac{K_n(\alpha a)}{I_n(\alpha a)} = \frac{I_n(\alpha r)}{K_n(\alpha r)} + \frac{1}{2\alpha} \left( \frac{\alpha^2}{\omega^2} - \frac{1}{c_w^2} \right), \]

where \( \alpha, \omega \) are the densities of the rod and liquid. Numerical solution of /2/ for a steel rod 1 mm in radius, immersed in water, for frequency 5 MHz gave stresses \( \sigma_r, \sigma_z \) and acoustic pressure \( p \) shown in Fig. as function of radius \( r \).

Wave velocity was only a little smaller than in water /c_w/c_w = 0.999985/.

EXPERIMENTS

Using the pulse echo technique and a steel needle of 0.75 mm radius, immersed in water, with one end inserted into a central hole made in a piezoelectric disc transceiver we obtained an echo from the second end of the needle. The wave velocity was practically the same as in water. Inserting into the needle an additional rod of 0.5 mm radius we obtained an additional echo from the end of the rod thus showing that the wave under consideration penetrated also the wall of the needle.

CONCLUSIONS

It exists a wave guided by a solid rod immersed in liquid which propagates with the velocity only a little smaller than in liquid. The numerical results obtained for the rod can be also applied for a needle if its wall is thin and/or the frequency is high enough.

REFERENCE

INTRODUCTION

It was studied the possibility of getting sonic stepped structures equivalent to sonic transformers of revolution with axial symmetry, and in whose the generatrix is a continuous function of the axial coordinate.

THEORETICAL BASIS

The theoretical starting point considers the different phenomena presented in the plain wave propagation through the stepped structures; the more or less influence of any of these phenomena was manifested by a different starting equation relating the variables $r$, $\phi$, and $l_i$, $i = 1, 2, 3, \ldots, n$; in our work $n = 12$ (Fig. 1).

EXPERIMENTAL TECHNIQUES

Once a given concentrator of continuous generatrix was fixed, a set of different equivalent stepped concentrators were designed; in some cases, up to three distinct series of concentrators (different starting equation). The values of the magnification, and resonance frequency of the stepped transformer were compared with those of the "continuous" structure. In all cases the resonant frequency was computed from the nellygrams; the magnification and frequency response was derived by means of adequate pick-ups.

RESULTS AND CONCLUSIONS

From a sample of about 300 stepped structures we could conclude:

1) The expression giving the value of the pressure in any point of the stepped transformer gave the best approach to the "continuous".

2) In all cases the magnification has been function of "n" and both the stepped and the continuous values of the magnification approach each other when "n" increases.

REFERENCES


CUT-OFF FREQUENCIES OF WAVEGUIDES OF COMPLICATED DOUBLY CONNECTED CROSS SECTION

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INTRODUCTION

The present study deals with the determination of the lowest cut-off frequency of cylindrical (polygonal) waveguides with an inner regular polygonal (circular) boundary in the case of soft walls.

The problem is mathematically equivalent to the propagation of electromagnetic waves (TM modes).

METHODOLOGY

Two methods are used to obtain the desired eigenvalue.

First a conformal mapping approach coupled with a variational technique is used.

Then, the fundamental eigenvalues are computed using a very simple empirical approach. The results obtained using both techniques show, in general, very good agreement.

The empirical approach is then used in the case of polygonal waveguides with an inner circular boundary and the results agree favourably well with those already published in the open literature.

The empirical approach was suggested by Randall (1) as a procedure for estimating the optimum over-relaxation factor in the S.O.R. method for solving, over irregular regions, equations of the type:

\[ \frac{\partial^2 \psi}{\partial x^2} + \frac{\partial^2 \psi}{\partial y^2} + g( x, y, \psi, \frac{\partial \psi}{\partial x}, \frac{\partial \psi}{\partial y}) = 0 \]

RESULTS

Figures 1 and 2 depict a comparison of results for polygonal waveguides with circular, concentric, inner boundaries.

REFERENCES

(1) T. J. Randall, Computer Journal (1968), 10, 400.
AN ULTRASENSITIVE ULTRASONIC SYSTEM WITH APPLICATIONS TO NONDESTRUCTIVE EVALUATION, MEDICAL DIAGNOSIS AND STUDIES OF NON-LINEAR EFFECTS

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INTRODUCTION

We have developed an ultrasensitive ultrasonic system which can provide orders of magnitude improvement in signal-to-noise ratio over conventional devices. Major components of the system include (a) real-time A-scan averaging, (b) real-time B-scan averaging, (c) pulse compression, (d) dynamic focusing and (e) transducer matching. The system is being applied to problems in material inspection, medical diagnosis and non-linear ultrasound.

THEORETICAL BASES

Ultrasonic signal averaging is accomplished by repetitive pulsing and adding the results of all the scans in a coherent fashion. An ideal averager integrating over N scans will improve the sensitivity by a factor of \( \sqrt{N} \) over single scan detection. Pulse compression techniques allow more average power to be put into the material without exceeding the peak power limits of the transmitter. A long pulse is transmitted, which, on reception, is compressed to a time interval compatible with the desired range resolution. Dynamic focusing allows more power to be transmitted and received from the region of interest. Matching the transducer to the material under study increases transducer efficiency in both transmission and reflection.

EXPERIMENTAL TECHNIQUES

High-speed digital techniques are used to average the A- and B-scan signals. The pulse compression system (Parks and Linzer, Proc. SPIE, to be published) is based on linear f.m. chirp techniques and incorporates frequency-dependent time-gain compensation and a SAW device for compression. Dynamic focusing is achieved along the transducer axis by means of an electronically-focused concentric annular array. One-quarter wavelength transducers are used to match the transducer to the material under study.

RESULTS AND CONCLUSIONS

The ultrasensitive ultrasound system is being applied to a number of significant materials and biomedical problem areas. In nondestructive evaluation (NDE), the system is being used to penetrate nuclear reactor components made of coarse-grain austenitic steel and to increase the sensitivity of inefficient but otherwise highly-desirable transducers such as contactless electromagnetic transducers (Maxfield, et. al. Proc. 1976 IEEE Ultrasonics Symposium, to be published). Applications to medical ultrasonics include examination of deep-lying structures and overcoming increased tissue losses when higher frequencies are employed to obtain increased resolution. Finally, we are exploring the application of our system to the area of non-linear ultrasound, both from a fundamental standpoint and as a new device for nondestructive characterization of both biological and non-biological material.
INTRODUCTION

It is necessary to know the overall sensitivity of ultrasonic diagnostic equipment or to be able to set it at any desirable sensitivity, which is necessary for the quantitative diagnosis or for the preservation of the equipment. Since adjustable range of the sensitivity of diagnostic equipment is not so wide, considerably small reflection is required for the test piece in order to carry out the test under nearly actual conditions. For this purpose, a method by using a reflection of steel ball was investigated and verified to be practical (1).

PRINCIPLE AND OBJECT OF SENSITIVITY MEASUREMENT

Target strength $R_p$ of a steel ball of radius $a$ is expressed as shown in eq. (1).

$$ R_p = \frac{2a}{x} |\frac{1}{2}| $$

where $x$: distance between steel ball and sound source, $|\frac{1}{2}|$: Stenzel's function. $|\frac{1}{2}|$ is a function of $ka$ ($k$: wave length constant) and approaches the value 0.5 ($ka > 1$). Deviation of $|\frac{1}{2}|$ value from 0.5 is less than $\pm 20\%$ ($ka > 4$).

Fig. 1 shows an example of test object. Stainless steel ball bonded at the top of injection syringe is set at the bottom of a vessel. In this figure, $D$ is diameter of transducer and $\lambda$ is wave length.

RESULTS AND CONCLUSIONS

Fig. 2 shows experimental results where the distance between transducer and steel ball is 200 mm constant. Measured values of target strength coincide well with theoretical values.

In actual measurement, steel ball is set at a fixed place far from the last maximum of sound pressure along beam axis. When echo amplitude from steel ball becomes fixed level by adjusting the attenuator of the equipment, overall sensitivity of this condition can be defined from eq. (1) by removing minus sign from calculated value in dB. Any sensitivity can be obtained by adding the change of variable attenuator to the above value.

REFERENCE

(1) M. Ide, Japanese Journal of Medical Ultrasonics (1976) 3, 45
NEW DEVELOPMENT IN DOPPLER ULTRASOUND FLOWMETERS

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INTRODUCTION

One of the major problems associated with the use of Doppler shifted ultrasound for blood velocity information obtained transcutaneously from vessels close to the skin is the uncertainty, in some cases, of the validity of the haemodynamic parameters which are being obtained. This is true of both continuous wave and pulsed Doppler systems, because the current methods of analysing and extracting analogue signals of velocity and acceleration from the Doppler shift are subject to considerable error.

THEORETICAL BASES

A more complete approach, taking into account physical considerations of scattering and attenuation and geometrical constraints of insonating blood vessels through living tissues, is presented which gives useful data about the best frequency of insonation and criteria necessary to compensate for unavoidable attenuation.

EXPERIMENTAL TECHNIQUES

A new technique of analogue processing is proposed which uses phase-lock-loop techniques to derive haemodynamic data from the Doppler spectrum.

RESULTS AND CONCLUSIONS

This signal processing system will yield (i) a fully directional instantaneous mean velocity (ii) a fully directional instantaneous peak velocity and (iii) an indication of the cross sectional area of the insonated vessel. The performance of the instrument has been evaluated in vitro and its application in vivo has been assessed in the investigation of both peripheral and cardiovascular disease in man.

REFERENCES

When designing ultrasonic surgical instruments one has to choose the transducer material and the material and shape of the velocity transformer, to evaluate the output power and necessary limits of the automatic frequency and amplitude control of the power supply generator. The decision depends on the mode and the amplitude of working tip vibrations, required for the specific operation and on the mechanical resistance of the acoustic load; general requirements concerning miniaturisation and simplicity of the instrument should also be taken into account.

The value of specific load resistance for 44-Kc-instruments, measured by electric impedance method, varies for different operations from $10^3$ to $10^5$ dyn.sec/cm$. In most cases the load resistance amounts for 30 - 200% of the internal mechanical loss resistance of the instrument. The frequency shift under load conditions does not exceed 0.25-0.5%.

Comparison of velocity transformer materials speaks in favor of some titanium alloys with high fatigue strength and low mechanical losses. As experiments and calculations show the most high efficiency of the instrument is achieved when using the catenoidal shape transformer, but the Gaussian-shape transformer has the smallest losses and deformations at a given tip amplitude.

The amplitude of 44-Kc transducer made of piezoceramic or ferrites usually does not exceed 2 microns and is always limited by the mechanical strength. The strength of metal magnetostrictive materials admits the amplitude limit of about 8-10 micron, but in case of liquid cooled systems the nonlinearity restricts the permendur and nickel transducers amplitudes correspondingly by 5-7 and 2-3 microns.
INTRODUCTION

Several investigators have determined thresholds for production of ultrasonic focal lesions in mammalian central nervous system tissue over a broad range of the acoustical parameters, i.e., intensity, frequency, and duration of exposure (1-2-3-4). These studies have provided valuable information as to levels for structural damage to tissues, the safety of medical applications of ultrasound, and the mechanism of the action of ultrasound on tissue. Similar data are being obtained on mammalian liver, kidney and testicle. The present study is the first to determine thresholds for structural damage in a tissue other than central nervous system and is thus providing valuable data for comparison to brain thresholds. In addition, the lesion volumes are being measured for relation to the dose parameters and comparison to results for similar measurements on brain (5).

EXPERIMENTAL TECHNIQUES

The animal is anesthetized, the organ to be irradiated is mobilized and exposed, and then irradiated at several sites by a focused ultrasonic field. Each site is irradiated with a different intensity, but a fixed duration and frequency. Twenty-four to seventy-two hours later the animal is sacrificed and the tissue is fixed in a 10% formalin solution. The tissue containing the irradiation sites is sectioned and stained with hematoxylin and eosin. Each site is then examined under the light microscope for presence or absence of structural damage.

RESULTS

Thresholds for ultrasonic focal lesions have been determined at 2 and 6 MHz in mammalian liver, kidney and testicle. For exposure durations of 1 to 60 seconds these thresholds are similar to those in brain (6). The threshold intensity at a given exposure duration is higher for kidney than for liver and higher still for testicle. These differences can be explained on the basis of a thermal model for the action of ultrasound on the tissues (Frizzell et al., to be published). The thresholds are relatively independent of frequency.

Data on thresholds and the relation of lesion volumes to the dosage parameters in mammalian liver at 1 and 3 MHz and exposure durations less than 1 second are discussed.

REFERENCES

(6) Presented in part at the 87th meeting of the Acoustical Society of America, New York, April, 1974.
INTRODUCTION

Etude expérimentale de la propagation des ondes élastiques dans des milieux hétérogènes du type composite et stratifiés, à des fréquences ultrasonores de quelques mégahertz et à diverses températures.

METHODE EXPERIMENTALE UTILISEE

Méthode des échos d’impulsions dans la variante dite de comparaison de phase (superposition), avec un chemin acoustique fixe déterminé par l’échantillon ou des cales d’épaisseur, dans le cas d’une matrice liquide.

L’intervalle de fréquences est de l’ordre de quelques mégahertz, entre 1 et 10 MHz.

RESULTATS

Nous présentons les premiers résultats de notre étude. Ces résultats concernent la mesure de la célérité du son dans des milieux constitués d’une maille métallique noyée dans une matrice du type résine, à des températures pas trop éloignées de l’ambiante.

Les matériaux sont choisis de façon à éviter la présence de la dispersion visco-élastique, le but de ce travail étant uniquement l’étude de la dispersion dite géométrique.
INTRODUCTION

Parameters of living systems and biopolymer water solutions vary with the ultrasound intensity and are characterized by extremum in the region of 12 0,2 wt/sm² at the frequency of 0,8 - 1 Mc.

RESULTS AND CONCLUSIONS

It was shown that dielectrical losses of skin and muscles of warm-blooded animals reach their maximum at 1 wt/sm². Active and reactive constituents of impedance change reversibly provided the ultrasound intensity is lower than 1 wt/sm². At higher values of intensity the change prove to be irreversible. The specific viscosity, the number of SH-groups titred by HgCl₂, as well as the optical density of actinomycin solutions in the highest at the intensity of 1 wt/sm². At the same intensity ultrasonic luminescence of blood serum is also maximum. The mechanism of physico-chemical action of ultrasound on water is different at higher and lower than 1 wt/sm² intensities.

The similar phenomena (such as the influence of ultrasound on motoric and secretory activity, on the content of RNA in the liver tissue, on the permeability of membrane and so on) have been mentioned in the reports of other investigators.

Now the reasons for the above phenomena are discussed in terms of physico-chemical action of ultrasound. The appearance of extremum at 1 wt/sm² is accounted for by complex ultrasound action including different factors (chemical, mechanical, thermal etc.) whose relative efficiency depends in various ways on the ultrasound intensity.
ACOUSTICAL NUCLEAR RESONANCE IN DOPED DIELECTRIC CRYSTALS

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Komashnja, V.L.

INTRODUCTION

The nuclear quadrupole spin-phonon coupling be due to dynamic electric field gradients resulting from the relative displacement of ions (atoms) of the ideal crystals lattice in the acoustical wave. Stronger gradients can arise in real crystals in the vicinity of lattice defects, from which the local excitations will spread over the entire crystal by means of the spin diffusion [1]. This hypothesis is confirmed by theoretical calculations [2] and experimental studies carried out on doped crystals.

THEORETICAL BASIS

In case of the "defect" mechanism of the quadrupole coupling and small diffusion barriers, the factor of the acoustical saturation of the quadrupole transitions (QT) is determined by the equation [1]

\[ Z = \left\{ 1 + 16 \pi N T D^{2/3} |C_q w_i|^{1/4} \right\}^{-1}, \]

where \( N \) is the defect centers concentration, \( D \) the spin diffusion coefficient, \( T \) the spin-lattice relaxation time, \( C_q(w_i) \) the QT-probability, \( w_i \) the resonance frequency. On a rough estimation the concentration \( N \) sufficient for the absolute prevailing of the "defect" mechanism over the "crystalline" one is of the order of \( 10^{25} \), which corresponds to relatively pure crystals.

EXPERIMENTAL TECHNIQUE

The studies of the acoustical saturation of the QT were performed at room temperature with the pulse method by comparing the nuclear induction signals before and after the application of 7 MHz sound.

RESULTS AND CONCLUSIONS

The saturation of the QT with \( m=42 \) of the nuclei Li\(^7\), Na\(^22\), Al\(^27\) has been studied in crystals: LiF doped with Co\(^{2+}\) and Mg\(^{2+}\); NaF doped with Cu\(^{2+}\) and Ca\(^{2+}\); Al\(_2\)O\(_3\) doped with Cr\(^{3+}\) and NaCl with various impurities; also pure crystals. In all crystals the enhancement of the quadrupole coupling by the iron group ions was found. In the NaF crystals this takes place not only in the acoustical nuclear resonance but also in the spin-lattice relaxation of the Na\(^{2+}\) nuclei. In the crystals Al\(_2\)O\(_3\):Cr\(^{3+}\) the enhancement effect exists also in case of the saturation of the QT by the alternating electric field. Thus iron group ions are sources of the amplified local dynamic gradients in the "defect" mechanism. The presence of their electrons density on ligands creates the co-valent contribution to the interionic bonding sensitive to the changes of interionic distances in an acoustical wave or electric field.

REFERENCES

IV. Physical acoustics

L. UNDERWATER ACOUSTICS
STEERED PLANAR NEARFIELD CALIBRATION ARRAY

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INTRODUCTION

The Trotz planar nearfield calibration array (NFCA) (1) is a two-dimensional array of small, reciprocal hydrophones whose responses are shaded to produce a nearly uniform plane wave throughout a volume in the nearfield of the array. When a transducer to be calibrated is placed in the plane-wave volume, the signal received by the NFCA is proportional to the farfield pressure distribution of the transducer in the direction opposite to the plane wave. Existing planar NFCAs are shaded so that the plane-wave direction is normal to the array. Determination of the farfield distribution of a transducer by use of these NFCAs requires that the transducer be rotated within the plane-wave region of the array. New shading coefficients are sought that allow the nearfield plane wave to be steered to large angles from the normal to the array.

THEORETICAL RESULTS

The basis of the NFCA concept is the NFCA reciprocity principle. A mathematical statement of the NFCA reciprocity principle was derived from the Helmholtz integral formulation in a previous paper (2). A numerical procedure based on this mathematical statement was developed to calculate shading coefficients for a planar NFCA that are optimum in a least squares sense. Application of this procedure produced complex shading coefficients suitable for steering the plane wave up to 70 degrees from the normal to the array.

EXPERIMENTAL RESULTS

A synthetic steered planar NFCA was created by using a single vertically suspended NFCA line shaded to produce uniform outgoing cylindrical waves throughout a volume in its nearfield. Both the amplitude and phase response of the line to a large piston transducer were measured at horizontal positions corresponding to locations in a rectangular planar NFCA. The responses were then computer processed using the appropriate line shading coefficients to produce the farfield pressure distribution for the azimuthal directions -70° ≤ φ ≤ 70°. Repeating the procedure with the transducer rotated 180 degrees (equivalent to originally having a second line present on the other side of the transducer) produced the azimuthal distribution for 110° ≤ φ ≤ 250°. The calculated results are in good agreement with the farfield pressure distribution measured directly in the farfield.

CONCLUSION

Element shading coefficients suitable for steering a planar NFCA up to 70 degrees from the normal to the array have been obtained and successfully tested experimentally. In addition this study demonstrated that two vertical lines of hydrophones can be used to determine a large part of the azimuthal farfield pressure distribution of a sound source as it moves slowly between the two lines.

REFERENCES

PASSIVE SONAR "CHAYKA" FOR DETECTION OF BIOLOGICAL NOISE

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INTRODUCTION

Passive sonar "Chayka" is intended for the study of bioacoustic fields at different seas and to work out many questions related to the problem of using of passive location principle with the aim of detection of sea fisheries subjects.

EXPERIMENTAL

The principle of vector addition of two video signals on the sonar screen has been used in passive sonar "Chayka" for the location of noise-producing subjects. The level of signals is depended upon the angle at which the signal arrives to acoustic antenna. It contains four sonic detectors which are orientated for four directions: bow, stern, port side, starboard. Direction pattern of acoustic antenna in any of the four channels has a cosine form in 4 to 30 kc frequency band. Received voltages are amplified and supplied to the plan position indicator causing angular variance of the beam on the screen of CRT in appropriate direction. There is also aural indications both with using of frequency conversion and without it. Passive sonar "Chayka" guaranties simultaneous survey of underwater noise within 360°.

THE PRINCIPAL RESULTS

By means of passive sonar "Chayka" at the marine examination was located the source of both puls and audio signals. Besides was located choral biological noise produced by shrimps. Passive sonar "Chayka" bearings the noise of inshore shrimps with the highest intensity in the range of 25-30 kc to two miles. It can identify the sources of known biosignals (dolphins, shrimps, fish and so on) by using sound indication with frequency conversion.

The noise of working auxiliary engine of the ship does not disturb to observing of various sounds produced by dolphins very actively: series of location pulses, singing sounds, whistle, squeak, whine and so on. They was complexity modulated sounds having the most intensity in the range 25-30 kc. Especially high level of the whistle had been while dolphins was feeding on the fish school. Such behaviour of dolphins may be used for the interest of fisheries: to define by means of passive sonar the bearing to the whistle of feeding dolphins, to come to fish school and to catch out it.

The development of bioacoustic investigations is of great importance for marine fisheries. Passive sonars will be a useful addition to the existing fishfinders, especially to those subjects of marine fisheries which lack efficient finding equipment.

REFERENCE

INTRODUCTION

Sonar system performance is influenced significantly by the anisotropy of sea noise. The anisotropy is of different character at the low frequencies where distant shipping noise prevails due to long range propagation and at the somewhat higher frequencies where surface noise is dominant. This paper considers the array gain variations observed for idealized isotropic and dipole noise as well as a horizontally-omnidirectional noise model having vertical directivity due to distant shipping and surface contributions. Array gains are presented for thick horizontal circular discs (cylinders) that form continuous unshaded distributions of various sizes and vertical steering directions. One of the thickness options considered is that of an out-of-phase dipole that has a cardioid pattern. This is an extension of previously-reported work (1) relative to rectangular horizontal planar arrays. That work describes the environmental aspects and presents sample curves of sound velocity profile and vertical distribution of ambient noise that are not presented here.

ARRAY DIRECTIVITY

The array gains were calculated by integrating the product of the array and noise directional characteristics. The array is electrically steered over a wide range of vertical angles near the horizontal plane (called edgefire) to accommodate incoming signal arrivals from various ranges. If the cylinder is very thin, there are two beams, symmetrically above and below the horizontal plane, and the principal arrivals from distant signals and noise are received with one steering angle. In order to separate the descending and ascending arrivals and/or increase array gain, additional vertical directivity is achieved either by increasing the height of the cylinder or by the use of the cardioid element.

ARRAY GAIN RESULTS

The effects due to array steering are considered up to 30° above and below the horizon. The gain has a minimum, particularly noticeable at low frequencies, at those angles for which distant shipping noise is greatest. When the signal is due to a distant source, the required steering angle coincides with the minimum gain case. However, the best performance is obtained at or near that steering because the signal gain deteriorates rapidly if the beam is not aimed at the signal. If the signal source is at a relatively short range such as that due to a bottom bounce arrival, the signal angle can differ appreciably from that of the dominant noise so that higher array gain is achieved. The gain is generally proportional to frequency squared for steering angles well removed from the horizontal plane and more nearly linear dependence otherwise.

The various noise models yield different array gains. While isotropic and surface dipole noise are simple models that lead to approximate array gain values, system performance changes so significantly with gain that the final array analysis should be done with the best possible noise model.

The change of gain with vertical structure of the array is considered.

REFERENCES

INTRODUCTION

A procedure is developed for determining the optimal upper operating frequency for a passive broadband detection or localization sonar with low sidelobes. This optimal upper frequency may then be used to determine optimal array aperture as a function of number of elements. The approach employed here is somewhat different than conventional procedure which typically assumes aperture is fixed and then determines appropriate system bandwidth and finally the requisite number of elements. The procedure described herein is motivated by the recognition that for modern high gain sonars cost is significantly dependent upon the number of element channels. Thus we desire to minimize the number of elements needed to achieve a given sonar performance objective and to describe the trade-offs in performance versus number of elements along the optimum contour in aperture/number-of-element parameter space.

THEORETICAL BASES

Under the assumption of optimal filtering it can be shown that sonar performance is proportional to the integral

$$I = \int_{f_{\min}}^{f_{\max}} f^P \frac{S}{N} G^A T^{-2} df$$

where $p=0$ for the detection function and $p=2$ for the bearing determination function and where $S$ is the target source spectrum, $N$ is the single element noise spectrum, $G_A$ is array gain, and $T$ is the propagation loss for the range and ocean environment of concern. The side lobe constraint enters by limiting the frequency range of integration to $f< f_{\max}$ where $f_{\max}$ is the frequency at which high level grating lobes of the array first appear.

As a instructive example, assume that the noise field is isotropic and that a planar array is under consideration, in which case

$$G_A = n f_{\max}$$

where $n$ is the number of array elements. Consequently, $f_{\max}$ should be choosen so that the function

$$I(f_{\max}) = \frac{1}{f_{\max}} \int_{f_{\min}}^{f_{\max}} f^2 \frac{S}{N} G^A T^{-2} df$$

is maximised. The integral above is a monotonically increasing function of $f_{\max}$. The increase with $f_{\max}$ will be rapid until $f_{\max}$ exceeds the frequency at which transmission loss increases rapidly with frequency due to volumetric attenuation. Beyond this effective transmission cut-off frequency the integral will not vary significantly with $f_{\max}$ and the function $I(f_{\max})$ will decrease due to the $f_{\max}$ factor in front of the integral.

RESULTS AND CONCLUSIONS

The optimal upper operating frequency will typically be in the region where transmission loss starts to increase rapidly with frequency. In fact we could sensibly define the "effective transmission cut-off frequency" as the value of $f_{\max}$ determined by the described optimization process. The optimal upper operating frequency and, hence, optimal array aperture for a given number of elements is dependent upon a) the sonar function of concern and b) the spectral characteristics of the target, the signal transmission path, and the sonar self-noise. This design procedure emphasizes the necessity for a systems approach to sonar array design.
SIDELOBE LEVEL REDUCTION OF THE PROJECTORS FOR ACOUSTICAL HOLOGRAPHY

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INTRODUCTION

In the underwater viewing system using acoustical holography by means of a multi-beam scanning, it is necessary for the projectors to have high uniformity and to insonify only a given area. The authors applied the ceramic-dicing and amplitude-shading techniques to the projectors to realize high uniformity and reduce sidelobes, and used those for the initial purpose.

EXPERIMENTAL RESULTS

The ceramic-dicing technique was applied to the lead-zirconate titanate disk transducers, 29mm in diameter by 5mm thick, to eliminate unwanted resonances and decrease gain-phase deviations. Depth of mutually-perpendicular grooves was 4mm, and these grooves were spaced 3mm or below a half wavelength apart. This resulted that unwanted modes of vibration were not revealed between 100kHz and 400kHz. These transducers were resonant near 300kHz.

A technical advantage of dicing is that many poles of the diced transducer can be used to the array elements. Because the performance of the poles is exactly similar, controlling of supply voltages on those, the shaded velocity distributions can be obtained. Therefore, it is easy to realize further reduced sidelobes performance in the directivity pattern.

The experimental shading coefficients were determined by modifying Gaussian distribution as shown in fig.1. This resulted that sidelobes were -33dB or more below the main lobe at 200kHz as shown in fig.2.

CONCLUSIONS

It is easy to realize further reduced sidelobes performance of the diced ceramic transducer by means of the amplitude shading. The authors realized that sidelobes were -30dB or more below the main lobe.

16 elements of such a transducer were manufactured and used for the elements of the 4 x 4 projector array in the underwater viewing system. The standard deviation of responses is 1.0dB for those projector elements.

REFERENCES

(1) K. Nitadori; Acoustical Holography Vol.6, P507
(2) R.L.Cook; IEEE Trans. SU-19, 4, (1972) P444
Arrays for many sonar applications have customarily been designed for equal side-lobe levels, due as much to the convenience of the well known analytical Dolph-Chebyshev technique, which yields the narrowest possible beam for a given degree of uniform minor lobe suppression, as to any specific requirement. Though no direct approach existed for the case of planar arrays, cross-multiplying the coefficients for two orthogonal line arrays enables usable results to be obtained for rectangular arrays. The method has been extended to the case of difference patterns, but the side-lobes are again not uniform. Some problems, for example arrays using a hexagonal or hexagonal grid, remained intractable.

The relationship between the array output and the contributions from the individual elements of an arbitrary array is essentially a non-linear one. However, for symmetrical arrays of fixed geometry the contributions from pairs or groups of elements symmetrically disposed about the centre may be combined using a common shading coefficient, so as to cancel out one of the two quadrature components, and the relationship becomes linear, so that linear programming techniques may be used.

One of us has recently made use of an algorithm intended for linear approximation or curve-fitting (1). It obtains coefficients that minimize the maximum error from the desired response in a large number of specified constraint directions, this desired response being specified as zero in the controlled minor lobe region and at a point on the "skirt" of the main beam, and some finite value at the desired peak. It has solved a number of problems for symmetrical, though not necessarily equi-spaced, arrays, including the honeycomb array.

In its original form the procedure produces a uniform peak minor lobe level (though in the case of planar arrays not all the lobes necessarily reach this level). The technique has recently been expanded to the case where some other minor envelope is desired.

Figure 1 shows one example, for a 12x12 element truncated array, in which the outer lobes were specified to be 20 dB lower than the inner ones. The inner lobes are at -35 dB, compared with -38 dB for the uniform case in reference 1.

Currently we are exploring the use of non-linear goal programming techniques (2) for cases where symmetry does not exist, or in which element position or phasing must be varied.

REFERENCES

UNDERWATER ECHO DISTORTION CAUSED BY SOLID DIOPTRICS

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INTRODUCTION
This paper studies the phase change suffered by a train of plain waves, traveling through a fluid medium, when is reflected by a series of plain parallel dioptrics located between mediums of different acoustic impedance. This work covers both normal and oblique incidence.

THEORETICAL BASES
The shape of the reflected wave pulse is calculated by superimposing all the successively reflected waves produced by all the dioptrics that form the reflective system; this calculation is made considering the amplitude and phase, and also, in the case of oblique incidence, the geometrical setup as well as both the relative position of the emitter/hydrophone system and the dioptric.

EXPERIMENTAL TECHNIQUES
All the experiences were done in the hydroacoustic Laboratory of the Institute of Acoustics. The emitted sonar pulse was, after reflection, picked up by a hydrophone, then amplified, recorded and digitized and stored. In a X,Y recorder we were able to reproduce, if needed, up to cycle of a 40 khz signal in a length of 40 mm record.

RESULTS AND CONCLUSIONS
The X,Y graphs, representing the non reflected and reflected pulses were compared. This experimental method has permitted to show the good agreement between theory and experience, and specially the dependence of the reflected pulse phase on

1°) The acoustical characteristics of the layers of the dioptrics (Acoustical impedances, velocity of propagation).
2°) Distance between dioptrics.
3°) The angle of incidence.
4°) The relative distance between emitter and hydrophone and the first dioptric.

REFERENCES
Vol. 34, No 10, pp. 1582

de Lustrac, J.
Carbo, R. - Acustica, 1968,
Vol. 20, No 1, pp. 21

Phase change vs. angle of incidence
(Dioptric: perpex-water d/A = 0,077, f/A = 34)
INTRODUCTION

The scattering characteristics of submerged objects differ greatly before and after they are covered with sound absorbing layers. The few papers that have theoretically dealt with acoustic scattering from sound-absorbing structures have treated the structure as if it were elastic (i.e., non-absorbing) and then at the end they have introduced an "ad-hoc" complex wavenumber with an empirically determined imaginary part to account for losses. In our present work, however, the complex wavenumber in the coating is related directly to the elastic and viscous material parameters of the acoustic absorber, via a basic and general viscoelasticity formulation we have developed (1) on the basis of the Kelvin-Voigt model.

THEORETICAL BASES

We further (2) study the scattering problem present when a plane wave is normally incident on an infinite hollow elastic cylinder covered with a bonded layer of viscoelastic and sound-absorbing material. The structure contains air in the cavity and it is immersed in water. We obtain the classical normal-mode solutions of the problem in all four media. We then compute the monostatic (i.e., sonar) and some bistatic cross-sections in the exterior fluid only (i.e., the water).

RESULTS AND CONCLUSIONS

A computerized parametric study covering a wide range of behavior with a fine grid is presented. We display many cases (Fig. A) showing the effects of shell and coating thickness variations, and more cases showing the effect of varying the coating's viscosity. We quantitatively determine oscillations in the sonar cross-sectional values due to resonances in the layers, and show how the higher viscosities damp-out the oscillations and reduce the amplitudes to smaller values, down to an optimum point, beyond which the reverse effect begins to take place. Most of the results are for the low frequency range $\omega c < 20$ but some high-frequency results are shown.

REFERENCES

An infinitely long, circular cylindrical elastic shell containing an acoustic medium is immersed in a second acoustic medium of different properties and of infinite extend. An incident plane pressure pulse, whose front is parallel to the axis of the shell suddenly strikes the shell. The resulting transient two-dimensional acoustic field is studied. The exact solution is obtained by using dual integral transforms - Laplace with respect to time and Fourier with respect to angular coordinate. The modified Bessel functions appearing in formal solution are approximated by Olver's and Langer's asymptotic expansions. Inverse Fourier transform is made the first. For large real and positive values of the Laplace transform variable zeros of denominator of the integrand are found. The Laplace transform of radiated and diffracted pulses are found by residues technique. Steepest descent method is used for finding the Laplace transform of reflected pulse. Two terms and estimate are kept in all calculations. The inverse Laplace transform is performed by using the theorems of convolution and translation and exact table inverses. Equations of reflected, radiated and diffracted pulse fronts are given. The critical angles of acoustic pulse fronts duplication are defined. Are found discontinuities of pulses at wavefronts and their changing along the front line and at any distant from it. The influence of properties of the shell and internal fluid on sound pulses of these three types is analysed. Obtained results are valid only in the neighbourhood of the pulse front.
ON THE IDENTIFICATION OF SHELLS BY MEANS OF ECHO-SIGNALS

INTRODUCTION

Let a thin spherical or cylindrical shell be surrounded by the infinite acoustical medium. The possibility of determining the parameters of the shell and the liquid inside the shell by means of the theoretically calculated echo-signals generated by an incident sinusoidal pulse is investigated. The determination of shell parameters with vacuum inside was analysed in (1).

THEORETICAL BASIS

The echo-signal from the shell was calculated with the aid of Fourier and Watson transforms. The Timoshenko-type shell theory and the linear wave theory for both the surrounding and filling liquids were used. The formulae for calculating the echo-signals are presented in (2,3).

The calculated echo-signals consist of many sequences of echo-pulses that may be classified as reflected, radiated, transmitted, and radiated-transmitted pulses. Every sequence of pulses except the reflected one consists in its turn of many series of single pulses. The amplitude and the arrival time of every single pulse are taken as the main parameters of echo-signals.

RESULTS AND CONCLUSIONS

The correlations between the parameters of echo-signals and the parameters of the shell and of the liquid inside the shell were established. The main correlations are as follows. The arrival time of the reflected pulse does not correlate with any parameter of the shell. The arrival time of the transmitted pulse depends only on the velocity of sound of the liquid inside the shell and the arrival times of radiated pulses depend generally on the dilatational wave velocity in the shell material. The amplitude of the reflected pulse at the low frequency of incident pulse depends as well on the geometry of the shell as on the density of the shell material and on the impedance of the liquid inside the shell. At the high frequency of the incident pulses it depends only on the geometry of the shell. The amplitudes of the transmitted pulses depend on the density of the shell material, on that of the liquid inside the shell and on the frequency of the incident pulse. There exists a frequency of the incident pulse by which not all the series of the transmitted pulses will be generated. The amplitudes of the radiated pulses correlate with all parameters of the shell and the liquid inside the shell.

As a result, it follows that the echo-signal from a spherical or cylindrical shell contains sufficient information for determination the parameters of the shell itself and the liquid inside the shell.

REFERENCES

(1) Я.А.Метсавъзр, Акуст. ж., 1975, 21, 141; (2) Я.Метсавъзр, Д.Пикк, Изв. АН Эст. ССР. Физика.Математика, 1976, 25, 26; (3) Я.А.Метсавъзр, Д.Пикк, Прикладная математика и механика, 1976, 40, 648.
The response of a submerged elastic body to an incident acoustic wave is usually represented as a series of normal modes. Each mode may resonate at a set of eigenfrequencies characteristic for that mode. It is shown that these resonances are caused by the coincidence of the phase velocities $c_\phi$ of the waves that form the standing-wave resonance, with the velocities $c_v$ of any of the circumferential surface waves (Rayleigh, Stoneley, Whispering Gallery, or Franz waves) which constitute the diffracted field (see Fig. 1). Furthermore, the surface waves themselves are shown to undergo a resonance when satisfying the coincidence condition.

The mode resonances appear as interference minima $(1,2)$ in backscattering amplitude vs. $ka$ (Fig. 2, for an aluminum cylinder in water). The bistatic cross section at the $(n,2) = (2,1)$ resonance (quadrupole mode excited by Rayleigh wave), e.g., exhibits constructive interference at 0° and an interference null at 180° (Fig. 3). The dashed curve represents the quadrupole scattering pattern of the resonance alone. "Total transmission" in the forward direction is indicated (3). [Supported in part by Office of Naval Research]

When an acoustic plane-wave strikes an elastic cylinder in a fluid, the cylinder's response and subsequent re-radiation generates a scattered field which may be treated mathematically by the Sommerfeld-Watson Transformation. In this treatment, the incident energy can be traced to the scattered field via several distinct processes. An interesting aspect of the technique is that at higher frequencies there is a correspondence between the modes separated mathematically and found experimentally. This parallels the situation at low frequencies where the normal modes of the cylinder are experimentally realized (individually excited) and the conventional normal mode analysis gives these modes.

The modes in this analysis are grouped into three categories. The "Rayleigh" and the infinity of "Whispering Gallery" surface wave (or circumferential) modes which are excited as the incident-wave goes through a critical angle for each mode, the "Stoneley" and the infinity of "Franz" modes (circumferential) which are excited when the incident wave grazes the cylinder, and the infinity of through waves which refract into the cylinder and re-radiate as they impinge the surface.

The contribution of each of these modes to the scattered field has been evaluated for all angles and for values of $ka$ (wave number times cylinder radius) from 5 to 200. The radiation pattern for each mode turns out to be complicated but easily interpreted physically both with regard to its shape and the variation of its intensity with $ka$.

Figures 1 and 2 are examples of the data. They show (Figure 1) the scattered field pressure amplitude as a function of angle for a $ka$ of 25 for the "Rayleigh" mode and (Figure 2) how the amplitude of this pattern changes with $ka$. The resonances at low values of $ka$ occur at the normal mode resonance frequencies thereby showing the correspondence between the theories.
ESTIMATION OF THE OCEAN BOTTOM REFLECTION COEFFICIENT

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INTRODUCTION

For a horizontally stratified ocean bottom, the complete specification of the acoustic plane wave reflection coefficient contains all the information concerning the bottom necessary for the solution of acoustic problems in the overlying ocean. An inverse method is presented, based on the well known Backus-Gilbert inversion procedure (1), to approximate the acoustic reflection coefficient of the bottom. No assumptions about the bottom are made. It is treated as a black box reflecting plane waves with the proper reflection coefficient. Numerical calculations have been made.

INVERSE METHOD

The bottom reflected field of a harmonic point source consists of the superposition of the reflections from all the downward propagating plane waves which make up the field of the point source (2). At a single frequency, the complex potential of the reflected field may be represented schematically as follows:

\[ \psi(r, z) = \int V(u) G(u, r, z) du \]  \hspace{1cm} (1)

where \( V \) is the complex reflection coefficient, \( G \) is a known complex kernel function, \( r \) is the range of the receiver from the point source, \( z \) is the sum of the distances of the receiver and source from the bottom, and \( u \) is the integration variable, which may be taken as the angle of incidence as in Ref. (2). The problem is to determine how 'best' to approximate \( V(u_0) \) (where \( u_0 \) is a specific value of \( u \)) from values of \( \psi \) found for various \( r \) and \( z \). From eq. (1), a linear combination of values of \( \psi \) at \( n \) different locations is given by

\[ \sum_{i=1}^{n} c_i \psi_i = \int V(u) \sum_{i=1}^{n} [c_i G_i(u)] du \equiv \int V(u) \delta(u - u_0) du \]  \hspace{1cm} (2)

where the subscript \( i \) on \( \psi \) and \( G \) denotes evaluation at \( r_i, z_i \). The problem is then reduced to finding the set of complex constants, \( c_i(u_0) \), that minimizes the integral of the square of the difference between \( \delta(u - u_0) \) and the delta function, \( \delta(u - u_0) \), subject of the constraint that the integral of \( \delta \) be unity. The method of Lagrange multipliers is used to satisfy the equation of constraint. A set of \( n+1 \) linear equations on the \( n \) complex constants \( c_i(u_0) \) and the one complex Lagrange multiplier results. The matrix which must be inverted is an \((n+1)\times(n+1)\) Hermitian matrix that depends only on the receiver locations and not on the particular value of \( u_0 \). The resolution of the inversion is determined by examining the height and width of the peak at \( u = u_0 \) of the approximate delta function \( \delta(u - u_0) \).

RESULTS AND CONCLUSIONS

This method has been used to construct approximate delta functions which are quite sharp with only 20 receiver locations. Great improvement should be forthcoming as a greater number of receivers are used. The method is time consuming but considering that only one \((n+1)\times(n+1)\) matrix need be calculated and inverted for all values of \( u_0 \) (at a single fixed frequency) the method seems quite economical. Error reduction techniques can be incorporated in the method in a straightforward manner.

REFERENCES

(2) L.M. Brekhovskikh, Waves in Layered Media, Academic Press (1960), 243
INTRODUCTION

Starting from the Helmholtz integral expressions have been derived for the first and second moments of the rate of change of the amplitude and phase of a signal reflected from an one-dimensional irregularly moving surface with a relatively narrow power spectrum. The active scattering area of the surface is considered to be narrow as compared to the correlation distance of the surface waves.

ANALYSIS

We consider the elevations $\zeta$ of the surface in the active scattering area as a stochastic process represented by

\[ \zeta(x, t) = R(x, t) \cos \phi(x, t) \]  

in which $t$ represents the time and $x$ the spatial coordinate which is positive in the direction of propagation of the surface waves. The samples of this process have a length of the order of the characteristic width of the active scattering area. The amplitude $R$ and the frequency $\omega = 2\pi f$ are assumed to vary linearly in these samples:

\[ R(x, t) = R(0, t) + \frac{\partial R(0, t)}{\partial x} x \]
\[ \phi(x, t) = \phi(0, t) + \frac{\partial \phi(0, t)}{\partial x} x \]  

in which $R(0, t), \phi(0, t), R' = \frac{\partial R(0, t)}{\partial x}$ and $\phi' = \frac{\partial \phi(0, t)}{\partial x}$ are stochastic variables which all refer to the center $x = 0$ of the active scattering area. By using the theory of progressive surface waves we may express the stochastic variables $R'$ and $\phi'$ in the partial time derivatives $\dot{R}, \dot{\phi}$ and $\ddot{\phi}$ which reduces Eq. 2 to

\[ R(x, t) = R(0, t) - \frac{\dot{R}(0, t)}{g} x \]
\[ \phi(x, t) = \phi(0, t) - \frac{\dot{\phi}(0, t)}{g} x \]  

Since the probability density $p(R, \dot{R}, \dot{\phi}, \ddot{\phi})$ is available, the probabilistic behaviour of the process in Eq. 1 is known which enables us by substituting Eq. 1 into the Helmholtz integral to describe the time dependent behaviour of the signal in a statistical way.
THE DOPPLER SHIFTED COMPONENT OF ACOUSTIC WAVES SCATTERED FROM A MOVING ROUGH SURFACE

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INTRODUCTION

When a plane wave is scattered from a moving randomly rough surface, outgoing doppler shifted waves are generated. The characteristics of these waves are related to the spatial and temporal characteristics of the rough surface as well as the wave vector of the incident acoustic wave. In this paper we not only derive the spectral distribution of these outgoing waves but we also show that in a certain frequency wave vector regime, doppler shifted surface waves can also be generated.

THEORY

The scattering from a time dependent surface is treated by an extension of the time independent perturbation theory of Bass (1) where the acoustic field is decomposed into mean and stochastic components; the perturbation parameter is the product of the height of the roughness, \( \alpha \), and the vertical component of the wave vector \( \mathbf{k} \). The mean field, \( v(\mathbf{r},z,t) \) is the incident and specularly reflected wave which does not undergo a doppler shift. The boundary conditions for the stochastic field, \( w(\mathbf{r},z,t) \), in the presence of a stationary two-fluid boundary have also been derived (2, 3) and the generalization to a moving randomly rough pressure release surface is presented in this paper. A general expression for \( w \) is obtained by noting that the stochastic field must originate from the boundary and hence we can use a Fourier decomposition for \( w \) of outgoing waves,

\[
\tilde{w}(\xi,n) \exp\left[i(\xi \cdot \mathbf{r} + n z) - i\omega t\right],
\]

where \( \xi = \sqrt{(\alpha^2/c^2-\xi^2)} \). The energy spectrum is then given by \( \langle \tilde{w}(\xi,n;z) \tilde{w}^*(\xi,n;z) \rangle \); the brackets denote an average with respect to the stochastic variable \( \alpha \) and \( \tilde{w}(\xi,n;z) \) includes the \( z \) exponential factor.

RESULTS

We consider a sea surface composed of gravity waves which obey the frequency wave number dispersion relation \( \omega_s = \sqrt{gk_s} \). A scattered field with outgoing waves will be generated with doppler shifted frequencies of \( \tilde{\omega} = \omega_s + (g|\mathbf{k} - \xi|) \) where \( \mathbf{k} - \xi \) is the difference between the transverse components of the wave vector of the incident and scattered waves. The relative amplitudes of the scattered waves are weighted by the sea surface spectrum evaluated at these values of wave vectors. This is a well known far field Kirchhoff result (4); however in this treatment we also find the near surface result that there is a low frequency regime where doppler shifted surface waves can be generated which decay exponentially as the observation point moves away from the surface. This is the doppler shifted acoustic analog to the Wood Phenomenon (2, 5) which was first observed using an optical diffraction grating.

REFERENCES

(5) J. R. Wood, Phil. Mag. (1902) 4, 396
OBSERVED SCATTERING FROM SALT FINGERS PRODUCED BY THE TWO-SOLUTE METHOD

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INTRODUCTION

Salt fingers are a mechanism of vertical transport in fluids. If a layer of warm, salty water overlies cooler, fresher water, salt fingers may form at the interface. Initiated by the difference in molecular diffusivities, a column or finger of downward flowing warm, salty water has fingers of upward flowing cooler, fresher water as nearest neighbors. Salt fingers have been studied extensively in the laboratory by optical means and have been demonstrated to exist in certain areas of the ocean.

More controlled experiments are accomplished in the laboratory using two solutes of different molecular diffusivities such as sugar and salt. The first study of the affect of these fingers on horizontally propagating 1-40 MHz sound implied attenuation due to scattering by the salt fingers. In the present paper, we experimentally correlate magnitude and angular behavior of the scattered signal to the sound frequency, and the characteristic horizontal dimension of the salt fingers.

EXPERIMENTAL

Both the transmitting transducer and the separate receiver were constrained to move about at fixed radii from the center of the salt finger region. A cylindrical, thin-walled plastic container held the salt fingers, while the transducers were situated in distilled water outside the finger region. The angular behavior of the scattered sound was investigated by fixing the receiver and varying the angle between it and the incident sound beam. The pulsed sound generator was operated at a few frequencies between 0.7 MHz and 13 MHz. The typical finger width naturally grew within the range of 0.1 mm to 1.0 mm. Scattering utilizing the combination of high frequency-small finger width appeared at different angles and magnitude than scattering resulting from a lower frequency-larger finger width combination.

REFERENCES

UN MODELE STATISTIQUE POUR LES ECHOS REVERBERES PAR UNE SURFACE FLUCTUANTE ALEATOIRE

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INTRODUCTION

La réverbération ou rétrodiffusion est un phénomène parasite lié aux sonars de détection active. Elle est constituée par une partie de l'énergie acoustique renvoyée vers l'émetteur par les hétérogénéités du milieu de propagation.

MODELE STATISTIQUE DE LA REVERBERATION DE SURFACE

D'après la figure 1, il est évident qu'un signal sera reçu par le récepteur R quand la normale au plan tangent de la surface coïncidera avec l'incidence d'angle à. S'il n'en est pas ainsi, le signal ne sera pas reçu. La suite des signaux reçus en R est obtenue par effacement aléatoire de la suite des signaux émis. Le temps séparant deux signaux consécutifs est aléatoire. La loi de distribution de ces intervalles dépend de la densité de probabilité des pentes de la surface. Dans le cas général, on montre qu'elle est de la forme :

\[ E \{ K \} = \frac{e^{-\bar{\beta}T}}{T} \]

où \( \bar{\beta} = \frac{\log(1-p)}{T} \)

est la densité équivalente des processus, \( p \) est la dp des pentes de la surface ; \( T \) la période de récurrence des signaux émis. On peut donc déterminer \( p \) si l'on sait mesurer \( \bar{\beta} \).

DETERMINATION DES DENSITES EQUIVALENTES

On a simulé les vagues de la mer sur un modèle réduit en cuve acoustique. On a analysé la répartition des intervalles de temps séparant les signaux reçus. La Figure 2 montre sur papier semi-log les lois \( E \{ K \} \) pour plusieurs incidences. La Figure 3 donne les variations de \( \bar{\beta} \) en fonction de \( p \).

C. GAZANHES A. Telec (1975), 30, n° 7-8, pp 291-297
G. BONNET Rapport CEPHAG N° 21/65, Grenoble, 92 pages.
ACOUSTIC CHARACTERISTICS OF PELAGIC FISH SCHOOLS OFF SOUTHERN CALIFORNIA

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INTRODUCTION

Acoustic characteristics of pelagic fish schools were measured in the California Current system during numerous cruises from November 1974 to October 1975. Among the parameters studied were peak target strength (TS), integrated target strength (INT), and target extent (TE) which is the length of the target along the acoustic axis. The dominant pelagic schooling species in this area is the northern anchovy, Engraulis mordax, and an estimated minimum of 75 percent of the schools sampled were of this species (1).

DATA COLLECTION AND REDUCTION

Data were collected using a hull-mounted, side-looking 30 kHz sonar system for both survey (2) and stationary operations. During survey operations conducted at full speed (10 knots), the sonar transmitted 10 msec pulses once each second. The received narrow-band process was filtered, amplified, envelope-detected, and interrogated for the presence of significant echo-level by a minicomputer. The fish-school echo must be present on three consecutive pulses for minimum detection. The peak target strength was computed from

\[
TS = 40 \log(p/p_o)^2 + 40 \log R + 2aR
\]

where \(40 \log R + 2aR\) comprises spreading and attenuation losses, and the ratio of \(p/p_o\) is that of received to transmitted peak pressure. Measures of integrated target strength were made in only a few isolated cases, with the ship hove-to. Two different schools were sampled at four different pulse durations. INT was computed from

\[
INT = 10 \log \int_0^T \frac{1}{T} 10^{TS*/10} \, dt
\]

where \(T\) is the pulse duration and \(TS^*\) represents a continuous target strength function over echo duration \(T\).

RESULTS AND CONCLUSIONS

a. A summary of TS values for all survey cruises is presented in Fig.1.

b. For the same schools, average TE was equal to 64 m with a standard deviation of 42 m; a strong mode existed at 35 m.

c. Values of INT, for this limited data set, appear to be independent of pulse duration suggesting its use as a fish stock assessment parameter.

REFERENCES

(1) K. Maia, California Department of Fish and Game, 1974, Fish Bulletin 162.
INTRODUCTION

Propagation experiments in coastal waters are normally performed with receivers located at mid-depth. However, in validating acoustic propagation models, the field intensity close to surface or bottom of the water column is generally more sensitive to changes in environmental parameters than the field at mid-water depth (1).

In this paper, the calculated field distribution close to both boundaries of a shallow water propagation channel is compared with experimental data for the purpose of validating a particular propagation model.

THEORETICAL BASIS

The model used for computing the field intensity versus depth and range for various source frequencies is based on normal-model theory. It is a model that allows for an arbitrary variation of sound-speed profile with depth in both the water column and in the bottom. The model computes transmission losses as arising from compressional-wave attenuation in the bottom, volume absorption in the water, and scattering loss from a rough sea surface.

EXPERIMENTAL TECHNIQUE

The experiments were performed in almost isothermal water in March 1975. Measurements were made over a 30 km long propagation path in water approximately 110 m deep. Two runs were made. One with 9 hydrophones sampling the sound field at 1 m intervals close to the sea surface, and one with the hydrophone array placed close to the bottom. In both runs an extra hydrophone was suspended at mid-depth for checking reproducibility of the data. Explosive charges fired at mid-depth were used as acoustic sources, and the received broadband signals were analysed in one-third octave bands with centre frequencies ranging from 50 to 6400 Hz.

RESULTS AND CONCLUSIONS

A remarkably good agreement is found between theory and experiment for the entire frequency range. Thus, the main features exhibited by the experimental data concerning the optimal propagation frequency at various ranges and depths are accurately predicted by the acoustic model. This in turn means that the confidence gained earlier in this particular propagation model (see Ref. 1) has been further strengthened.

REFERENCES

INTRODUCTION

Sources of high-intensity sound, for instance underwater explosions, are frequently used in oil prospecting and in geophysical and geological studies of bottom and subbottom materials and profiles. The necessity of using high-intensity sound sources is among other things caused by the high absorption, increasing about linearly with frequency, found in most water-saturated marine sediments, and the description of the propagation of the finite-amplitude waves thus generated requires knowledge of the acoustic nonlinearities of the sediments in order to be able to explain the absorption in excess of small signal absorption, the formation of harmonics and in particular in order to develop procedures for the most profitable utilization of the acoustic power available in the signals.

THEORETICAL BASIS

Several experimental methods are available for a determination of the second order acoustic nonlinearity ratio \( \frac{B}{A} \) of fluids, but the strong absorption and the special mineral grain/pore water structure characterizing most marine sediments limit the number of potential procedures for a determination of \( \frac{B}{A} \). An approximate thermodynamical procedure has turned out to be most applicable for the \( \frac{B}{A} \) determination, and the acoustic nonlinearities thus determined for a number of water-saturated marine sediments characterizing the Danish Archipelago together with the viscosity number of these sediments have been used for a calculation of the discontinuity distance of an initial sinusoidal finite-amplitude wave, the prospective extend of a "sawtooth region", the effects of acoustic saturation etc. [1].

EXPERIMENTAL TECHNIQUES

The density, the isobaric compressibility and the specific heat at constant pressure of the sediments are measured by the use of standard thermodynamic laboratory procedures. The velocity of sound and the amplitude absorption coefficient at various frequencies are measured using propagating and standing waves of small amplitude. The velocity of sound as a function of pressure and temperature is determined through a comparison method using the pore water as a reference liquid.

RESULTS AND CONCLUSIONS

\( \frac{B}{A} \)-ratios varying between 11 and 12 are determined for the sediments considered, but only very low Gol'dberg numbers are determined due to the absorption. Increased Gol'dberg numbers have been obtained when using blasting caps - equivalent charge 0.8 g TNT - as sound sources, [2]. Experimental and theoretical studies of the decay rate with distance for the shock peak amplitude show good agreement.

REFERENCES

MATHEMATICAL PERFORMANCE TESTS OF THE COMPUTER MODEL "RAIBAC" FOR UNDERWATER SOUND PROPAGATION

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INTRODUCTION

Computer models of sound propagation are mainly used in two areas: (a) data analysis in underwater acoustics, and (b) prediction and optimisation in sonar. The ray-tracing model RAIBAC (1,2) has been designed specifically for the second purpose. The usual undersampling problem occurring in the representation of the sound field in the range/depth plane is avoided there by forming spatial averages instead of random samples.

PRACTICAL REQUIREMENTS

The main requirement for models applied in sonar is high speed of computation and reliability. A trade-off in resolution is easily accepted; a finite resolution cell in the range/depth plane of at least 200 x 10 m is even desirable.

MATHEMATICAL TESTS

A series of mathematical tests for propagation models had been developed by D.H. Wood (3,4) at SACLANTCEN. They consist of pairs of mathematical formulae for both, sound speed profile and corresponding exact solution of the wave equation. Figs. 1 and 2 show typical examples of graphical comparisons between model results and Wood's test functions. (Contour interval 2.5 dB).

CONCLUSION

Several of the available scientific type ray-tracing models have been tested together with RAIBAC. So far, it appears that the simple and fast averaging procedure applied in RAIBAC yields a degree of reliability that can only be achieved otherwise by normal mode methods.

REFERENCES

(1) W.Bachmann, B.de Raigniac, ICA8, 1974, I, 441
(4) D.H.Wood, SACLANTCEN Conf.Proc.No.17, 1975, 8, 45
The problem treated here is the calculation of the correct limiting form for the intensity level and depth dependence of sound propagating in a duct; taking high frequencies where one can in effect average over the modes, and averaging also in range. The starting point is the concise energy flux formulation in Weston (1) for the relative intensity $I$ at position $B$ due to a source at $A$, simplified here for the case of low angles, small changes in sound velocity and no attenuation:

$$I = \frac{2d\phi_A d\phi_B}{r c dT} \quad \text{where} \quad T = \int \frac{dh}{c}.$$  

Here $T$ is a characteristic time which is invariant to slow changes in the duct, $\phi$ is grazing angle, $d\phi_A$ and $d\phi_B$ are corresponding angular elements at $A$ and $B$, $r$ is range, $c$ is sound velocity and $h$ is measurement depth.

Let us apply this to a parabolic duct, in which the sound velocity increase over the axis value is proportional to the square of the distance from the axis. For our purposes this is a reasonable representation of any depressed sound channel in the ocean. Goodman and Duykers (2) showed that the ray paths are all approximately sine waves, with distance between axis crossings independent of amplitude, and therefore with strong focusing. Appropriate substitution into equation (1) and some manipulation gives a solution in terms of maximum axis angle $\phi$, duct thickness $2h_0$ and source and receiver depths relative to the axis $h_A$ and $h_B$:

$$T = \left[ \frac{2}{\phi} \frac{\tan \phi}{\pi} \right] \left[ \frac{1}{h_A} \ln \left( 1 + \frac{h_A^2 - h_B^2}{h_A^2 - h_0^2} \right) + \frac{h_B^2 - h_A^2}{h_A^2 - h_B^2} \right].$$

This is shown as the product of a bracketed scale factor and a bracketed depth factor, the latter defining a universal set of curves which depend symmetrically on the dimensionless depths $h_A/h_0$ and $h_B/h_0$.

Two examples are shown in figure 1, which need cover only the lower half of the duct. The most surprising feature is the occurrence of infinities on each occasion that source and receiver depth are the same, in fact equation (2) shows this is a characteristic of the whole family. Pedersen (3) has shown that ray caustics normally occur when source and receiver interfaces have the same sound velocity, and Urick (4) has illustrations of such "horizontal" caustics. We have demonstrated here that these caustics give a nominally infinite intensity even when smeared in range.

Only one application of equation (1) has been treated here but it can be used for other profiles, as well as changing profiles, also several generalisations are possible. It is desired to stress the usefulness in prediction and in interpreting experiments which arises from the hidden averaging in its form.

REFERENCES
(1) D.E. Weston, Proc Phys Soc (1959) 125, 365
THE ATTENUATION OF SOUND IN THE SEA

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INTRODUCTION

In a previous paper (1), the authors have described the four principal mechanisms which give rise to attenuation of sound in the sea. Recent experiments both in the laboratory and at sea have verified these mechanisms and allow us to further quantify the attenuation of sound for the frequency range 10-10,000 Hertz.

THEORETICAL BASES

It is well known that chemical relaxation processes are a major contributor to attenuation. For example, the dissociation of MgSO₄ causes significant absorption of sound in sea water below 100,000 Hertz. Laboratory measurements (2,3) have now lead to the postulation of a second relaxation process, a boric acid-borate reaction given by eq.(1), which is dominant below 10,000 Hertz.

\[ \text{B(OH)}_3 + \text{OH}^- \rightleftharpoons \text{B(OH)}_2 \text{OH}^- \rightleftharpoons \text{B(OH)}_4^- \quad (1) \]

At very low frequencies (100 Hertz and below) attenuation frequently can be attributed to scattering mechanisms. Various causes of this scattering have been suggested including internal waves (4).

EXPERIMENTAL TECHNIQUES

Low frequency attenuation measurements in the sea are generally conducted in the deep ocean sound channel (SOFAR duct) to insure totally refracted paths over long distances. Explosive signals are detonated at various ranges from a receiving hydrophone with both sources and receiver located at the depth of the sound channel axis.

RESULTS AND CONCLUSIONS

Measurements from deep ocean areas throughout the world show that for the frequency range below 1 kHz absorption is proportional to the frequency squared with a regional variation in magnitude. This variation is attributed to the sensitivity of the boric acid-borate reaction to local values of pH in the ocean.

In deep ocean areas the measured values of attenuation tends to be frequency independent below 100 Hertz. For some locations the attenuation is the value predicted by scattering from internal waves; however, for other locations the measured values are higher suggesting a possible contribution from other scatter mechanisms.

(Work supported by NOSC)

REFERENCES

(2)F.H.Fisher et al, IEEE OCEAN '75 (1975) 21-24
SERIES EXPANSIONS OF UNDERWATER ACOUSTIC FIELDS ABOUT CAUSTIC POINTS

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INTRODUCTION

This theoretical paper discusses a new power-series representation of the acoustic pressure in the neighborhood of a ray-theory caustic. The series are derived from U.A. (Uniform Asymptotics), as adapted for underwater acoustics (1). This U.A. formulation is expressed in terms of the travel times and pressures of geometrical acoustics. Hence it can only be implemented on the ensonified side of the caustic. The series representation was developed for application in the shadow zone as well.

APPROACH

In the U.A. formulation the pressure is expressed in terms of Airy functions, Ai, by

\[ p = \left[ F \text{Ai}(-x) + i G \text{Ai}(-x) \right] \exp(i \phi) \]  

(1)

where \( F, G, x, \) and \( \phi \) are functions of the sums and differences of the travel times and pressures of the two rays associated with the caustic. Each of the constituents of eq. (1) is expanded about the caustic point as a Taylor series in the ray parameter, for fixed receiver depth. The expansion for the range is inverted yielding an expression for the ray parameter in terms of a series in \( (4R)^k \), where \( \Delta R \) is the range displacement from the caustic point. This inversion is then used to eliminate the ray parameter in order to express each ray theory item as a series in \( (4R)^k \). In the inversion there are two ray parameters associated with \( \Delta R \) — one corresponding to each branch of the range curve. These two parameters are real in the ensonified region and complex conjugates in the shadow zone.

RESULTS

The terms \( F, G, x, \) and \( \phi \) are demonstrated to be real-valued Taylor series in \( \Delta R \). The disappearance of odd powers of \( (4R)^k \) can be associated with the complex conjugate nature of the ray parameters in the shadow zone and the fact that only the sums and differences of travel times and pressures are required in \( F, G, x, \) and \( \phi \). The first two non-zero coefficients have been determined for \( F, G, \) and \( x \) and three for \( \phi \). These coefficients are expressed in terms of various order derivatives of range with respect to ray parameter, evaluated at the caustic point, and trigonometric functions of the caustic-ray angle.

Although U.A. is not formulated as a caustic-point expansion, this new series representation can be compared with the N.U.A. (Non-Uniform Asymptotic) result of ref. (2), which is a caustic-point expansion using higher-order corrections to the WKB approach. The first three terms of eq. (20) of ref. (2) reduce to a result identical to that obtained from the first-order terms of \( F, G, \) and \( x \). The fourth term of eq. (20) of ref. (2) (involving \( c \text{Ai}(4\phi) \)) has no counterpart in the series expansion of eq. (1). The U.A. approach of eq. (1) is superior to N.U.A. in that the region of validity of N.U.A. is unknown a priori; whereas U.A. is guaranteed to fair in with geometrical ray theory (3). Comparison of N.U.A. and U.A. low-frequency propagation losses for deep-ocean profiles show that differences of several dB are typical at distances well-removed from the caustic.

REFERENCES

INTRODUCTION

Amplitude and phase fluctuations of low frequency, continuous wave, underwater acoustic transmissions are presented and interpreted in terms of the interaction of sound with ocean internal gravity waves. Oceanographic data is presented and compared with the acoustic data to show that internal waves are responsible for much of the acoustic fluctuation.

THEORETICAL BASIS

Internal waves displace isotherms and consequently cause fluctuations in refractive index and sound velocity. Recent theoretical developments have led to the prediction of a universal power spectrum for internal waves (1). Energy is distributed between the local inertial frequency and the local buoyancy, or Brunt-Vaisala, frequency. It has been predicted that spectra of acoustic fluctuations in this frequency band will be proportional to the power spectrum of internal waves (2).

EXPERIMENTAL TECHNIQUES

A portable acoustic range has been developed to enable the quantitative measurement of acoustic fluctuations. The range consists of moored low-frequency, continuous tone acoustic sources, moored narrowband acoustic receivers, and free-drifting hydrophones. A unique acoustic navigation system allows source and receiver tracks to be determined with an accuracy that is a small fraction of the low frequency acoustic wavelength (3). The range has been deployed in several locations to measure the magnitude and spectral characteristics of fluctuations of the amplitude and phase of the low frequency transmissions. Simultaneous oceanographic measurements have been made.

RESULTS AND CONCLUSIONS

In this paper we present the results of three separate experiments at acoustic frequencies of 100 and 220 Hz. Measurements were made at ranges of 10 to 600 km in the Sargasso Sea area of the Atlantic Ocean. The data reveal that acoustic fluctuations in the internal wave frequency band are strongly influenced by internal waves, and that the effects of internal waves on the acoustic signal are range dependent. At long ranges multipath interference dominates the fluctuation spectra. Comparison of this data with recent theoretical developments involving gravity wave vs multipath induced acoustic fluctuations are favorable.

REFERENCES

THE ADDITION FORMULA FOR A SEPARABLE EQUATION

In its simplest form, the addition formula exactly expresses the Green's function of

\[ u_{xx} + u_{yy} + (a_1(x) + b_2(y)) u = 0 \]  \tag{1}

as

\[ G(x,y,x_0,y_0) = \int G_1(x,x_0,t) G_2(y,y_0,-t) \, dt, \]  \tag{2}

where \( G_1(x,x_0,t) \) is the Green's function of the parabolic equation

\[ i u_t + u_{xx} + a_1(x) u = 0, \]  \tag{3}

and \( G_2(y,y_0,t) \) is the backwards Green's function of a second parabolic equation

\[-i u_t + u_{yy} + a_2(y) u = 0. \]  \tag{4}

Equations 3 and 4 inherit their \( x \) or \( y \) boundary conditions from equation 1.

RESULTS AND CONCLUSIONS

It is convenient to use the Green's functions of equations 3 and 4 because their data on the line \( t = 0 \) do not depend on the coefficients. The representation of Polyanskii, which does not have this advantage, can be derived by setting \( a_2(y) \) to zero and performing a fractional integration by parts of equation 2.

When the variation of \( a_2(y) \) is weak, the Green's function of equation 4 can be estimated. In this case, the method of stationary phase applied to equation 2 yields a Fock-Tappert type approximation and demonstrates the nature of the error committed.

REFERENCES


ERASING METHOD OF THE SECOND IMPULSE ON UNDERWATER WIRE EXPLOSION

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INTRODUCTION
The exploding wire source, investigated by the authors, is useful and effective in certain fields, as compared with the other electrical, mechanical and chemical means. However, these explosive type sources have a problem of the second impulse (bubble impulse) generation, arising by the cavity motion. This bubble impulse generation may considerably influences to the pulse measurement techniques. In this paper, the erasing method of this bubble impulse is treated by the laboratory works.

ERASING THE SECOND (BUBBLE) IMPULSE
The wire explosion source, charged 800J, wired Cu, length 8cm, diameter 0.26mm, produces 150dB peak pressure of the first impulse and 143dB that of the bubble impulse 15msec. after the first impulse at 50cm. This bubble impulse can not be neglected as comparing the first impulse peak. Then the effective erasing method of the bubble impulse without influences to the first impulse becomes one of the practical problems for its use. The authors have been carried out the simple erasing method, divided into three main forms; 1. wire explosion close to the surface of the water, 2. wire explosion with the blowing air pipe, 3. wire explosion with the baffle board. Considerably effective result can be obtained by the second form. The experimental result and theoretical result are shown in Fig. 1. The experimental result well agrees with the calculated value. In this result, increasing the length of the pipe diameter, the similar result of the first form must be obtained without air blowing. In this method, the reflected impulse of the first shock wave from the surface of water is considerably controlled by the non-linear motion of the water surface. Other words, high power impulse reflection may be deformed by the tensile strength of the water. Then the influences of the water surface to the first impulse reflection can be neglected, and the bubble impulse is effectively erased by the unsymmetrical motion of the cavity. The third form gives the complex selection of the baffle board.

CONCLUSION
Unexpected bubble impulse in the explosive source, such as exploding wire source, for the pulse measurement can be effectively erased by the simple air blowing to the generated cavity. Using this method the high power mono-impulse source can be realized, then it is noted that the useful results would be obtained to the physical simulation or impulse measurement on the pulse techniques.

REFERENCES
(1) MATSUDA, M et al.; 7th I.C.A. 4-6
RESULTS OF A HIGH DATA RATE UNDERWATER ACOUSTIC COMMUNICATION EXPERIMENT

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INTRODUCTION

The Naval Underwater Systems Center, jointly with Sperry Research Corporation, conducted an experiment at lake Seneca, New York to investigate the feasibility of high rate underwater acoustic communication using parametric sonar. A number of phase-shift-keyed, frequency-hopped signal formats were transmitted over a range of 4 Km.

The transmitted data rate varied from 273 to 6600 bits/second depending upon the signal format that was used. A band of 8800 Hz was used for all formats.

The receptions were recorded on a tape and later processed in the laboratory. All 2-phase and 4-phase encoded signals were received without error, whereas the 8-phase encoding was subject to a small probability of error of about 10^-3. The results of the data analysis are displayed on propeller diagrams. The performance is presented in terms of signal-to-noise ratio and probability of error.

RESULTS AND CONCLUSIONS

These results from the limited data analysis indicate that high data rate underwater acoustic communication is indeed feasible using wide bandwidth parametric sonar. (Work supported by NAVSEA)
TURBULENT BOUNDARY LAYER WALL PRESSURE FLUCTUATIONS ON A LONG FLEXIBLE WALL CYLINDER

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INTRODUCTION

Some limited experimental and theoretical work has been done by Willmarth and Yang (1) and White (2) for turbulent boundary layer flow in the axial direction over a long rigid cylinder. An experimental investigation which measured the characteristics of turbulent boundary layer wall pressure fluctuations (TBLWPF) over a long flexible cylinder was conducted. The purpose of this investigation was to determine if any differences existed between the TBLWPF on rigid (1) and flexible wall cylinders and flat plates. Knowledge of the characteristics of the TBLWPF is important so that the proper wall excitation model can be used.

EXPERIMENTAL TECHNIQUES

A long, fluid filled, neutrally buoyant flexible wall cylinder suitable for at sea operations was constructed having a diameter of 7.6 cm. This cylinder was made up into short lengths varying from 2.3 m to 4.6 m so that the distance from the nose to the TBLWPF sensors could be varied. The TBLWPF sensors were four small flush mounted hydrophones aligned so that both lateral (circumferentially) and longitudinal (axial) correlation measurements could be accomplished. The long cylinder was towed at sea from a research vessel with a heavy cable so that the cylinder was completely submerged in a wake free area astern of the vessel. The tow cable was torque balanced and since the cylinder was neutrally buoyant the turbulent boundary layer developed on it was essentially axi-symmetric. Tests were conducted at speeds of 3.1 to 9.3 m/sec and at four axial locations of the TBLWPF sensors. The length to diameter ratio at these four axial locations varied from 146 to 512.

RESULTS AND CONCLUSIONS

Autospectra pressure data were obtained at the four axial locations and are presented in non-dimensional form and compared with (1) and typical smooth-wall flat plate spectra. Cross-spectra pressure data were obtained at two locations along the cylinder and are presented in the form of phase and coherence plots for both the lateral and longitudinal directions. These data are compared with (1).

A single representation of the non-dimensional TBLWPF autospectrum is obtained and it is shown that $P \propto f^{\alpha}$ is an appropriate model within the frequency range investigated. This differs from the $P \propto f^{-1}$ relationship characteristic of rigid-wall cylinders. There was little difference in the cross-spectral representations between rigid and flexible wall cylinders. Data from the aft end of the cylinder (highest L/D ratio) indicate a sweeping-off of the turbulent boundary layer and higher pressure fluctuation levels. (Work supported by NAVSEA)

REFERENCES

(1) W. W. Willmarth and C. Yang, "Wall Pressure Fluctuations Beneath an Axially Symmetric Turbulent Boundary Layer on a Cylinder," ORA Project 02149, Univ. of Michigan, Aug 1969
INFLUENCE OF OCEAN BOTTOM STRUCTURE ON SPECTRAL AND CORRELATION CHARACTERISTICS OF REFLECTED FM SIGNALS

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INTRODUCTION
Some spectral and correlation characteristics of frequency modulated (FM) sound signals reflected from the ocean bottom have been investigated provided that transmitter and receiver are in motion.

EXPERIMENTAL TECHNIQUES
Measurements of statistical characteristics of the reflected signals were carried out at normal incidence of sound in deep ocean with plain bottom. The transmitter and receiver were located at the same point. Sound signals with the linear frequency modulation in the band of 250-300 Hz and central frequency of 3.15 kHz were used. The impulse duration and the time interval between impulses were equal to 5-7 s and 0.3-0.5 min, correspondingly.

RESULTS AND CONCLUSIONS
The steady cross correlation of the amplitude fluctuations of the reflected FM signals was found for a large horizontal spacing between transmitting-receiving points. Sometimes maximum of the cross correlation function \( N(\tau) \) was found at some time delay between signals received at different points. The time delay is increased and the magnitude of the correlation maximum is decreased as a distance between receiving points is increased (Fig. 1). Curves 1-5 show the dependence of the cross correlation function \( N(\tau) \) on the time delay \( \tau \) for five different spacings 10, 20, 30, 40 and 50 meters correspondingly. The power spectrum of the amplitude fluctuations of reflected signals had a row of sharp peaks.

For the explanation facts obtained a geoaoustic model of the ocean bottom is proposed. According to this model the ocean bottom consists of several layers with rough boundaries. The following assumptions are made: random roughnesses are small as compared with the wavelength of sound and boundaries of these layers are parallel in average. It is shown that the steady correlation and the presence of the sharp peaks are due to the layered structure of the bottom and the time delay of the correlation maximum is due to a small changing of a layer. A method of determination of the layer thickness according to power spectrum of the reflected signals is proposed. It is shown that the unambiguous determination of the bottom structure is possible provided that one of boundaries reflects sound stronger than others.
Application of computation methods to the solution of the boundary value problem for Helmholtz equation is shown in the article
\[(\alpha + k^2)\varphi(x, y) = 0 \quad \Delta = \frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2}\]
The limit \(\lim_{W(x, y, t)} W(x, y, t) = 0\) is considered, where function \(W(x, y, t)\):

1) meets parabolic equation with Helmholtz operator in the right part
\[\frac{\partial W}{\partial t} = p(\alpha + k^2)W\]
2) satisfies the same boundary value problem, as function \(U(x, y)\) satisfies;
3) satisfies random initial condition, when \(t = 0\), \(p\) - is complex coefficient \(\{1\}, \{2\}\).

For the solution of parabolic equation for \(W\) and Helmholtz equation for \(U\) "hopscotch" circuit, consisting in operator replacing by proper difference operator \(h\),

\(h\) - the step of the difference approximation, and grid knots are divided in two subsets of even and odd knots \(\{1\}, \{4\}\).

Application of such circuit to the following tasks is considered:
field computations in the regions of complex shape, filled by absorbing medium; field computations near the shielded radiator; field computations in the regions filled by nonuniform medium and other tasks, three-dimensional ones among them. Application of numerical methods to determine natural frequencies of mechanical vibrations of two- and three-dimensional bodies of different forms, and also regions, filled by nonuniform liquid and solid mediums is considered. Computation were performed for the case of medium with small losses. For such mediums near inherent values of wave number the norm of numerical solution is equaled to the sum of field amplitude modules, computed in all grid knots, increased sharply. The results of the computations are presented in the form of the figures.

The effectiveness of applying of numerical methods for computation of sound field in different regions and inherent vibrations of the bodies of complex form is shown.

REFERENCES:
A CORRELATION METHOD FOR THE MEASUREMENT OF THE ROTATING NOISE SOURCES

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PRINCIPLE OF THE METHOD

By means of two sensors (Fig.1) acoustical waves from the rotating source of useful signal USS are received. Received as well are the unwanted waves from static (SSD) or rotating (RSD) sources of disturbances and the waves coming from USS, but after the reflection (RED). The received signals are fed through time-dependent delay lines having such delays $\tau(t)$ which allow for the time differences of the signals coming directly from USS. The crosscorrelation function of the two signals obtained by the averaging over at least one period of rotation of the source, thus estimates the average autocorrelation function of the useful signal. Since the system is focused onto USS the contributions from SSD, RSD and RED are suppressed.

ON THE POSSIBILITIES OF THE METHOD

The attenuation of some out-of-focus sources was calculated (Fig.2) on the basis of the mathematical model of the method which was confirmed experimentally [1]. White noise hydroacoustical signals with the bandwidth $B$ were assumed. USS was taken to rotate at the radius of 10 cm, while sensors were positioned symmetrically in the rotation plane, 25 cm sideways in level with the rotating axis. RSD was in the rotation plane, shifted from USS either anglewise only, by an angle $\phi$ (curve 1), or radially only, lying at the radius $p$ (curve 2). SSD was assumed to lie on the rotation axis at a distance $z$ from the rotation plane (curve 3). The attenuations obtained for RED at the near-by plane boundaries and for other positions of SSD or RSD were high, similarly to the situation shown in Fig.2. The only exception was RSD being shifted from USS in parallel to the axis only. In this case the resolution was poor.

REFERENCE

1. B. Bajić, Elektrotehnika (1976) 19 (in Serbo-Croatian)
STUDY ON CORRELATION METHOD FOR ULTRASONIC CURRENT METER

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INTRODUCTION

Measurement of current velocity through the straight by using an ultrasonic flow meter based on the measurement of the pulse propagation time is impossible due to disturbance of circumstance noise in sea. Then we considered use of M-sequence signal instead of pulse sound. When using M-sequence signal the measurement of the cross-correlation of the transmitted signal and the received one after propagation or that of two received signals after propagation towards the opposite directions may derive the propagation time or its difference even under the conditions of high circumstance noise and multi-path propagation due to the surface-bottom reflection. Then a fundamental experiment was carried out.

EXPERIMENTS

Fig.1 shows a frequency response of output-to-input voltage ratio of the projector and the hydrophone used in the experiment.

Fig.2 shows a block diagram of the measuring system. M-sequence signal of the 8th order was used and when it was generated by clock signal of 100 kHz the best correlation waveform was obtained and the sound propagation time was measured with accuracy of about 2usec. Experiments were performed in sea of bottom depth of about 20m. Fig.3 shows an example of a measured correlation waveform when the transducers interval was about 70m. Nevertheless SN ratio observed at the hydrophone terminal was about 0dB, propagated signals were successfully detected after correlation.
INTRODUCTION

The nature of the FM signal modulated by "random telegraph signal (RTS)" — binary Poisson process — and its application to underwater acoustic measurements have been reported previously (1, 2). The present paper is concerned with the characteristics and applications of the FM signal modulated by "smoothed random telegraph signal (SRTS)."

THEORETICAL BACKGROUND AND EXPERIMENTAL TECHNIQUES

While the FM signal modulated by RTS is determined by three parameters: \( \lambda = \text{threshold level}/\text{r.m.s. of the Gaussian noise}, k = \text{modulation index} = (2\pi \Delta f/\lambda), \) where \( \Delta f \) is the frequency deviation and \( \lambda \) the mean number of zero-crossing per second) and the center frequency \( f_0 \), a fourth parameter \( T = \tau R \) (time constant of the low-pass RC-filter of Fig. 1) becomes available in case of the FM signal modulated by SRTS. This makes possible the formation of various types of power spectrum by appropriate choice of the parameters \( \lambda T \) and \( \lambda \) as indicated in Fig. 2. The power spectrum is obtained as electronic Fourier-transform of the auto-correlation function obtained by the passage of the FM signal through an electronic digital correlator.

Modulation by SRTS has the advantage that the flat power spectrum is obtainable by choosing \( \lambda T = 1 \), \( k \gg 1 \) and \( \lambda = 0 \), making unnecessary the use of a wide-band white noise required in the case of modulation by RTS because of the relation \( k = 2\pi \Delta f/\lambda \sqrt{2} \).

As an example of this method to underwater acoustic measurements, Fig. 3 shows (a) the power spectrum \( \Phi_x(\omega) \) of the transmitter input \( x(t) \), and (b) the cross-spectrum \( |\Phi_{xy}(\omega)| \) between \( x(t) \) and the microphone output \( y(t) \). Here the sensitivity product \( |G(\omega)| = |\Phi_{xy}(\omega)|/\Phi_x(\omega) \) is directly proportional to \( |\Phi_{xy}(\omega)| \) because \( \Phi_x(\omega) \) is flat. Reflection from a square plate showing resonance minimum and diffraction effects are also obtained.

REFERENCES

(1, 2)
E. Kashiwagi, Acustica (1977) 37, No. 2, No. 3
(to be published).

Fig. 1 Electrical circuit for generating RTS and SRTS.

Fig. 2
The power spectra of the FM signal; \( \lambda = 0 \) in the left column and \( \lambda = 0.3 \) in the right one.

Fig. 3 (a) Power spectrum \( \Phi_x(\omega); k=32, \lambda T = 1, \lambda = 0 \), (b) cross-spectrum \( |\Phi_{xy}(\omega)| \).
INTRODUCTION

Collapsing cavitation bubbles are a major source of sound radiated by marine vehicles, yet only a few researchers have attempted to calculate this sound. Many of the difficulties encountered in exact analyses can be overcome by recognizing two bubble collapse regimes and by making simplified first-order calculations for each regime.

THEORETICAL ANALYSIS

Complete analyses of bubble dynamics include effects of surface tension, heat conduction, viscosity and compressibility, as well as that of residual gases. An analysis by Hickling and Plesset (1) has shown that the residual gas effect is dominant if the partial pressure of the permanent gas, Q, is more than about 1% of the collapse pressure, P. For this condition the Noltingk-Neppiras (2) equation for bubble wall motion is applicable. Using this equation to determine instantaneous velocities and accelerations, the volume acceleration, \( \dot{V} \), can be calculated and the instantaneous sound pressure obtained from:

\[
p = \rho \frac{\dot{V}}{4\pi r}
\]

Another approach which gives similar results and is less mathematically complex uses Rayleigh's (3) analysis for the collapse of an empty cavity to calculate the wall motion and the Noltingk-Neppiras equation to determine the minimum bubble radius. When this is done and several minor approximations made, the fraction of the potential energy of the bubble that is radiated as sound is given by:

\[
\frac{E_{ac}}{E_{pot}} = 0.004 \sqrt{P_A(P/0)}
\]

where \( P_A \) is the ambient collapse pressure in atmospheres.

Eq. 2 was obtained by ignoring physical effects which slow bubble collapse and thereby reduce sound radiation when permanent gas contents are below 1 or 2%. For such very low gas contents the equation may predict sound energies exceeding the initial potential energy of the bubble. Recognizing that no more than about 2/3 of the potential energy can ever be radiated as sound, one can estimate the sound radiated for low gas contents without carrying out an analysis involving compressibility, heat conduction, etc. Eq. 2 is used for all cases for which it predicts less than 40% radiation. For lesser gas contents, a smooth transition is made to an ultimate limit of 67%.

The present analysis is a further simplification of one published by the writer (4) and is also related to an analysis by Khoroshev (5).

REFERENCES

(1) R. Hickling and M. S. Plesset, Phys. Fluids (1964) 7, 1-14
(3) Lord Rayleigh, Phil. Mag. (1917) 34, 94-96
INTRODUCTION

To gain a deeper insight into the relationship between bubble dynamics and acoustic cavitation noise. The temporal development of cavitation-noise spectra has been measured with high resolution in both time and frequency.

EXPERIMENTAL TECHNIQUES

Cavitation is produced by a PZT4-cylinder totally submerged in water at about 20 kHz. The driving oscillator and the data-acquisition are controlled by a minicomputer. The cavitation noise, which is monitored by a microphone, is low-pass filtered and directly digitized with conversion rates up to 1 MHz. The storage problem is overcome by assembling a buffer storage of 64 k bytes (8 bit). When the buffer storage is filled, the conversion is automatically stopped and the data are transferred to a magnetic tape unit at a rate of about 10 kHz. The spectral development of the noise is calculated on a large computer. Taking 4 k samples for each short-time spectrum we get a spectral resolution of 122 Hz and a resolution in time of about 4 msec at a sampling rate of 500 kHz.

RESULTS AND CONCLUSIONS

Several measurements in water were done in this way. As shown by Esche [1], there are strong sub- and ultraharmonic components at n/2 the driving frequency, if the driving pressure has passed the cavitation threshold. The temporal development of these selected spectral lines at f₀/2, 3f₀/2, 5f₀/2 and 9f₀/2 is shown in Fig. 1 together with the driving pulse (dashed line). A rather sharp onset of the cavitation noise at the threshold is observed as well as the interesting feature, that all observed spectral lines are of fairly equal amplitudes and set in simultaneously with the first subharmonic. The loss in amplitude at 48 msec could not be explained yet.

Many more experiments were done using different pulse shapes. The results strongly support a new hypothesis for the generation of the first subharmonic [2]. Further results are that in no case a subharmonic at f₀/3 could be observed and that the best lines to determine the cavitation threshold may be the lines at 5f₀/2 or 9f₀/2 instead of the first subharmonic.

[1] Esche, R., Acustica 2 (1952) AB 208-218
INTRODUCTION

In two earlier papers (1-2) the author developed a theory for the expansion of a high pressure gas bubble in a fluid-saturated porous medium. The special cases of an incompressible fluid (1) and a compressible fluid (2) flowing through a rigid porous solid matrix, respectively, were studied. It was found that the Darcy resistance factor exerted a profound effect on bubble expansion, and (in the compressible case) on the acoustic radiation field.

In this paper, the rigid frame restriction is dropped, the earlier work extended, and the much more formidable problem of two-phase flow, with both fluid and solid compressible, is considered.

THEORETICAL BASIS

The porous medium is treated as a continuum whose characteristics are described in terms of its hydrologic parameters, porosity and permeability, and suitable density and elastic constants, derived from the properties of the fluid and solid constituents.

The behavior of the fluid and solid components are determined by the Biot equations (3). This pair of simultaneous partial differential equations, solved by the method of Laplace transform, is matched to the boundary conditions at the surface of the expanding bubble, and the equations of motion of the bubble radius derived. Solution to this equation yields the radius-time curve for the expanding bubble, and the accompanying radiation field, in the so-called "acoustic approximation", well known in underwater gas bubble dynamics (4).

RESULTS AND CONCLUSIONS

In the general case, the most pronounced effect on the bubble expansion is expected under conditions where the relative velocity between fluid and solid frame is greatest; conversely, the least effect is obtained when the fluid and solid move together.

In general, the expanding bubble radiates acoustic waves of the first and second kind; but, since the limiting cases really represent single phase flow, only a wave of the second kind is expected in the former, while a wave of the first kind propagates in the latter case, respectively.

REFERENCES

(1) D. Epstein, J. Geophys. Res. (1967) 72 3701-3710
ASYMPTOTIC GRAZING FIELD ON A LOCALLY EXCITED, HOMOGENEOUS FLUID-LOADED PLATE BELOW CRITICAL FREQUENCY

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INTRODUCTION

The problem of concern, as shown in Fig. 1, is the pressure field on the surface of a locally driven plate below the critical frequency. The solution of this problem relates to the propagation of sound along ship hulls. We consider here only the grazing field propagating principally in the fluid with sonic speed. We have excluded the flexural wave carried principally by the plate. As far field radiation is directive with a null at grazing, Ref. (1), a higher order asymptotic solution is required to describe the pressure on the plate surface.

THEORETICAL BASES

For a line force excitation the fluid pressure is determined by the Fourier transform integral

$$p_{fl}(x, z) = \frac{F}{2\pi} \int \frac{1}{1 + \frac{1}{Y_p} \frac{k_0^2}{C^2}} \frac{J(\alpha x + \sqrt{\frac{1}{C^2} - k_0^2} z)}{dk}$$

where $Z_0$ and $Y_p$ are the wavenumber/frequency spectral plate impedance and fluid admittance, respectively. Below critical frequency the plate impedance will be purely massive and wavenumber independent for $k < k_0$. The dominant low wavenumber contribution to the integral is evaluated by stationary phase methods for $\alpha \neq 0$. However, for $\alpha = 0$, no stationary phase point exists. The dominant low wavenumber contribution is found to be due to the contribution to the integral around $k_0$. Evaluation of the integral for $\alpha = 0$ is made by deformation of the contour of integration, transformation of variables, and $k_0 \rightarrow \infty$ asymptotic approximation of the resulting integral which has an exponential factor $e^{-\left(x/k_0\right)C}$ in the integrand.

RESULTS AND CONCLUSIONS

We find that the asymptotic solution for the grazing pressure field falls off with reciprocal distance by one integer higher power than the nongrazing far field solution as summarized in Table 1.

Note, as a point of application, that hull grazing pressures should not be inferred from far field radiated noise measurements by simply applying a 30 or 40 dB attenuation to the radiated power source level. Rather these power laws should be used along with an additional range independent correction to account for the different constants in the two transfer functions of concern.

The results can be interpreted by noting that the effect of the plate is the same as that of a layer of fluid of mass equal to the plate, as shown in Fig. 2. The field at and near grazing is the same as that determined by the resulting Lloyd's Mirror phenomenon.

REFERENCES

Surfaces in contact with flow often vibrate. Sometimes, as in acoustic liners, they do so because they are put there to absorb flow borne sound, but they may be deliberately forced, as in the SONAR transmitter. In either case the motion is governed, through the surface’s mechanical properties, by the external driving force plus the fluid loading. If the fluid loading is heavy enough it is bound to play a crucial role in determining the response. How easily can it be estimated? Linearized potential theory is perfectly adequate if the wavelength of surface undulation is much greater than any viscous or boundary layer length scale (1,2). But there are important situations where it fails.

Ffowcs Williams and Lovely (3), in a recent paper, examined how flow modifies the impedance of a circular piston vibrating in an infinite plane baffle bounding a nominally uniform flow. The most significant flow effect turned out to be when mean flow suction brought an initially detuned piston to resonance. But, important as the suction was, they were unable to estimate its magnitude, linearized potential theory having failed to deal with the sharp piston edges properly. That difficulty moreover was not a feature of the real situation at all: there the piston actually protrudes into a turbulent boundary layer where potential modelling offers a poor description of the flow.

In this paper we reconsider the whole question of the pressure on a surface vibrating in a mean flow, and formulate the analysis in a way which avoids some of the problems which attend potential modelling. We take the high Reynolds number limit, neglect viscosity and solve Poisson’s equation for the pressure using Green’s function $G$ for the problem of a vortex sheet at a finite distance from a rigid boundary. We insist that $G$ is finite everywhere for all time, a constraint which makes our solution non causal. The relevance of this result to the piston problem, and, in a wider context, to the problem of turbulence stimulated by monopole forcing is discussed.

REFERENCES

ON THE ROTATIONAL NOISE SPECTRUM OF THE CAVITATING SCREW PROPELLER

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INTRODUCTION

Rotational noise of a screw propeller operating in the non-cavitation regime has a discrete spectrum with the peaks at the frequencies $f_n = nf_0$, $n=1,2,...$, where $b$ is the number of blades, and $f_0$ is the frequency of the rotation $[1,2]$. However, when the screw is cavitating, some new peaks in addition to $f_1, f_2, ...$ were found by Aleksandrov $[3]$. According to Dokuchaev $[4]$, there are two additional peaks per each basic line. On the basis of $[1-4]$ the form of the screw propeller rotational noise spectrum is therefore envisaged to be as follows: the spectrum is discrete and non-cavitation and cavitation regimes differ with respect to the number of lines associated. The subject of this paper is the formulation and verification of a hypothesis which changes this image.

HYPOTHESIS

In the developed-cavitation operating regimes of the screw propeller, there are cavities rotating together with the blades. These cavities increase blade thickness and also affect the transfer of the forces from the blades to the water. Thus, these cavities change the simple-source and dipole strength of the screw. The dynamic behaviour of the cavities is partially random in nature, so that the rotational noise source has not a fully periodic character. As a consequence, the spectrum lines of the cavitating screw propeller rotational noise become broader.

EXPERIMENT

The far field measurements of the basic spectrum component of the rotational noise generated by the strongly cavitating screw of a ship were taken by means of the hydrophone that was fixed in relation to the bottom. The ship was cruising at a constant speed and a steady course. The figure shows the frequency ($f$) dependence of the maximum amplitude of the signal received by the hydrophone in some small angle interval around the screw rotation plane. The line broadening of 5% has been obtained. This amount exceeds the one that could have occurred due to rotation frequency instability or Doppler effect and could result from the proposed cavitation effect. Some further experiments were performed, which did not show up such a broadening, so that the hypothesis was neither proved nor contradicted so far.

REFERENCES (all in Russian)

THE THEORY OF FLUCTUATIONS FOR LONG RANGE SOUND PROPAGATION IN THE OCEAN

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INTRODUCTION AND THEORY

Sound field fluctuations are believed to be caused primarily by naturally-occurring temperature fluctuations. The stochastic, Wiener-Hermite expansion used here for this problem is based on the white noise process, $H(x)$, with the properties that it is Gaussian at each point, statistically independent at different points and with the covariance

$$<H(x_1)H(x_2)> = \delta(x_1-x_2).$$ (1)

This process can be used to represent any Gaussian process using a suitable (to be found, usually) deterministic kernel, $P_1$:

$$P_1(x) = \int P_1(x-x_0)H(x_0)dx_0,$$ (2)

where $P_1$ is a Gaussian random function. To represent non-Gaussian portions of a random process one employs polynomial combinations of $H$ with weighting coefficients chosen from those of the generalized Hermite polynomials. Any random process can be represented by a series of such terms, beginning with a term like (2) (and in general including a deterministic portion, $P_0$).

In the present work we treat the Helmholtz equation (single frequency) with random coefficients,

$$[\nabla^2 + k^2(1+\delta c/c^2)]P(x) = 0$$ (3)

where

$$P = P_0 + P_1 + P_2 + \ldots$$ (4)

with $(2\delta c/c) + (\delta c)^2/c^2$ assumed Gaussian. By construction the different terms in (4) are statistically orthogonal; as a result a hierarchy of coupled equations for $P_n$ is easily obtained.

One of the advantages of this approach is that $P$ itself is expanded (rather than moments as in the usual case). The extent of Gaussianity of the fluctuating field is obtained, as is not the case for most other methods. Additional advantages of this method will be discussed.

RESULTS

The lowest order term (in $\delta c$) for the sound pressure correlation function is the same as that found in previous work. In the high frequency limit the method permits the calculation of a limited number of higher order scattering terms. These are used to check the validity (or lack thereof) of the widely accepted Rytoff approximation. Results for regions where the pressure-fluctuations dominate will be discussed.
IV. Physical acoustics

M. Molecular acoustics
HYPERSOUND IN PHYSICAL INVESTIGATIONS

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INTRODUCTION

In this paper we discuss the possibility of using acoustic magnetic resonance in the physical investigations: acoustic dynamic nuclear polarisation and quantum detection of acoustic phonons.

THEORETICAL BASES

A nuclear polarisation is the creation of a steady-state of oriented nuclear spins, and is used in nuclear physics experiments (nuclear targets). The nuclear spins in crystals with paramagnetic impurities are usually oriented owing to their interactions with electron spins. By applying a microwave field of suitable frequency, it is possible to induce electron and nuclear spin flips simultaneously, resulting in a strongly non-Boltzmann nuclear population distribution or nuclear dynamic polarisation (DNP). The application of hypersonic saturation gives new possibilities for the creation of DNP in large-species of conductors and the generation of coherent phonons. It is possible to use dynamically oriented nuclei for the detection of acoustic phonons created at different physical phenomena in solids.

EXPERIMENTAL TECHNIQUES

For the polarisation of nuclei $^{27}$Al in ruby we used acoustic magnetic saturation of paramagnetic electron spins Cr$^{3+}$. The hypersound the frequency transitions of ions Cr$^{3+}$ was generated by lithium niobate single crystal which was placed into microwave resonator. The change in the DNP was measured with respect to the change in the intensity of $^{27}$Al nuclear magnetic resonance signal.

The quantum phonon detector is based on the above principle. Measured acoustic vibrations are generated in the quartz rod, which was bonded to the detector crystal (ruby crystal). The acoustic phonons are detected with respect to the change in the polarisation of $^{27}$Al nuclei by electromagnetic or acoustic saturation of Cr$^{3+}$ electron spins in ruby.

RESULTS AND CONCLUSIONS

The enhancement of DNP (more than 200 times) was observed by acoustic saturation with frequency 9.5 GHz in ruby at 4.2°K. The sensitivity of the quantum phonon detector was demonstrated by the observation of acoustic vibrations with the relative deformation amplitude of the order of $10^{-7}$. 
The problem is to define and calculate the constant $Q$ of interaction between exciton and phonon. This constant depends on the type of phonon/acoustical or optical/ and kind of assumed model of lattice. In general case the hamiltonian of the displaced atoms should be analyzed. For the exciton wave functions we find the matrix of derivatives of hamiltonian related to normal coordinates. Introducing the operators of creation and annihilation we obtain the quanta presentation of the problem.

We analyze two limit cases:

a/ the states of neighbouring atoms have only a very small area of the overlapping, all electron states are strongly localized.

The pairs electron - hole exist on one site.

This is the Frenkel model.

b/ Electron orbits are much greater than the interatomic spacing and overlapping area is large. Electron and hole are respectively in the conduction and valence bands, the interaction of hole and electron is weak. This is the Wannier model.

By first model it is possible to use the wave function of the exciton as a whole, in the second one the calculation should be carried on separately for electron and hole. As a result we obtain the constant $Q$ expressed by wave vector and characteristic constants of lattice and exciton.

Depending on the kind of the excitons model the interaction inside one zone or between the zone should be assumed. The widening of spectra lines is also calculated.
APPLICATION DE LA MÉTHODE DES ÉTATS COHÉRENTS POUR ÉTUDES DES EFFETS QUANTO-ACOUSTIQUES

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INTRODUCTION

La méthode traditionnelle de présentation des oscillations acoustiques, appliquée au principe en vigueur des états stationnaires non-influencés aux chiffres de phonons définis avec précision, dans le cas d'une onde acoustique typique gigahertz (10^8 W/cm²) se propageant au corps solide, où il faut prendre en considération 10^9 phonons dans 1 cm^3 - paraît être peu utile.

Bases théoriques

À base d'abstraction de la mécanique quantique, l'onde acoustique est un état quantique macroscopique. On peut la traiter comme une déformation appliquée aux autres systèmes quantiques, représentée par un certain opérateur D dépendant uniquement des déclins des atomes du réseau cristallin en position d'équilibre. En agissant à l'aide de cet opérateur sur l'hamiltonien du système principal H, on obtient un hamiltonien du système déformé par l'onde acoustique H_d, qui est la somme des hamiltoniens du système principal, de l'onde acoustique et de la réaction réciproque:

$$D^*HD = H_d = H + H_{ac} + H_{int}, \quad D^*D = 1.$$  (1)

L'opérateur de déformation a les propriétés de créer les états cohérents: $$|\omega\rangle = D|0\rangle,$$ qu'on peut définir comme des fonctions propres des opérateurs d'annihilation $$\alpha$$:

$$a|\alpha\rangle = \alpha|\alpha\rangle,$$  (2)

où $$\alpha$$ est une valeur complexe propre, dépendante du déclin causé par l'onde acoustique. Le hamiltonien de l'onde acoustique est donc une grandeur quasi-classique, dépendante des coefficients de déformation:

$$H_{ac} = \sum_k |\alpha|^2 \omega_k \alpha^* \alpha.$$  (3)

RÉSULTATS ET CONCLUSIONS

La méthode des états cohérents a été introduite premièrement en acoustique quantique en 1972 par Sokolov(1) et Szumilin(2) dans le but de calculer la réaction de l'onde ultrasonore avec les oscillations thermiques du réseau. Tout compte fait, on obtient un modèle écrivant la propagation du deuxième son(2). Ensuite, cette méthode a été appliquée par l'auteur à la recherche de l'effet du soulèvement phononique /phonon drag/ et provenant de la force acoustoélectrique analogique à la force thermoelectrique /thermoelectric power/(3).

La méthode des états cohérents constitue une grande facilité de résoudre les problèmes où se trouvent les équations de transport décrivant les processus qui ont lieu dans les systèmes déséquilibrés déformés par l'onde ultrasonore.

RÉFÉRENCES

(1) A.I. Sokolov, JETF (SU), (1972), 62, 4, 1496.
LASER GENERATED ACOUSTIC WAVES IN SOME LIQUIDS

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INTRODUCTION

The effect of acoustic waves generation by the absorption of light is well known since the end of the last century from the works of Bell, Roentgen and Tyndall (see cf. ref. (1)). Recently there has been done a revival of this effect which is now generally known under name photoacoustic. The present report is devoted to the experimental study of sound generation in some liquids by light pulses of ns and ps duration and medium energy.

THEORETICAL BASES

If the thermal conduction in the medium could be neglected, the basic equations describing the effect are:

\[ \frac{\partial^2 p}{\partial t^2} = \frac{\beta}{c_p} \frac{\partial H}{\partial t} \]

\[ H = \alpha I \exp(-\lambda z - i\omega t) \]

where \( p \) is the acoustic pressure, \( \beta \) is the logarithmic coefficient of thermal expansion, \( c_p \) is the specific heat per unit volume per unit time absorbed, \( I \) is the output intensity of the laser, \( \lambda \) is the attenuation coefficient for the laser beam propagating in the \( +z \) direction modulated at frequency \( \omega \). The equations were solved for different boundary conditions by authors (1-2).

EXPERIMENTAL TECHNIQUES

As a light source served the Nd:YAG laser with the electrooptic modulator or dye switch for ns or ps pulses respectively. The acoustic pressure pulses were monitored either by means of radially polarized piezoceramic ring in the form of simple quatta or piezoceramic diacs of different bandpass width. The signal from the transducer was directly displayed on laboratory oscilloscope or the double beam storage oscilloscope.

RESULTS AND CONCLUSIONS

It is shown that the generated sound pulses are of the width less than 1 ps for pure water and narrower for liquids with greater absorption coefficient. If the total energy of light pulses does not exceed the value of mechanical or dielectric strength of liquid, the amplitude of pressure pulse depends linearly on total input energy of laser pulse and coincides for both ns and ps light pulses. Acoustic power varies quadratically with the power of incident light pulse. The obtained results for pure water are found to be in good agreement with the above theory.

REFERENCES

(1) P.J. Westervelt, R.S. Lenson, J. Acoust.Soc.Am. (1973) 54, 121
(2) F.V. Bunkin, V.M. Komissarov, Akust.Zhur. (1973) 19, 305
Some results about sound generation by lasers in different media are reviewed. For the acoustic wave excited in the immovable medium can be realized Mach numbers $M_0 \sim \phi I_o / \rho_o C_0 C_p$. Here $I_o =$ light intensity, $\phi =$ volume expansion coefficient, $C_0 =$ sound velocity, $\rho_o =$ density, $C_p =$ specific heat. Mach number depends on the geometry of light absorption region and on the value of parameter $\lambda = \alpha C_0 \tau_L$ ($\tau_L$ is duration of laser pulse or intensity modulation characteristic time, $\alpha =$ light absorption coefficient).

Analogous effect takes place when the radiation is unmodulated, but variable in time heating of the medium is the result of mass transport by free flowing stream. At supersonic streamlining of laser beam with Gaussian transverse intensity distribution $I(x,y) = I_0 \exp[-(x^2 + y^2)/a^2]$ the density displacement $M = \phi / \rho_o$ along $x$-axis (which is parallel to the stream) is equal to

$$M(x,y=0) = \frac{\sqrt{2} \phi a}{2(1-a^2)} \left[ \frac{x^2}{B^2} \right] - \frac{1}{\sqrt{1+B^2}} \left[ 1 + \Phi \left( \frac{B}{\sqrt{1+B^2}} \frac{x}{a} \right) \right]$$

Here $B^2 = (V/C_o)^2 - 1$, $V =$ stream velocity.

In the report are considered in detail problems of sound generation in moving and immovable medium at different geometry and laser radiation regime. Approximate models taking into account nonlinearity and dissipation effects are discussed. Theoretical results are compared with experimental data.
When the laser pulse falls upon the boundary between the transparent and the light absorbing medium, in the subsurface layer, at the distances of the order of light travel length, fast energy discharge leading to the sound radiation occurs.

Acoustical radiation is usually determined by the effect of heat expansion of the volume into the medium, in which energy release occurs, when the densities of released energy are reasonable. This effect is proportional to the density of released energy and to the ratio of heat expansion coefficient to medium heat capacity $C_p$.

Along with, the characteristics of acoustical radiation depend on the geometric parameters, particularly, on the form of the region of energy release and on the relationship of its characteristic dimension $a$ to the characteristic length of the acoustical signal equal to the sound travel length in time $\delta$ equal to the duration of the light pulse $\lambda = c \delta$, where $c$ is the sound speed. If the dimensions of the radiating area are small, then the characteristics of the acoustical signal are determined by the speed of the energy discharge. If the region is large, $a > \lambda$, then the excessive pressure is appeared in it as a result of the energy release, leading further to the radiation of the relaxation wave. In this case the excessive pressure is determined by released energy and does not depend on the speed of the energy release, and signal duration is determined by the time of sound travel along the disturbed region.

For the case when the form of the region of energy release is close to the cylindrical one (of the radius $Q$ and the length $\ell$) signal amplitude in the transversal direction is determined by the expression

$$ P_t = \frac{E \cdot c^2}{C_p S \tau} $$

where $S = \pi a^2$ is the cross-section area of cylindrical radiating region, $\tau = \frac{\delta}{c}$ is the duration of the pulse. In the longitudinal direction signal amplitude is determined by the expression:

$$ P_l = \frac{E \cdot c^2}{C_p S \ell \tau} $$

and the pulse duration is $\tau = \frac{\lambda}{c}$, thus $P_l = \frac{\ell}{C_p}$ because of the fact that in the transversal direction the signals are added according to their amplitudes, and in the longitudinal direction - according to its durations.

REFERENCES:

INTRODUCTION

The influence of interstitial impurities (O,N,C) on hydrogen mobility in b.c.c. transition metals has been established by various authors (1-3) while there is inadequate information about the role of substitutional impurities on the H diffusivity.

This paper presents the effect of substitutional titanium on the diffusion parameters of H in niobium.

EXPERIMENTAL TECHNIQUES

The dissipation coefficient $\zeta^{-1}$ of the damping of thin sheets (7-10 $\mu$ m thickness clamped at one end and freely vibrating in the Hz range) was measured as a function of temperature from 140 to 400 K. Samples were niobium base alloys containing 2 and 5 at-% titanium as a substitutional solute. They were electrolytically charged with hydrogen whose concentration in the specimens was kept below 0.2 at-% in order to avoid hydride formation. Hydrogen content was determined by the electrical resistivity increase.

RESULTS AND CONCLUSIONS

After the hydrogen loading the curves of the dissipation coefficient $\zeta^{-1}$ vs. temperature display a pronounced peak. This effect is due to the long-range diffusion of interstitial hydrogen through the specimen, induced by the periodic gradient in dilatation caused by the vibration (Gorsky effect) (4). The diffusion coefficient $D$ vs. temperature $T$ can be deduced from the analysis of the $\zeta^{-1}$ experimental curves according to the Gorsky theory.

The present results show that an exponential dependence of $D$ exists upon reciprocal temperature for each Nb-Ti alloy over the whole temperature range investigated. The straight lines obtained plotting $\ln D$ vs. $1/T$ shift towards lower values of $D$ with increasing Ti content and, at the same time, the corresponding slopes increase.

By comparison with the data on pure niobium (5,6) it can be observed that the values of $D$ of the alloys are significantly lower than those of the unalloyed material, yielding a value about 50 times smaller at 160 K for the 5 at-% Ti alloy. These results are explained in terms of trapping of hydrogen by substitutional titanium.

REFERENCES

4) W.S. Gorsky, Phys. Z.SU (1935) 3, 457
EFFECT OF $\gamma$ - IRRADIATION ON THE GLASS TRANSITION TEMPERATURE OF POLYETHYL-METHACRYLATE

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INTRODUCTION

Most of the irradiation effects in polymers studied so far have been concerned with the final chemical changes and the resultant modification in physical properties. Of increasing interest are the effect of irradiation on the molecular transition and its relation to the viscosity average molecular weight. Heat treatment of polymers after irradiation can change the effect of radiation retained by the specimen. It is to our knowledge, the first time to be reported data on the glass transition of polymers by ultrasonic technique and how it could be altered by time and heat treatment.

EXPERIMENTAL

Polyethyl-methacrylate PEMA chosen for this study, was molded to form circular transparent discs suitable for ultrasonic measurements. Investigation of the change of the absorption of longitudinal ultrasonic waves of frequency 4.0 MHz was carried out in the specimen as a function of temperature through the glass transition temperature of the polymer. The sample was then exposed to a dose of $\gamma$-irradiation equal to $3.3 \times 10^6$ rad., at an ambient temperature of $30^\circ$ C. Measurements were made immediately on the specimen after being removed from the source, and then repeated daily to study the effect of time and heat treatment on the molecular transition of the polymer.

RESULTS AND CONCLUSIONS

On the temperature-longitudinal ultrasonic absorption curve, the glass transition in PEMA is associated with a well defined peak which occurs at $47^\circ$ C. After $\gamma$-irradiation the peak shifts on the temperature scale and its amplitude increases substantially. Increasing the time after irradiation and exposing the specimen to heat treatment will result in decreasing the peak height and temperature till it reaches an asymptotical value. The decay follows an exponential curve. The variation of $T_M$ can be represented by the equation

$$t = \frac{E}{kT_M}$$

where $t$ is the time elapsed after irradiation, $E$ is the activation energy of the decaying process, $T_M$ is the temperature at which the maximum of the peak occurs. The activation energy of the process calculated according to the above equation is given by $1.1587$ K. Cal. / mole.
In the past years the so called \( \alpha \)-peak has been widely investigated in some bcc metals and it was attributed to various different relaxation mechanisms. The aim of this work was to clarify the effects due to the presence of interstitial hydrogen in permanently deformed specimens of niobium.

The dissipation coefficient \( Q \) of flexural vibrations in the 20-100 KHz range was measured from room temperature to LN\(_2\) temperature.

The results showed first of all that the \( \alpha \)-peak is just an hydrogen cold-work peak. In fact, the specimens deformed after an annealing and degassing treatment in an ordinary oil diffusion pump vacuum (10**-5** torr) gave a substantial \( \alpha \)-peak while after a similar treatment carried out in an oil free UHV vacuum furnace (10**-8** torr) they presented no \( \alpha \)-peak at all also for larger deformations. Subsequent hydrogen charging produced in the first case a large increase of the already present dissipation peak and in the second case the appearance of a similar large peak. Aging treatments at temperatures just above room temperature produced in any case a continuous decrease of both the dissipation peak and the temperature \( T \) of its maximum value \( Q_m \) but with rates depending on the hydrogen content.

In order to frame all the experimental facts within a simple model, the dissipation mechanism due to dislocation dragging of the hydrogen atmosphere discussed by Schoeck was improved by a new formulation which takes into account an "instantaneous" deformation of the dislocation line between two hydrogen atoms plus a "relaxing" deformation of the dislocation line between two pinning points. The motion of the dislocation line is then given by:

\[
A \frac{dx}{dt} + Bx = C \frac{\dot{\gamma}}{\dot{\gamma}_0} + \frac{\dot{\gamma}^2}{\dot{\gamma}_0^2} \frac{1}{\dot{\gamma}_0^2} \frac{1}{\dot{\gamma}_0^2} \exp(-\frac{W}{kT})
\]

where \( A \) is proportional to the linear hydrogen concentration and inversely proportional to the diffusion coefficient of the hydrogen atmosphere, \( B \) is inversely proportional to the dislocation segment \( 1 \) between two pinning points and \( C \) is proportional to the square of the dislocation length \( l_H \) between two hydrogen atoms. From this model, in the simple case of a single relaxation time \( \tau = A/B \), one obtains:

\[
Q_m^{-1} = \frac{1}{\dot{\gamma}_0^2} (1 - 1_H^2) \quad \text{and} \quad \tau \propto \frac{1}{\dot{\gamma}_0^2} \frac{1}{\dot{\gamma}_0^2} \exp(-\frac{W}{kT})
\]

where \( W \) is the activation energy of the relaxation process.

From these equations it follows:

\[
\ln \left[ \left( Q_m^{-1} + \frac{1}{\dot{\gamma}_0^2} \frac{1}{\dot{\gamma}_0^2} \right) \frac{1}{\dot{\gamma}_0^2} \right] = -\frac{W}{kT} + \text{const.}
\]

These equations basically say:

i) when \( \dot{\gamma} \ll \dot{\gamma}_0 \), i.e. when the hydrogen concentration is less than that of the pinning points, there is no dissipation peak.

ii) increasing the hydrogen concentration the \( Q_m^{-1} \) increases rapidly towards a limiting value which depends on the density of the dislocation lines \( \Lambda \) and on the density of the pinning points.

iii) when the hydrogen concentration is large enough, the term \( \frac{\Lambda}{\dot{\gamma}_0^2} \) is negligible in comparison to the \( Q_m^{-1} \) of eq. (2) which coincides with the similar equation given by Schoeck's theory.

Our experimental results on niobium are perfectly accounted for by all the predictions of this new model.

THE RELATION BETWEEN ENERGY GAP AND VELOCITY OF ULTRASONIC WAVE PROPAGATION IN SEMICONDUCTORS

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INTRODUCTION

In this work the existence of a dependence between the energy gap and the velocity of ultrasonic wave propagation for the polycrystalline semiconductors of group II-V and monocrystalline semiconductors of group IV, III-V and II-VI has been shown. Using the acoustical method, the change of the energy gap with composition for the solid solutions has been obtained too.

THEORETICAL BASES

The interaction potential of atoms according to Lenard-Jones was assumed. The potential energy interaction of atoms can be expressed by the velocity of ultrasonic wave propagation. This energy is proportional to the energy gap [11,12]. Finally, between the energy gap and velocity of ultrasonic wave propagation for the polycrystalline semiconductors the following expression has been obtained:

\[ E_g = \frac{a}{2} \left( c_l^2 - \frac{1}{2} c_t^2 \right) + b \]

where \( c_l \), \( c_t \) velocities of longitudinal and transverse wave propagation, \( a \), \( b \) constants characteristic of the several group of semiconductors.

On basis of the paper [13] and formula (1) the change of energy gap with composition for the solid solutions has been obtained.

RESULTS AND CONCLUSIONS

The dependence between energy gap and velocity of wave propagation for semiconductors of group II-V for example is presented in Fig. 1. This dependence has a linear character. This fact confirms the reasonableness of formula (1). These dependences for the other group of semiconductors have been derived too.

The obtained change of the energy gap with composition for the solid solutions has a nonlinear character.

REFERENCES

(2) N. N. Shirota, Khimicheskaya svyaz v poluprovodnikakh i termodynamika, Minsk (1966).
THE INFLUENCE OF SURFACE ON RAYLEIGH'S SURFACE WAVE PROPAGATION

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INTRODUCTION

In this work the influence of surface states on the surface wave propagation has been investigated. The system piezoelectric-semiconductor has been considered. Determination of the characteristic parameters of these surface states using an acoustical method has been proposed, too.

THEORETICAL BASES

If an acoustic wave interacts with the carriers, the existence of relation between their surface concentrations on the surface traps $n_t$ and the bulk concentration of carriers in conduction band at the surface $n_0$ can be shown. It is as follows:

$$n_t = \frac{q\tau}{1 - i\omega\tau} n_0$$

where $\tau$ - time of relaxation for the several of trapping levels, $g = W_v N_t - n_t$, $W_v$ - capture cross section of the carrier charge by trap, $v = \frac{e\nu}{k_B T}$ thermal velocity, $N_t$ - density of trapping levels, $n_{eq} - equilibrium carriers concentration on the trapping levels.

By calculations resulting from known equations of: motion, piezoelectric effect, Poisson's and continuity can be shown, that the coefficient of damping wave resulting from wave interaction with electrons is as follows:

$$\nu = \frac{e \eta H}{\xi} \omega \tau^2 M (1 + \frac{\xi}{\xi_0})^2 + \frac{\omega^2 \tau^2 (1 + \xi_0)^2 (1 + \frac{\xi}{\xi_0})}{(1 + \frac{\xi}{\xi_0})^2 + \omega^2 \tau^2 (1 + \frac{\xi}{\xi_0})^2}$$

symbols as in the paper [1], $a = \frac{\xi_0}{1 + \omega^2 \tau^2}$, $b = \frac{\xi}{1 + \omega^2 \tau^2}$.

CONCLUSIONS

From the formula (2) it results, that surface states displace the value of the critical field towards greater values of the field.

$$E_{kr} = \frac{\nu^4}{\mu} \left(1 + \frac{\rho}{\sigma} \frac{\omega^2 \tau^2}{1 + \omega^2 \tau^2} \right)$$

where $\mu$ - carriers mobility.

From the obtained curve the characteristic parameters of surface states can be determined by investigating the dependence of the critical field on the expression $\omega^2 \tau^2$.

REFERENCES

UN MODELE ACOUSTIQUE DE L'EAU

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INTRODUCTION

La variation parabolique de la vitesse de propagation des ultrasons avec la température dans l'eau résulte de l'addition des deux effets opposés de l'agitation thermique : l'effet vibratoire et l'effet structurel.

THEORIE

En vue de déterminer quantitativement ces deux contributions on a admis le modèle des deux espèces : des molécules associées par l'intermédiaire des liaisons d'hydrogène et des monomères, calculant les propriétés acoustiques de ces composants.

Dans ce dessein on a supposé la valabilité de certaines règles d'additivité et de l'indépendance de la température de la constante de Rao pour chaque espèce moléculaire du modèle.

RESULTATS

En utilisant les données fournies par la spectroscopie I.R., on détermine les densités et les valeurs de la vitesse de propagation de l'ultrason dans chacun des deux composants, dans l'intervalle de température : 0 - 100°C à la pression normale.

On poursuit la variation de la vitesse avec la température dans les composants posés et dans des mélanges de différentes concentrations, ainsi que la dépendance de la vitesse de l'ultrason de la fraction de monomères à différentes températures constantes.

DISCUSSION

L'on constate un accord satisfaisant avec les résultats expérimentaux pour les concentrations d'équilibre correspondantes aux températures respectives.

On calcule ensuite les termes vibratoires et structurels de la vitesse de propagation de l'ultrason, nécessaires à une interprétation quantitative de l'anomalie du comportement de l'eau au point de vue acoustique.

Le travail s'occupe de la déviation de la loi de Rao par l'introduction et l'évaluation d'un incrément de la liaison d'hydrogène.

Sont présentées de même les compressibilités adiabatiques des monomères et des groupements associés, évaluant la contribution de compressibilité des interactions des deux espèces moléculaires.

La méthode est étendue aussi sur les solutions aqueuses de sel, l'accord avec les résultats expérimentaux étant favorables au modèle, autant en ce qui concerne l'interprétation de l'effet de l'agitation thermique que des interactions ion-solvant.

BIBLIOGRAPHIE :
(1) Luck, W.A.P.; Disc.Farad.Soc.43, 115, 1967
(2) Kaulgud, M.V., Ind.I.Phys. 36, 11, 577, 1962
(3) Leroy, Y., Liebaert, R., C.R. 257. 642. 1963
RECHERCHES ULTRASONORES SUR LA STRUCTURE DE QUELQUES LIQUIDES ASSOCIES

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INTRODUCTION

Les équations avec lesquelles on peut mettre en corrélations la vitesse de propagation de l’ultrason et les propriétés structurelles des liquides - et aussi la loi de Rao - se limitent seulement à établir les déviations des liquides réels de l'idéalité, en se basant sur des interprétations à la fois contradictoires.

THÉORIE

Des considérations de l’acoustique moléculaire ont permis la déduction d’une expression pour la vitesse de l’ultrason et encore d’une grandeur qui ne dépend pas de la température, toutes les deux valables pour les liquides-ideaux, mais aussi pour des liquides avec des molécules associées.

\[ v = c_0 e^{\kappa(n-1)} \quad \text{et} \quad a = \ln \left( \frac{v_m}{\rho} \right) \quad n = \frac{S + D}{S} \]

\[ v = \text{vitesse moyenne de molécules}; \quad a = \text{rendant d’heurt}; \quad D = \text{diamètre moléculaire}; \quad S = \text{distance intermoléculaire}; \quad \text{respectivement} \]

\[ v e^{3/a (n-1)} = R_A \]  

RÉSULTATS EXPÉRIMENTAUX

Pour vérifier les relations ci-dessus on a mesuré la vitesse de l’ultrason dans une série des liquides et solutions, caractérisés par différents moyens d’interaction moléculaire et ionique, en utilisant une méthode de diffraction de la lumière sur le réseau ultrasonique.

On a poursuivi la variation de la vitesse avec la température et la concentration des solutions aqueuses des halogénures alcalins et alcalino-terreux et on a calculé la constante de Rao et la vitesse moléculaire relative.

DISCUSSION

On constate un accord satisfaisant avec les relations (1) et (2). En évaluant les paramètres "a" et "n", on a trouvé : a(n-1) = valeur pour laquelle la relation (2) devienne identique à la relation empirique de Rao.

On sait que l’appréciation du degré de l’association moléculaire des liquides par la déviation de la constante de Rao n’est pas en concordance avec les données offertes par les spectres de diffusion combinée. En analysant la relation (2) on a constaté que la variation de la vitesse moléculaire avec la température ne résulte pas exclusivement de l’effet structurel de l’agitation thermique, par conséquent elle ne peut être considérée comme une mesure de l’association relative.

BIBLIOGRAPHIE

(1) Kaulgud, M.V., Acustica, 10, 316; 1960
The purpose of this paper is to show theoretically the physical basis for the empirically observed relation between the nonlinearity parameter of liquids and reciprocal sound speed. Nonlinear sound propagation in liquids is characterized (1) by a parameter, $B/A$, given by:

$$\frac{B}{A} = 2\alpha c \left( \frac{\partial \alpha}{\partial \rho} \right)_T + \frac{2c T a}{c_p} \left( \frac{\partial c}{\partial T} \right)_p$$

where $c$ is sound speed, $\rho$ is density, $\alpha$ is thermal expansion coefficient, $c_p$ is heat capacity at constant pressure, $T$ is absolute temperature, and $p$ is pressure.

It has been noted empirically, in what is called (1) Ballou's rule, that $B/A$ for various liquids is a roughly linear function of $c^{-1}$, but there has been, as yet, no explanation of why this relation should exist.

It will be assumed here that the bulk modulus, and hence sound speed, depends only on the potential energy between molecules. The potential energy, in turn, is taken to depend only on the intermolecular separation which is proportional to the cube root of the volume. Hence, the sound speed depends only on volume and not on temperature or pressure except as these variables affect the volume. It follows that:

$$c \frac{dc}{dV} = -\frac{\alpha c}{\rho} \frac{d\ln c}{d\ln V} \cdot \left( \frac{\partial c}{\partial T} \right)_p = \frac{ac}{\rho} \frac{d\ln c}{d\ln V}$$

and eq. (1) reduces to:

$$\frac{B}{A} = 2 \frac{d\ln(c^{-1})}{d\ln V}$$

Taking as an illustrative, but not restrictive, example, a Lennard-Jones potential, the sound speed becomes:

$$c^2 = 4U_0 \left[ 5\left(\frac{V}{V_0}\right)^{-4} - 3\left(\frac{V}{V_0}\right)^{-2} \right]$$

where $U_0$ is the heat of vaporization. Using eq. (4) in eq. (3) yields:

$$\frac{B}{A} = \frac{20 - 6\left(\frac{V}{V_0}\right)^2}{5 - 3\left(\frac{V}{V_0}\right)^2}$$

Combining eqs. (4) and (5) to obtain $B/A$ as a function of $c$ and expanding in powers of $c^{-1}$ about $c_0^{-1}$ (where $V = V_0$) leads to:

$$\frac{B}{A} = 4.85 + 12.1V_0 c^{-1}$$

Using a representative value of $U_0$ for liquids, eq. (6) is equivalent to Ballou's rule, but with different numerical constants. The agreement with experiment (1) is, however, as good as Ballou's rule.

REFERENCE

SOUND ABSORPTION IN He II AT LOW TEMPERATURES AND TRANSVERSE RELAXATION IN THE PHONON SYSTEM

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We have developed a theory of sound propagation and absorption in He II at low temperatures where only phonons take part in transport phenomena. In accordance with recent experimental data (and in contrast to the assumptions made in some early papers) the phonon velocity dispersion is supposed to be positive, though small. Because of the smallness of the dispersion two characteristic times of relaxation exist at low temperatures: the longitudinal time $\tau_L$ and the transverse time $\tau_T (\tau_L >> \tau_T)$. The longitudinal relaxation establishes equilibrium among the phonons propagating along each direction in the momentum space while the transverse relaxation establishes equilibrium among different directions.

We have calculated the sound absorption coefficient $\gamma$ and the sound velocity change $\Delta c/c$ in the region $\omega_L^{-1} < \omega < \omega_T^{-1}$ ($\omega_s$ is the ultrasonic frequency). In this region both effects are determined by the transverse relaxation, and we have

$$\gamma = \frac{\pi^2 (1+u) T^4 \omega_s^4}{\rho kT_0^3} \ln 6.01$$

$$\frac{\Delta c}{c} = \frac{\pi^2 (1+u) T^4 \omega_s}{\rho kT_0^3} \ln 6.01$$

where $T$ is the temperature, $\rho$ is the total density, $u$ is the Gruneisen parameter and $c$ is the sound velocity.

For sound intensity of the order of $\rho_n c^3 (c/\tau_L)^{-1/2}$ or higher where $\rho_n$ is the normal density nonlinear effects in sound absorption become important. The physical mechanism for this nonlinearity is heating of the phonons propagating in narrow angular interval near the sound propagation direction under conditions of slow transverse relaxation.
The absorption of travelling sound wave in He II by normal component is accompanied by energy and momentum transfer, and consequently the normal component is both heated and dragged. In the steady state the drag may be more important than heating which means that a stationary temperature difference is produced by the sound. We have calculated this temperature difference produced by the sound wave propagating in a tube filled with He II. The calculation is made under the following assumptions: 1) The reflection of the phonons from the walls of the tube is specular; 2) The condition $c/\gamma' \ll 1$ takes place where $c$ is the sound velocity, $\gamma'$ is its absorption coefficient and $\tau$ is the characteristic phonon time of relaxation. In this case the normal component is dragged by the sound as a whole.

The steady state solution exists under these conditions only if the input sound intensity $S$ does not exceed some critical value $S_c = 3^{5/2} \cdot 2^{-9} \cdot \rho_n c^3$ where $\rho_n$ is the normal density at the input end of the tube. If $S < S_c$, we have for the ratio of the temperatures at two ends of the tube

$$T_1/T_2 = 1 - S/\rho_n c^3 \quad (1)$$

In the opposite case, where $S_c - S < S_c$ we get

$$T_1/T_2 = (3^{5/2} 2^{-6})^{1/4} (1 - S/S_c)^{1/4} \quad (2)$$

Although this effect cannot make the temperature $T$ indefinitely small because the approximation used becomes poor when the product $c/\gamma'$ is of the order of unity, it can, nevertheless, lower the temperature substantially.
INTRODUCTION

The ultrasonic absorption in water and other associated liquids and in the corresponding ionic solutions, e.g., LiCl and KBr solutions, has been measured in the pressure range 0-2500 bar at frequencies between 50 and 100 MHz.

EXPERIMENTAL TECHNIQUE

The ultrasonic absorption was measured using the Debye-Sears method in a pressure cell made of stainless steel with two sapphire windows of 30 mm inner diameter. The ultrasonic wave was introduced via a delay rod. The temperature was varied between -30 and +80°C.

THEORETICAL BASIS

The excess ultrasonic absorption (1) may be quantitatively described as a function of pressure, temperature, type of liquid and ions using the following formula

\[
\frac{2\delta}{c^2} = \frac{\kappa_T}{c} \sum_{i=1}^{3} \sum_{j=1}^{2} \frac{(\nu_{ij} \nu_{i1} + \nu_{ij} \nu_{i2} + \nu_{ij} \nu_{i3})^2}{RTV_{i1} \frac{\nu_{ij}^2}{\nu_{i1}^2} + \frac{\nu_{ij}^2}{\nu_{i2}^2} + \frac{\nu_{ij}^2}{\nu_{i3}^2}},
\]

where \( c \) is the sound velocity, \( \kappa_T \) the total compressibility, \( V_{i1}, V_{i2}, V_{i3} \) the molar volumes and \( n_{i1}, n_{i2}, n_{i3} \) the number of moles of three discrete structural forms \( Ai, Bi, Ci \) in different regions \( i \) (i.e. bulk and first and second solvation layer of the anion and cation, respectively). Between these structural forms fluctuations are assumed to exist with mean relaxation times \( \tau_{ij} \) leading to the excess ultrasonic absorption of eq. (1).

RESULTS

The excess ultrasonic absorption decreases strongly with increasing pressure especially in solutions with ions that form a special liquid structure with longer relaxation times around themselves such as Li+.

However, a concentration of about 1M is necessary for the effect to become noticeable. The pressure dependence levels off at a few hundred bars. For some sufficiently highly concentrated solutions the excess ultrasonic absorption tends to increase again at higher pressures. The microscopic dynamical properties of the ionic solutions which result when the ultrasonic absorption data are interpreted according to eq. (1) will be discussed.

REFERENCES

ULTRASONIC ABSORPTION STUDY OF THE ASSOCIATION BEHAVIOR OF ANTIHISTAMINES IN AQUEOUS SOLUTION

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INTRODUCTION

In aqueous solutions some antihistamines show an association behavior of the micellar type (1). However, for several antihistamines equilibrium studies did not permit a choice between the micellar and a non-micellar type of association (1). This led us to study the ultrasonic absorption of bromodiphenhydramine (II), diphenylpyraline (II), diphenhydramine (III) and tripelenamine (IV) hydrochlorides, mepyramine (V) and chlorpheniramine (VI) maleates with the purpose of characterizing more precisely their association behavior. Indeed, it has been shown that the excess ultrasonic absorption of self-associating compounds presents characteristic features depending on whether the association is micellar (2) or non-micellar (3).

EXPERIMENTAL

The variation of $\alpha/f^2$ ($\alpha =$ absorption coefficient, $f =$ frequency) with the concentration $c$ of I and II (figure 1) and III presents an abrupt increase at above a certain concentration as in the case of a micellar association (2). The same behavior is observed for II with 0.1 M NaCl but at a lower $c$ than without salt. The antihistamines V shown on figure 1, and IV and VI present a monotonous increase of $\alpha/f^2$ with $c$ as for a non-micellar association (3). Spectroscopic measurements, between 1 and 155 MHz on solutions of 1, II, IV, V and VI reveal relaxation processes with relaxation frequencies between 2 and 20 MHz depending on the antihistamine and $c$.

DISCUSSION

The concentrations at which $\alpha/f^2$ shows an abrupt increase for I, II and III correspond also to an increase of scattered light (1). This result and the effect of NaCl indicate that the observed absorption is probably due to a micellar association of these compounds (2). The large value of $\alpha/f^2$ found for V and VI seems to indicate that this absorption is also due to an association process, but of a non-micellar type (3). The absorption of IV may be due to an isomerization reaction since the relaxation frequency is almost independent of $c$.

REFERENCES

LIGHT SCATTERING STUDY OF ULTRASONIC RELAXATION IN
DICHROMETHANE AND DIBROMOMETHANE

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INTRODUCTION

Recently, pulse technique has been used to measure liquids in the ultrahigh frequency range. However, absorption measurements are inadequate to study relaxation accompanied by a nontrivial dispersion.

We have established a new technique, high-resolution Bragg reflection method (1,2), which is useful for a simultaneous measurement of the velocity and absorption over the range 100MHz—1GHz. Using this technique, we have studied relaxation in CH2Cl2 and CH2Br2.

EXPERIMENTAL SYSTEM

The light scattered by the sound wave in the liquid is detected with an optical heterodyne system, in which frequency of the local oscillator light is shifted by 28MHz (fig.1). When the ultrasonic frequency is f, the beat signal at f+28MHz is detected and recorded as a function of the incident angle of the light (fig.2). The incident light is also detected as a beat signal at 28MHz. The angular width in the curve of the scattered light gives α, while the angle between the two peaks gives the Bragg angle, from which v is determined.

RESULTS

Measurements were made over 60MHz—700MHz with the accuracies of 0.5m/s for v and 2% for α. Hypersonic velocities were also obtained with Brillouin scattering. For both liquids, results were perfectly described with a single relaxation frequency (fig.3). The relaxation strength $\varepsilon_0 = (\nu_0^2 - \nu^2) / \nu_0^2$ was determined and compared with the theoretical value $\varepsilon_t$, calculated with the assumption (3,4) of vibrational relaxation associated with all but the lowest mode. They agreed within 1% (TABLE I). The dispersion curves for CH2Cl2 suggested that the lowest mode would relax at higher than 100Hz. The volume viscosity was estimated to be $\eta_v > 3n_0$, the shear viscosity.

REFERENCES


<table>
<thead>
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<th>TABLE I. Relaxation parameters at 20°C.</th>
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<td>CH2Cl2</td>
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<td>CH2Br2</td>
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INTRODUCTION

The ultrasonic method of measurements was used to determine shear mechanical impedance of trifluoropropylmethylsiloxane fluid.

THEORETICAL BASES

The theoretical curves were fitted to the experimental results with help of equation used by Barlow at al. (1) in form:

\[ 2KJ = J_0 \left( 1 + \frac{1}{\omega \tau_m} \right) + \frac{2KJ}{\omega \tau_m} \]

where \( J_0 \) is the limiting high frequency shear compliance, \( \omega \) - angular frequency, \( \tau_m \) - Maxwell relaxation time and \( K, \beta \) - empirical constants.

EXPERIMENTAL TECHNIQUES

The method of shear mechanical impedance measurements was given in previous paper (2). The measurements were made at frequencies of 10, 30 and 450 MHz in temperature range -100 to 50 °C. Four samples of fluorosiloxane fluid were used to measurements, the molecular weight of which were over the range 1720 to 73 600. In one case the mixture of two samples was measured.

RESULTS AND CONCLUSIONS

The viscoelastic relaxation region was determined for all samples. The values of shear mechanical impedance are given as a function of frequency.

The results of measurements can be represented with help of equation /1/ by matching the values of the empirical constants \( K \) and \( \beta \). It was found that the values of \( K \) are related to the viscosities of samples by the equation \( \log 2K = \alpha \eta^{n} \), where \( \alpha = 1/3 \), \( \eta \) - viscosity in cP, and \( \beta \) is near to 0.5. The same relation with different factor \( a \) was found for other polymer liquids, polybutenes (3) and polyacrylates (4). It is shown on Fig. 1. The points D and E2 are out of line probably because of entanglements.

REFERENCES

The two separate binodal curves of the ternary liquid system Methanol-Isooctane-Nitrobenzene for some temperatures are shown in Fig. 1 (1). The dashed line connects the plait points and the "col" is the common one of the two binodal curves where they meet. At this point (C), the critical temperature has a minimum as a function of two independent compositions. We performed ultrasonic absorption and velocity measurements along the plait point line as a function of temperature and frequency.

In the last years the ultrasonic propagation has been extensively studied in the critical region of both one and two component fluid systems, while a few data exist on ternary mixtures and are limited to a plait point of definite composition. Theoretical (2) and experimental (3) findings show a smaller and smaller influence of critical fluctuations as the number of components increases.

In this work we attempt to understand the behavior of sound propagation close to plait points at different temperature and compositions along the critical line extending between the critical points (L and K in Fig. 1) of the binary mixtures. In particular we are interested in the behavior at the "col".

Preliminary results show that the critical absorption is in fact lower than in the binary systems, and the same is true for the sound velocity dispersion, in agreement with theoretical predictions.

REFERENCES
INTRODUCTION

Ultrasonic studies have been made in some organic liquids and their mixtures in order to prove a relationship between the potential depth and sound velocity.

THEORETICAL BASES

In a previous paper (1) the following relationship between the potential depth $\Phi_0$ and the sound velocity $v$ has been reported

$$\Phi_0 = \frac{1}{n} \left( \frac{M}{T} v^2 - TR \right) \quad (1)$$

where $M$ is the molecular weight, $T$ is the absolute temperature, $R$ is the universal gas constant, $\gamma$ is the ratio of specific heats and $n$, $m$ are the exponents of the Lennard-Jones potential.

In ideal liquids and mixtures it has been used a Lennard-Jones (6-12) potential. In non-ideal liquids and mixtures (2) a modified Lennard-Jones potential was taken into account.

EXPERIMENTAL TECHNIQUES

Ultrasonic velocity has been measured as a function of temperature for a lot of organic liquids and mixtures, using an ultrasonic interferometer.

RESULTS AND CONCLUSIONS

The potential depths evaluated from eq (1) are as expected, proving the validity of the theory.

It has been observed a decreasing of the potential depth with increasing of the temperature. In ideal mixtures, the values of $\Phi_0$ were situated between those of the pure components. In non-ideal mixtures $\Phi_0$ might be higher than the values of the pure components, showing the existence of the molecular complexes.

REFERENCES

(2) S.A. Mihailenko, et al., Physics of Liquids (USSR),(1974) 2, 3.
AN ACOUSTIC METHOD OF DETERMINING THE DISPERSIVE COMPOSITION OF ORGANIC SUSPENSIONS

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1. The methods of determining the degree of dispersion and homogenisation of organic suspensions which have been applied so far in industry are extremely time-consuming and therefore not adequate for controlling technological processes. The present paper tries to prove that the dispersive composition at any given moment can be quickly determined by means of ultrasonic methods, in which an ultrasonic plane wave is used to bring about a fast sedimentation of the dispersive phase, whereas an ultrasonic shearing wave is used to determine the course of changes in the concentration of the dispersive phase during sedimentation in an ultrasonic field.

2. Basing on the accepted assumptions concerning the dispersive and dissipative phase, as well as on King's theories concerning the acoustic pressure upon a sphere /1/, a formula has been developed for the distribution of the concentration of particles dispersed in a polydispersive system during their acoustic sedimentation. Moreover, the relations has been investigated which exists between the damping coefficient of the ultrasonic resonator of shearing waves /2/ and the change in the volumetric concentration of the dispersed particles. Applying these relations, it has been possible to determine the changes in the concentration of particles in the dispersive phase during their acoustic sedimentation - expressed by the change of the damping coefficient of the resonator - in such a form that would be a full analogue of this expression represented as the sedimentation curve in the analysis of gravitational sedimentation. Thus it was shown that in the analysis of acoustical sedimentation we continue in the same way as in the analysis of gravitational sedimentation after passing over from the sedimentation curve to the curve of distribution.

3. The results of experimental investigations comprising measurements of the degree of dispersion and the degree of homogenisation of some organic suspensions prove that the described method makes it possible to reduce the analysis of dispersion to eight minutes, which meets the conditions prevailing in industrial production processes as well as the express methods of control.

REFERENCES
The velocity and absorption of the ultrasound and their temperature dependences were measured in aqueous solutions of a number of globular and fibrillar proteins by means of the fixed path ultrasonic interferometer. The ratio of the protein apparent specific adiabatic compressibility \( \Phi_k \) to the adiabatic compressibility of the solvent \( \beta_0 \) was calculated from the data of the measuring the ultrasound velocity. The value of \( \Phi_k/\beta_0 \) is shown to be an adequate characteristic of the protein tertiary structure in aqueous solution (see Fig. 1). The position of a protein on the \( \Phi_k/\beta_0 \) scale allows one to determine its general conformation and makes it possible to study the nature and kinetics of the protein conformational transitions by means of ultrasound velocity measurements.

The value of \( \Phi_k/\beta_0 \) and its temperature coefficient are determined to a first approximation by the ratio of the protein surface available for the solvent to the protein volume and also depend upon the character of the external atomic groups of the macromolecule.

The measurements of the ultrasound velocity and its temperature dependence in the solutions of amino acids and some model substances allowed one to find the quantitative relation between acoustic parameters of a protein solution and the hydration of the definite side groups of the polypeptide chain, made possible the quantitative estimation of the protein globula compressibility.

The results obtained enable one to develop a new approach to the ultrasonic investigation of biopolymers.

![Fig. 1 The scale of \( \Phi_k/\beta_0 \) for the proteins](image-url)
INTRODUCTION

The piezoelectric effect in cholesteric liquid crystals was reported in which the application of shear vibration to a cholesteric liquid crystal layer caused the alternating electric potential of the same frequency as the excitation[1]. Examination is repeated for a smectic liquid crystal, for which lecithin (egg) with distilled water (a principal ingredient of biomembranes) is tested.

EXPERIMENTS

The experimental arrangement is the same as in our previous work. The liquid crystal layer is sandwiched between a pair of glass plates with nesa-coated electrodes. The layer thickness is 100 µm and the area 1X1 cm². One of the plates is fixed while the other is connected to the vibrator of 25 kHz. Its lateral vibration develops a shear vibration in the layer. The mixture ratio of the lecithin and the water is 4:1, which is a lyotropic liquid crystal above 0°C.

RESULTS

Fig.1 shows a temperature characteristic of the DC resistance, in which there is a transition around 45°C, while for the cholesteric liquid crystal layers the resistance simply decreases with temperature. The order of the magnitude is 10³Ω while it is 10⁻⁶Ω in the cholesteric liquid crystals. The electric potential generated for the vibratory displacement 2 µm is measured at various temperatures, which is shown in Fig.2. Fig.3 shows the potential for various excitation levels. The surfaces of the glass plates are treated for the molecular orientation, so that static shear deformation or DC electric voltage (3 V, 1 hour) is initially applied. Their effect disappears a few minutes after the vibratory excitation is applied.

REMARKS

The electric potential is very small in comparison with the cholesteric liquid crystals, and its magnitude is only one-tenth to one-hundredth [1][2]. No electric generation is observed for the nematic liquid crystals (MSBA and EBBA). The piezoelectric effect thus observed possibly relates to the mechanism associated with the bielectric generation in a receptor for mechanical stimulations.

REFERENCES

**Shear Wave Measurements in Nematic and Smectic Phases**

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Using a shear wave reflectance technique we studied some anisotropic friction coefficients in nematic and smectic-A phases.

1) Nematic with a first-order transition at lower temperatures

We measured at 15 MHz the real part $R$ and the imaginary part $X$ of the shear impedance for the nematic liquid crystal p-n-pentyl p'-cyanobiphenyl (PCB). These two quantities are found to be equal within our experimental errors. From these measurements we deduced the viscosities $\eta_A$, $\eta_B$ and $\eta_C$ (see fig.1) and we verified the Rapini equality $\eta_B = \eta_C$. (see fig.1: Schematic diagram of the experimental set-up showing the three orientations in which viscosity coefficients were measured.

2) Smectic-A phases

We studied two smectic-A phases: di-octylazoxybenzene and di-heptylazoxybenzene, the latter of which presents a nearly second-order nematic to smectic-A phase transition.

When the shear wave propagation is along the director (orientation C of the fig.1) we observed that $R$ and $X$ differ significantly indicating a relaxation process.

One can understand this behavior if we assume that the director is disturbed from equilibrium by the ultrasonic stress and then relaxes in a finite time $\tau$. We analyse our results on this basis.
INTRODUCTION

The smectic-A phase is a stratified medium where the molecules are arranged in two-dimensional liquid layers. When the nematic to smectic-A phase transition is of second order, some properties of the smectic phase appear on the nematic side of the transition. One can understand the origin of this behavior within the simple picture of smectic droplets in the nematic phase. These smectic droplets have a longitudinal dimension $\xi_l$ and a transverse dimension $\xi_\perp$. It has been proposed that the two correlation lengths $\xi_l$ and $\xi_\perp$ vary as:

$$\xi(I) = \xi_0 \left( \frac{T - T_c}{T_c} \right)^{-\nu}$$

with $\nu = 0.5$ in a mean field approach [1] and $\nu = 2/3$ from the helium-analogy [2]. The ultrasonic absorption is related to the correlation length and is expected to diverge [3]. This correlation length is defined from a development of the free energy in powers of the order parameter $\psi = \psi_0 e^{i\phi}$ where the amplitude $\psi_0$ specifies the density of the layers and $\phi$ whose the phase $\phi$ specifies the position of the layers.

The dynamic behavior of the order parameter $\psi_0$ is assumed to follow a simple relaxation law:

$$\frac{d\psi_0}{dt} + \frac{1}{\tau} \psi_0 = 0$$

The relaxation time $\tau$ has been discussed by Brochard [4].

RESULTS

The ultrasonic absorption has been studied in magnetically aligned cyanobenzilidine-octyloxyaniline (CBOOA) which presents a quasi second-order nematic to smectic A phase transition. The frequency dependence of the absorption indicates the presence of a critical relaxation process near the A-N transition with a relaxation frequency in the low-MHz range. The volume and shear viscosities and the relaxation time of the order parameter are deduced from these measurements. Their temperature dependence is compared to the theoretical predictions. On the smectic side of the transition another relaxational effect is observed which is presumably related to the non-hydrodynamic relaxation of the director.

REFERENCES

IV. Physical acoustics

N. SOUND PROPAGATION
Dans le cadre de la lutte contre le bruit et pour la protection de l'environnement, ELECTRICITÉ DE FRANCE a entrepris une étude théorique et expérimentale de l'influence des conditions météorologiques sur la propagation du son au voisinage des Centrales.

De nombreuses études ont déjà été entreprises sur ce problème, mais, à notre connaissance, aucune n'a débouché sur une méthode de planification de l'environnement utilisable industriellement. Le but de notre étude est de tenter de mettre au point une telle méthode.

L'étude théorique est basée sur l'approximation géométrique : on écrit les équations différentielles des rayons sonores, considérés comme les courbes caractéristiques du système différentiel fondamental des équations de l'hydrodynamique. L'atmosphère est supposée stratifiée horizontalement, caractérisée par la température absolue, l'humidité relative, et les deux composantes horizontales du vent, mesurées à différentes altitudes et moyennées sur une durée de 10 minutes ; une interpolation cubique par morceaux permet d'obtenir des fonctions de classe $C^2$ nécessaires pour les calculs d'intensité sonore.

On considère aussi l'approximation géométrique comme terme d'ordre zéro dans le développement de la solution générale en série entière de $1/w$ (w, pulsation en régime sinusoidal) (cf. Blokhintsev (1)). Cette seconde méthode ne conduit pas aussi directement que la première aux équations différentielles des rayons, mais elle donne des renseignements complémentaires d'ordre thermodynamique et énergétique qui permettent, en particulier, d'évaluer le niveau sonore à distance. Par rapport au cas de la propagation en atmosphère réelle, en ne tient pas compte de la turbulence atmosphérique et le sol est simplement considéré comme une surface plane réfléchissante non diffusante.

Le fait que l'on néglige la turbulence implique que ce modèle ne peut prévoir que des niveaux sonores moyens (moyennés sur 10 minutes). Il faut cependant savoir que l'on peut étudier les fluctuations du niveau sonore dues à la turbulence à l'aide de la théorie géométrique (cf. CERNOV (2)). Pour l'instant, cette étude se limite à la prédiction du niveau sonore moyen. Un programme de calcul sur ordinateur permet de calculer les rayons sonores issus d'une source ponctuelle et passant par un point de réception donné : le long de chacun de ces rayons, on calcule l'atténuation atmosphérique et la divergence géométrique, et on additionne de façon incohérente les contributions de tous les rayons calculés, ce qui donne le niveau sonore moyen au point considéré.

L'étude expérimentale consiste à émettre des bandes de bruit de largeur 1/3 d'octave et à mesurer le niveau reçu en un certain nombre de points à des distances variant de 600 m à 1500 m. Le site retenu pour ces expériences de validation du calcul est un plateau très peu vallonné (près de PARIS), le site est équipé d'une station météorologique comportant un mât de 100 m de haut ; de plus, il peut y être procédé à des lâchers de ballons radiosondes.

Les premiers résultats seront exposés et discutés lors de la communication, et quelques dispositifs montreront les diagrammes des rayons obtenus dans des cas particuliers.

REFERENCES : (1) Blokhintsev : The propagation of Sound in an inhomogeneous and moving medium — JASA 46, Vol. 18 n° 2. (2) CERNOV : Wave propagation in a random medium — DOVER PUBLICATIONS.


ÜBER DEN EINFLUSS DER WETTERBEDINGUNGEN AUF DIE
SCHALLAUSBREITUNG IM FREIEN

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Bei Freifeldmessungen, in Entfernungen von einigen 100 m oder mehr
von einer Schallquelle, z.B. von einer Industrieanlage oder Autobahn,
beobachtet man je nach den jeweils herrschenden meteorologischen
Bedingungen eine erhebliche Streuung der gemessenen Schallpegelwerte. Dies kann zu einem größeren Zeitaufwand für die Messung
führen, wenn Unsicherheiten bei der Beurteilung von Geräusch-
immissionen vermieden werden sollen.

Um die meteorologischen Einflüsse besser beurteilen zu können,
werden Temperaturprofile bei schwachwindigen Wetterlagen gemessen
und auf der Grundlage der geometrischen Strahlgleichung
\[ \frac{d (n \lambda)}{ds} = \text{grad } n \] das Schallstrahlenfeld für Punktschallquellen
berechnet (s. Bild 1). Die Pegelabnahme mit der Entfernung,
Schattenbildung und der Einfluss des Bodens auf die Schallausbreitung
wurden untersucht. Für den Vergleich der Rechenergebnisse
mit Messwerten stand eine Meßkette mit 4 Mikrofonen zur Verfügung,
mit der die Pegelabnahme über eine Entfernung bis zu 1000 m
gemessen werden konnte.

Bild 1:
Temperaturprofil und Schallstrahlenfeld; innerhalb des Öffnungswinkels \( \alpha \) erfährt die Schallstrahlen Totalreflexion.
SIMPLE ECHOSONDES

The physical interactions occurring when the earth's atmosphere degrades the outdoor propagation of sound may, conversely, provide powerful methods for the remote sensing of atmospheric structure. Perhaps the most useful actualization of such methods appears in the echosonde (or acoustic sounder), an instrument completely dependent on the changes in sound propagation produced by turbulence. Since the invention of the echosonde by McAllister in 1968 and the recognition by Little in 1969 that its underlying physical principle consisted of turbulent scattering of sound, the numbers and design sophistication of echosondes in the world have increased almost explosively. During the talk, slides will show typical configurations and operations of so-called "monostatic" and "bistatic" echosondes, as well as a figure giving known locations of such instruments in various countries. In its simplest form the monostatic echosonde exploits the temperature fluctuations associated with turbulence in a stratified atmosphere to provide a picture of the structure of the boundary layer, to distinguish between stable and unstable conditions, and to provide measurements of inversion heights. Thus, this instrument is especially useful for investigations of atmospheric pollution and also can serve to advantage in studies of "noise pollution."

RECENT DEVELOPMENTS

The kinds of quantitative information that the simplest forms of echosondes can produce, however, remain quite limited. Many applications require absolute measurements of a greater number of atmospheric parameters. As an example, Brown has suggested schemes by which a combined monostatic-bistatic echosonde furnishing values of the turbulence intensity parameter $C_{12}$ and $C_{v3}$ (for temperature fluctuations and velocity fluctuations, respectively) can provide, in turn, the critical parameters needed by boundary layer meteorologists. But, achieving accuracy in such measurements appears less straightforward than once thought and requires careful use of correction factors such as that accounting for so-called "excess attenuation." Probably the greatest development in the last two years has gone into Doppler echosondes designed for the measurement of atmospheric winds and wind shears. These instruments are rapidly becoming extremely simple to operate and achieving accuracies within less than a meter per second. Finally, new impetus has recently been given to the development of other acoustic and hybrid remote sensing configurations, especially those that can provide measurements of absolute atmospheric temperatures and temperature profiles. Among these, both the phase-sensitive echosonde interferometer and the combined radio-acoustic system (RASS) appear to have some promise.
INTRODUCTION

A sodar is a pulsed acoustic radar operating at a fixed acoustic carrier frequency. This system, limited both in height and time resolution, is affected by spurious echoes from nearby obstacles, in a manner similar to the ground clutter affecting the electromagnetic radars, and makes use of large acoustic power to obtain a good S/N ratio. Therefore it has been taught convenient to test a new system based on the same principles adopted by the electromagnetic FM-CW radar. A simplified block diagram is shown in the figure.

THE SYSTEM

An acoustic carrier frequency of 2000 Hz generated by the voltage controlled generator (V.C.G.) is frequency modulated by the saw-tooth generator (S.G.) and radiated by the acoustic antenna T. The length of the saw-tooth is of the order of 2 sec followed by a 2 sec gap; the bandwidth of the frequency modulation is ±200 Hz. In this way the equivalent bandwidth obtainable in theory in reception is 0.5 Hz, and the height resolution is of the order of 1 m. Of course, various physical factors like the bandwidth spread due to turbulence, may make it impossible to reach the theoretical limits. The signal backscattered by the turbulence and received by R goes through a band-reject filter (B.R.F.) and through band-pass amplifier. Then it is multiplied with the transmitting signal in order to obtain a low frequency conversion to be analyzed by a real time analyzer (R.T.A.). A second detection system is obtained with the use of a digital computer analyzing the analogic signal recorded in the magnetic tape M.

CONCLUSIONS

The system described above is being built and tested to check the possibilities of this FM-CW sodar in various conditions.
NUMERICAL ANALYSIS OF SODAR SIGNALS: PHASE AND AMPLITUDE COHERENCE

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INTRODUCTION

The acousting sounding of the atmosphere is now widely used to detect the turbulence and wind in the boundary layer. The instrument used (SODAR) is an acoustic pulsed "radar" operating, in our case, at a frequency of 1000 Hz. The backscattered signal is modulated in amplitude and phase with a very limited bandwidth, and it may be represented in a synthetic way as \( a(t) \cos(\omega_0 t + \phi(t)) \), where \( \omega_0 \) is \( 2\pi \cdot 1000 \text{ sec}^{-1} \).

SYSTEM DESCRIPTION

We are in the process of building an acoustic sounder numerically controlled by a microcomputer. It is therefore important to extract the useful information from the signal, that is the slowly varying functions \( a(t) \) and \( \phi(t) \), in real time and with limited memory size. This goal may be accomplished in various ways. In order to check the most convenient procedure, a minicomputer is used to simulate the real situation; as a result, the useful part of the received signal is stored as a time series of complex numbers of the form \( a(t)\exp(j\phi(t)) \). The figure shows absolute value and phase of \( \sum a_n \exp(j\psi_n) \) as a function of height; the pulse repetition period was 5 sec.

EXPERIMENTAL TECHNIQUES

The procedure described above has been applied to the analysis of signals backscattered by atmospheric layers, received by 4 acoustic antennas distributed at various distances from each other. The aim of the experiment was to study the statistical properties of the acoustic field at the ground. In particular, the phase coherence in time of the signal received from the same layer together with phase scintillation of the signals received by the various antennas have been studied.

CONCLUSIONS

The different time and space behaviour of \( a(t) \) and \( \phi(t) \) for various atmospheric layers seems to be a useful means to characterize different atmospheric structures. Extensive studies along these lines are necessary.
INTRODUCTION

A knowledge of the characteristics of noise propagation outdoors is very important in the study of community noise problems, therefore, a lot of data should be obtained by field measurements.

As is well known, however, noise propagation, being influenced by weather conditions and absorption by ground cover, is so complicated a phenomenon, that it is rather difficult to distinguish each individual factor in field measurements.

We are, therefore, investigating this problem by use of scale model technique for aid in field measurements. In this paper, some results of a model experiment concerning the effect of wind and ground absorption on noise propagation are shown.

EXPERIMENTAL

A 1/100 scale model experiment was performed in a wind tunnel facility (boundary layer type, 1.8m x 1.2m in section). In this experiment, the mean wind velocity was set at 3.3m/sec. at the height of 2cm (corresponding to 2m in real) above the ground (glass fiber board, 15mm thick and 96kg/m² dense was used), and the vertical wind gradient was adjusted so as to approximately fit the 1/4 power law.

RESULTS

Fig.-1 shows the effect of wind on S.P.L. distribution above the absorptive ground, when an omnidirective sound source (a jet noise was used) was set 1cm (1m in real) above the ground. S.P.L. decrease in distance from the source, measured at the height of 1.5cm (1.5m in real), are shown in Fig.-2 and 3 for the frequency ranges of 50kHz and 100kHz (500Hz and 1kHz in real) in 1 oct. band. In these figures, the results of a field test, measured on open level grass land at an airport under wind velocity condition of 2.2m/sec. at the height of 3m above the ground, are shown together with the results of the model study. It can be seen that the experimental results in the model study are approximately similar to those of the field tests in respect to the excess attenuation due to wind effect and ground cover.
INTRODUCTION

Traffic noise has been recognized for over 40 years as a primary source of disturbance in cities. Until recently however prediction methods and discussions of errors in noise sampling have related mainly to the relatively simple problem of vehicles travelling steadily and freely in open areas. The importance to sampling of accelerating or decelerating traffic, distance of measurement and parameter selection will be commented upon briefly. Then Rathe's method (1) of determining $L_{eq}$ will be developed to incorporate gross attenuation factors created by atmospheric absorption and building shielding. The shielding phenomenon will be investigated in detail for a row of regularly shaped buildings to better understand the effect of building parameters.

THEORETICAL BASES

$L_{eq}$ may be determined theoretically from the abbreviated equation:

$$L_{eq} = 10 \log_{10} \sum_{j} \rho_{ij} S_{ij}$$

where $S_{ij}$ is the angle subtended at a fixed point by the ith road segment. $\rho_{ij}$ depends upon distance, attenuation and the relative frequency, speed and sound generating capability of the jth vehicle class travelling the ith segment. Computations may be reduced if, as is often the case with major throughfares in built-up locales, sound from the closest road dominates. Then shielding by buildings may be evaluated from equation (1) using geometrical acoustics to calculate the contributions to $S_{ij}$ of both the direct and reflected sound fields.

EXPERIMENTAL TECHNIQUES

Traffic noise was measured at representative locations in the City of Winnipeg. Experimental data concerning shielding was obtained from the literature (2). The effect of sampling was assessed for different traffic noise indices using an analog/digital computer.

CONCLUSIONS

1) Sampling becomes more critical as the intrusiveness of the traffic noise increases and as an index places more emphasis on the intrusiveness.
2) Gross $L_{eq}$ predictions based on four vehicle classes are within 3 dB of measured values and, hence, are useful for planning purposes.
3) Local sound fields are influenced greatly by shielding which depends critically upon the spacing, width/breadth ratio and absorption of buildings. Theory agrees reasonably with published measurements, diffraction probably being the major cause of discrepancies.

REFERENCES

(2) M.E. Delany et al., National Physical Laboratory Report AC 54 (1971).
Recent developments in understanding and predicting sound transmission in urban and suburban areas are reviewed. These developments are in four main areas: theoretical (analytical) models, computer simulation, physical scale models, and nomographs or design charts.

THEORETICAL DEVELOPMENTS

Theoretical developments have proceeded along two lines -- channel propagation (along streets) and two dimensional diffusion models. In its simplest form, channel propagation considers propagation between parallel planes, with losses at the vertical boundaries (building facades) simulated by an absorption coefficient. In order to predict the near-source reverberant build-up and noise levels in intersecting streets, one must include diffusion of sound due to building surface roughness. This has been done by assuming a room-like diffuse field and also assuming multiple specular reflections combined with a single diffuse reflection. Propagation in relatively open spaces with many scatterers can be considered a problem in diffusion of sound.

COMPUTER SIMULATION

Computer models for sound propagation take two forms. In the first, formulas developed from theory or other procedures (including computer simulation) form the basis for a sequence of calculations that predict sound levels. Several such models for predicting traffic noise are in current use. In the second form, the computer simulates the experimental situation by following packets of sound energy as they propagate through the space, reflect, and are absorbed. Such a procedure can be used to predict sound transmission in streets or in spaces of reasonably complicated shape.

PHYSICAL SCALE MODELING

Scale modeling has traditionally been used for indoor sound propagation studies. Since the early 70's, however, this technique has been applied increasingly to problems of outdoor sound propagation. Although it is still a technique under development, there are groups in most European countries and in North America applying it. Scale modeling has been applied to prediction of noise transmission in city streets and urban complexes. It has also been used in several site evaluation studies to predict the effect of barriers, building layout, or topographical changes on noise levels. Instrumentation for the generation and processing of high frequency sound pulses used in scale modeling has been developed and is being marketed.

GRAPHICAL METHODS

Graphics and nomographs are the conventional methods of presenting noise prediction techniques for application. The Swedish and U.S. procedures for predicting highway noise are basically graphical, although the latter has also been computerized. These nomographs may be based on theoretical calculations, field data, computer simulation, or some combination of these procedures. With some practice, they can be used quickly to predict sound levels in a variety of situations. Recently, Donovan has developed a set of nomographs that allow one to predict noise levels in situations in which sound travels along a network of streets, but do not include sound that propagates over the roofs or in between buildings to get from one street to another.
INTRODUCTION

Because the large population, 4.3 millions, lives mainly within the few narrow coastal plains, the demand for housing in Hong Kong is extremely high. Because of this big demand, tall buildings have been put up incessantly for the last two decades. This incessant building activities usually resulted in the pollution of the surrounding areas, which accommodated thousands of people, by different types of construction noises, notably the piling noise, jackhammer noise, air compressor noise, and others. The presence of these tall buildings renders different propagation characteristics of the noise thus generated from the one in the open environment.

EXPERIMENTS

The sites are classified into three categories of closed, semi-closed and open [1]. Closed environment consists of sites which have tall buildings on more than two sides of the road. Open environment has no building around it. Semi-closed is the ones in between.

The present investigation concerns the propagation of the sound produced by one type of the construction machinery, the jack hammer. The octave sound pressure spectrum obtained very near the jack hammer has near constant spectral level from 60 Hz to 80 kHz.

The propagation of the jack hammer noise in some of the closed and semi-closed environments in Hong Kong is shown in Figure 1. Also shown in Figure 1 is the attenuation of 6 dB per doubling distance. The best fitted attenuation curves obtained at the closed and semi-closed environments are definitely having lower attenuation with distance than the -6dB one. This means that higher noise level is experienced in these sites. The difference between them is small when it is very near the source and increasingly larger at larger distance. At 100 m the semi-closed environment would have a higher noise level, by about 4 dB(A), than the one of -6dB per doubling distance. At the same distance the closed environment would be 8 dB(A) higher.

The higher sound pressure level in the closed environment observed in Figure 1 agrees fairly well with the results of the aircraft noise [1].

REFERENCES

(1) N.W.M. Ko, J.S.V. (1975), 38, 512.
Experimentelle Ergebnisse zum Schalldurchgang bei durchströmten Düsen werden beschrieben. An runden Düsen von konischer Form wird bei hoher Unterschall-Austrittsgeschwindigkeit ($M = 0.3 - 0.7$) die Endimpedanz bestimmt. Dazu werden Schalldruckmessungen im Luftzuführungsrohr zur Düse ausgeführt, aus denen sich mit einem numerischen Verfahren für die Schallausbreitung durch die Düse mit Strömung die Endimpedanzen berechnen lassen. Parallel dazu werden Fernfeldmessungen des abgestrahlten Schalls im reflexionsarmen Raum ausgeführt. Für ebene Schallwellen in der Düse lassen sich so z.B. Energiebilanzenmessungen vor der Düse und im Fernfeld ausführen. Das Fernfeld wird durch die Anwesenheit der Strömung völlig verändert: Bei niedriger Frequenz tritt im Gegensatz zum Fall ohne Strömung eine starke Richtwirkung auf. So ist z.B. bei $M = 0.7$ und $Sp = 0.1$ die Schallabstrahlung in Strahlrichtung um 30 dB höher, als in entgegengesetzter Richtung.

Ein eindrucksvoller Effekt ergibt sich bei höherer Amplitude (> 130 dB) der Schallanregung mit reinen Tönen: Der breitbandige Strahlärm wird um 7 dB und mehr verstärkt. Besonders bei Schallanregung mit höheren Moden und nicht zu hoher Frequenz ist dagegen die akustische Tonanregung im Fernfeld kaum noch zu identifizieren (s. Bild 1: Mode 1, umlaufend /1, 2/). An diesen Verstärkungseffekt ist die Wellenbewegung beteiligt, die sich durch die Schallanregung im Strahl ausbildet. Experimente mit einem einfachen Schalldämpfer zeigen /1, 2/, daß die Strahlärmverstärkung auch unterdrückt werden kann. Demnach kommt die einfache Vorstellung einer Streuung von Schall an Turbulenz nicht als ausreichende Erklärung in Frage; dagegen spricht auch der nichtlineare Charakter des Effekts /1, 2/.

Bild 1: Fernfeldmessung 45° zur Strahlachse, $M = 0.5$, Bandbreite des Filters $Sp = 0.005$, Kurve a: mit Schallanregung bei $Sp = 0.3$, Kurve b: ohne Schallanregung.

Literatur:

INTRODUCTION

To determine the importance of various transmission paths through a complex structure (ship, building etc.), it is necessary to measure airborne and structure-borne sound intensity at many different points of the structure. This can be done successfully in practice by an intensity meter which is able to perform on-the-spot measurements with sufficient accuracy, and which is yet simple enough to operate with.

INTENSITY DETERMINATION

Measurements of both airborne and structure-borne wave intensities were performed successfully (e.g. (1), (2), (3)) employing various methods. It can be shown ((3), Pavić: Measurement of sound intensity - to be published) by applying finite difference technique that an equation for the net sound wave intensity assumes the following form:

\[
I(t) = C \left< \frac{\xi_1(t) \xi_2(t)}{t} \right> \quad \text{t-time, } C\text{-constant}
\]

where \(< >\), denoting time averaging, contains variables \(\xi_1\) and \(\xi_2\) representing some measurable quantities of the corresponding wave-field (for acoustical and longitudinal structural waves) or their linear combination (for flexural structural waves). Therefore, the intensity can be calculated on the basis of measurements by applying the characteristic operations (integration, multiplication and averaging) on the signals from the measuring equipment.

ACCURACY AND CONCLUSIONS

An obvious way to calculate various intensities according to the formula shown above is to design an analog electronic device which does the necessary computations. However, the requirements imposed to such a device which follow from the corresponding error analysis (3) make it unsuitable for use.

Following this specific requirements, a portable hybrid computing unit has been designed for the sound intensity determination, which uses analog signals from measuring transducers as inputs. While filtering and summation/differentiation (where applicable) operations are performed in an analog way, the integration, multiplication and averaging operations are done digitally. The upper frequency limit is 20 kHz and the possibility exists for digital DC signal rejection, applicable to either of the inputs. The current result of the computation is shown every 2 secs on a digital display. Besides its normal function, the device can be used as an RMS detector or a heterodyne filter.

REFERENCES

(3) G. Pavić, Doctoral Thesis (1976), University of Southampton
INTRODUCTION.

The acoustic energy produced by a noise reaches a receiver in different ways: as air-borne waves/direct wave, reflected or diffracted waves/ and structure-borne waves. To decrease the noise level reaching a receiver one should know the quantity of acoustic energy propagated in particular frequency bands through the each sound path and should limit the propagation in the predominant paths.

METHODS.

The transmission of the energy from the source to the receiver may be expressed by the sum of transmittances representing the particular paths. For i-th path we have/with dispersion/:

\[ G(\omega, T_i) = \frac{\int_{t_1}^{t_2} f_i(t) \exp \left( -j \omega (T_i(\omega) + \varphi_i(\omega)) \right) \exp(-j\omega t) \, dt}{\int_{t_1}^{t_2} f_i(t) \exp(-j\omega t) \, dt} \]

where \( f_i(t) = 0 \) for \( t < t_1 \) and \( t > t_2 \), \( f_i(t) = 1 \) for \( t_1 \leq t \leq t_2 \);

\( T_i(\omega) \) is a delay of sound propagation by i-th path, \( \varphi_i(\omega) \) phase shift of the impedance of i-th path.

The appearance of the parameter \( T_i \) twice in the above expression points to the possibility of using 2 methods for identification: a correlation one, using the phase shift \( \omega T_i \) and pulse method using the duration of sound \( t_2 - t_1 \) and delay \( T_1 \).

EXPERIMENTAL

Experiments were done by means of correlation and pulse techniques on the example of a barrier placed in enclosure. As a result, a comparative analysis of both methods was performed.

CONCLUSIONS.

It has been shown that apart from the common limitation resulting from the Heisenberg's principle \( \Delta f \Delta t > 1 \) both methods have their own additional limitations. The analysis of the above limitations allow to draw practical conclusions as to the choice of a more adequate method in given conditions.

REFERENCES.

Schomer, JASA Vol. 51 1972 nr. 4
J. Barger, Noise Control Engineering Vol. 6 1976
SOUND PROPAGATION AND GENERATION IN FLOW DUCTS WITH AXIAL TEMPERATURE GRADIENTS

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INTRODUCTION

In many duct acoustics situations (for example, automobile silencers) involving mean axial flow and temperature gradients, the fundamental acoustic mode dominates the sound field. This paper presents a linearised analysis of the problem (neglecting the effects of shear flow), including an approximate analytical, high-frequency solution.

THEORY

Phillips' equation, for one-dimensional acoustic propagation in a hard-walled duct where the mean static pressure is constant along the axis, reduces to

\[
\frac{\partial^2 p}{\partial x^2} - \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} - i \frac{\partial D}{\partial t} = -\rho_0 \frac{DQ}{Dt},
\]

where a source term of volume strength \(Q\) per unit volume, has been included; \(c\) is the local speed of sound.

Using the WKB method, eqn. (1) may be solved approximately (for sinusoidal time variation) in the absence of sources, to yield

\[
p(x,t) \approx e^{\text{int}} \left[ p_e e^{-\left(\frac{\omega k}{c^2} x - \chi\right)} + \sum_{n=0}^{\infty} \frac{\omega k}{c^2} x - \chi\right];
\]

this equation is valid at sufficiently high frequencies.

If one takes the space-time Fourier transform of eqn. (1), a solution to the sound generation problem may be found for prescribed source conditions by integration along an appropriate contour. For example, for a side-branch source at \(x=x_0\) in an infinite duct, the positive-going wave solution is given by

\[
p^+(x,t) = \omega \rho_0 \sum_{n=0}^{\infty} \frac{\omega k}{c^2} e^{\text{int}} \left[ p_e e^{-\left(\frac{\omega k}{c^2} x - \chi\right)} + \sum_{n=0}^{\infty} \frac{\omega k}{c^2} x - \chi\right] A \chi (1-M_n) (k_+ - k_-);
\]

where \(q(x)\) is the total volume velocity amplitude in the side-branch where it joins the duct. The subscript and superscript 's' denotes conditions at the source; \(k_+\) and \(k_-\) are the axial 'wavenumbers', which have real and imaginary parts, and \(A\) is the duct's cross-sectional area. The exponent \(n\) is equal to 2 for low flow speeds and 1 for higher flow speeds. The axial wavenumbers from eqn. (2) may be used in eqn. (3); the factor \(e^{\text{int}} \left[ p_e e^{-\left(\frac{\omega k}{c^2} x - \chi\right)} + \sum_{n=0}^{\infty} \frac{\omega k}{c^2} x - \chi\right]\) in (3) is equal to the \(x\)-dependent factor in the first term in eqn. (2).

CONCLUSIONS

One may solve eqn. (1) as above, or by a numerical method which would be accurate at low frequencies too. The above method is a useful approximation, which should be valid provided \(\omega c > \rho_0 g D p / \partial x\) at all points in the duct. The effect of mean flow on sound generation is illustrated by eqn. (3); precise source conditions determine the extent of the Mach number dependence of sound pressure. (At the time of submission of the abstract, experimental results were not available, but these should be ready by the date of the I.C.A.)
Obstructions in flow ducts like perforated plates, valves, changes of the cross-section, etc. largely influence the acoustic properties of the duct. The prediction of the sound transmission through the duct, or of the acoustical input impedance, especially in the case of superimposed flow, is not possible up to now, except by rather crude quasi-steady models which have been used to interpret experimental data [1,2]. Although the basic requirement of these models, namely the quasi-steadiness is usually not fulfilled, the predictions are in fair agreement with the experimental data, in many cases.

This is demonstrated for the case of orifice plates. The calculation of the reflection coefficients for plane waves according to a quasi-steady model is reported in [1]. The theory was extended to the calculation of the transmission coefficients. Both, the reflection and the transmission coefficients depend on the pressure drop at the orifice plate and on the derivative of the pressure drop with respect to the Mach number. The pressure drop was assumed to originate from isentropic flow up to the narrowest cross-section and diverging jet flow further downstream. The narrowest cross-section of the flow is smaller than the cross-section of the orifice because of the contraction of the jet emanating from the orifice. Therefore, this narrowest cross-section is computed from the measured pressure drop according to the assumption mentioned above.

The reflection and the transmission coefficients of a circular orifice with a relative cross-section of 0.78 (compared to the cross-section of the duct) was computed and compared to experimental data. The narrowest cross-section was about 0.6 at low Mach numbers (up to 0.45) and increased slightly to 0.64 at the highest Mach number (0.65) - the Mach number is based on the flow parameters in the narrowest cross-section. The experimental loss of acoustic energy at the orifice is in good agreement with the calculated loss at low Mach numbers and is somewhat higher than predicted, at high Mach numbers. An equivalent narrowest cross-section was computed also from the measured acoustic data, and it turned out that this "acoustical" narrowest cross-section remains constant (0.6) throughout the total range of Mach numbers investigated here. However, this single example is not conclusive and more data are needed to find out, in which cases, and eventually why the quasisteady models exhibit correct predictions.

Therefore, a new measuring technique is used, at present, to obtain rather precise data of the reflection and the transmission coefficients for both directions of sound propagation. First results will be reported.

REFERENCES

INTRODUCTION

The attenuation of a sound wave propagating through turbulent flow in ducts is caused by the excitation of a shear wave and a heat conduction wave at the wall. The shear wave propagates from the wall into the turbulent medium. Ronneberger and Ahrens have measured the wall impedance of such shear waves, $\frac{T}{\dot{u}}$, which determines the sound propagation at low Mach numbers ($T$ and $\dot{u}$ are the shear stress and the velocity in the shear wave, respectively). They showed that this impedance is strongly affected by the turbulence.

At present, two new experiments are conducted to get further information about the interaction between the shear wave and the turbulence near the wall. This information is needed for the calculation of the sound attenuation at high Mach numbers.

EXPERIMENTAL TECHNIQUES

In the first experiment, a shear wave is excited at the wall of an axially oscillating tube (inner diameter 4 cm) which carries a fully developed turbulent flow of water. The wall shear stress $T$ is determined by measuring the force which is exerted by the fluid on an axially movable part of the oscillating tube. This part of the tube (called inner tube) is driven electromagnetically by an outer tube which surrounds the inner tube and oscillates with the same amplitude and the same phase as the inner tube. The force can be evaluated from the electric currents. The velocity $\dot{u}$ of the oscillation is measured by the same way, namely by determining the acceleration force of a mass which is coupled electromagnetically to the oscillation of the outer tube.

In the second experiment the propagation of the shear wave into the turbulent medium will be studied within a thick turbulent boundary layer (oil) by hot film measurements. The shear wave is excited at the wall of a channel which carries a two dimensional turbulent flow. The wall is oscillating in streamwise direction. The Reynolds number is varied between $10^4$ and $10^5$, and the Strouhal number $Sv/\dot{u}_t^2$ (inner parameters) ranges from $10^{-4}$ to $10^{-1}$.

REFERENCE

Ronneberger et al., submitted to J. Fluid Mech. (1976)
The acoustic impedance of a perforated plate in front of the wall of a flow duct is given by the impedance of the cavity behind the perforate and by the impedance of the orifices. It was shown (ref. [1]-[3]) that the acoustic impedance of such orifices is strongly influenced by grazing flow. In most cases it is reported that the real part of the impedance of orifices or of porous linings increases with higher flow velocities, whereas the imaginary part decreases.

The orifice to be investigated is formed by a small tube (diameter $2a = 4$ mm) which leads to a horizontal plate, exposed to tangential flow. The acoustic reflection coefficient in the exit plane of the tube is measured as a function of the Strouhal number $S$, calculated with the flow velocity, the radius $a$ of the orifice and the sound frequency $\omega = 2\pi f$. In this way, the impedance of the end-correction of the orifice can be determined from the reflection coefficient.

Normalization of the measured impedance is carried out with the impedance of the end-correction with air at rest (i.e. $\approx 0.85 \rho c a$). At this experiment, the sound frequency was changed, whereas the grazing flow velocity was kept constant to obtain a fixed ratio of boundary layer thickness $\delta^*$ (displacement thickness) to orifice radius. The results are plotted as a function of the reciprocal Strouhal number $S^{-1}$ (see Fig. 1). For $S^{-1} > 8$ the straight lines for the real part of the impedance indicate that for high values of $S^{-1}$ the real part normalized to $\rho c$ approaches the dc flow resistance of the orifice with grazing flow [1]. Measurements carried out with the same orifice at different boundary layers above the orifice (laminar-turbulent, $0.13 < \delta^*/a < 1$) indicated that the ratio $\delta^*/a$ is the most important parameter. In particular the real part can be represented as a function of $\delta^*/a$ at small Strouhal numbers: the real part of the impedance decreases with increasing boundary layer thickness. This relation was confirmed by measurements of the dc flow resistance of the orifice. In the case that the length of the orifice becomes comparable to the radius, there arise additional changes of the impedance, probably due to the influence of flow fluctuations on the inner end of the orifice. The impedance is also changed e.g. by rounding off the upper edge of the orifice. The interaction among several orifices has been investigated with 2 and 9 neighbouring orifices.

**REFERENCES**

HIGHER ORDER ACOUSTIC MODES DUE TO INTERNAL FLOW DISTURBANCES IN RELATION TO ACOUSTIC RADIATION FROM PIPES

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INTRODUCTION

The work reported comprises experiments and analysis which show that higher order acoustic modes, generated by flow disturbances in pipes, can be responsible for large increases in vibrational response and acoustic radiation.

EXPERIMENTAL ARRANGEMENT

Effects of 45° and 90° mitre-bends have been measured in an intermittent induced-air-flow rig (centre-line flow speeds 65-170 m/s). Fully developed turbulent flow is established before the bend and re-established before a thin-walled (0.89 mm) steel test section mounted in an anechoic chamber 55 diameters downstream of the bend. Test section measurements included spectra of wall pressure, wall acceleration and acoustic radiation.

RESULTS AND CONCLUSIONS

The wall pressure spectral levels at the test section are greater than those for undisturbed fully-developed turbulent pipe flow, and, since fully-developed flow has been re-established, this disturbance is attributable to the internal acoustic field generated by the bend. For all flow speeds there is an increase in wall acceleration and acoustic radiation spectra, over those for straight pipe flow (typical results shown in Figure 1). There is little effect at frequencies \( \nu(=\omega/\omega_p)<0.1 \); but, as \( \nu \) increases, there are dramatic increases.

Sound can propagate only as plane waves if \( k_c=\omega M_p c/\nu_p c<1.84 \) (where \( \nu_p \) wave speed in a plate of the pipe material and \( c \) is the speed of sound in the outside fluid). As \( k_c \) increases, higher order, \((p,q)\), acoustic modes become propagational. For steel, \( M_p=15.37 \) and the first, \((1,0)\), higher order mode has a cut-off frequency of \( \nu<0.12 \); increases in acceleration and radiation are associated with this mode. The effects are confined to frequencies close to the cut-off frequency because:

(i) the joint-acceptance between the acoustic modes and the \((m,n)\) resonant (supersonic) modes of the pipe peaks very sharply at coincidence of the acoustic and structural waves - when circumferential mode orders are equal \((n=p)\) and the longitudinal wavenumber of the structural mode, \( k_y=m/\pi \), equals that of the acoustic mode, \( k_y \); and

(ii) coincidence occurs at a frequency which is very close to the cut-off frequency of the acoustic mode (see Figure 2). Similar effects are evident for the \((2,0)\) acoustic mode whose cut-off frequency is given by \( k_c=3.054, \nu=0.20 \). At higher frequencies the acoustic modes are closer together and individual effects are more difficult to identify.

![Figure 1. Spectral densities of wall acceleration and acoustic radiation for 90° mitre-bend relative to straight pipe flow. Test section: internal radius=36.3 mm, length=1-2.52 m. \( \omega_c=ring \) frequency.!!](image1)

![Figure 2. Coincidence of acoustic and structural modes.](image2)
NONLINEAR ATTENUATION OF N WAVE PROPAGATING IN A TUBE WITH DISSIPATION DUE TO WALL EFFECTS

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INTRODUCTION

It is well known\(^1\) that variation of N waveform in a tube with propagation distance is contributed by nonlinear and tube wall effects. In this article, variation of slope of straightline segment of N waveform at zero crossing with propagation distance is discussed.

THEORETICAL EXPRESSION

The waveform, \(f(x,T)\), at \(x=x\) is given by following expression,

\[
f(x,T) = \frac{1}{\pi} \int_{0}^{\infty} e^{-Q(w)x} F(U,w) \sin \left(\frac{\pi}{c_0} - \frac{\pi}{c_p}\right) dw , \tag{1}
\]

where \(T=t-x/c_0\), and \(c_0, c_p\) and \(Q\) are infinitesimal sound velocity in an unbounded medium, phase velocity in a tube and attenuation coefficient due to wall dissipation, respectively. \(F(U,w)\) is the frequency spectrum of N wave at \(x=U\). In Fig. 1, a calculated result of eq.(1) for \(x=500\) cm is shown in a tube with radius of 2.5 cm. On the other hand, a result obtained by dropping out the \(\exp(-Qx)\) from eq.(1) is shown in Fig. 2. It is found that slope of straightline segment of N wave at the zero crossing is almost constant, while the position of zero crossing moves on the time axis. Therefore, the variation of waveform due to velocity dispersion is almost negligible for the variation of the slope.

RESULT AND CONCLUSION

The contribution of \(\exp(-Qx)\) to the variation of the slope at the zero crossing is equivalent with the variation due to attenuation of sinusoidal sound wave with angular frequency \(w\) given by following expression without velocity dispersion effect,

\[
f'(x) = \left[ \int_{0}^{\infty} e^{-Q(w)x} \omega F(U,w) \omega dw \right] = -Q(w)x , \tag{1}
\]

where ' means differentiation with time at zero crossing \(T=t\). Then,

\[
\frac{1}{f'(x)} \frac{df'}{dx} = -Q(w) . \tag{2}
\]

The variation of the slope due to nonlinear and tube wall effects is approximately additive for calculation of total amount,

\[
\left[ -\left(\frac{1}{f'}\right) \frac{df'}{dx} \right]_{\text{total}} = \frac{1}{2} \frac{f'}{c_p} + Q(w) . \tag{3}
\]

This equation agrees well with experimental results.

REFERENCE

1) A. Nakamura et al Acustica, 22(1969/70)88
INTRODUCTION

An accurate mathematical model of sound propagation in aircraft engine ducts is essential to the optimal design of quiet aircraft. This paper describes a finite element model which has been developed for axi-symmetric ducts of circular or annular cross-section with possible inclusion of circumferential splitters. Duct cross-section may vary arbitrarily with axial distance.

METHOD

The model is based on the usual perturbation of the Navier-Stokes equations for small motions. Mean flow parameters are derived in the absence of fluctuating quantities and are then substituted into the equations for the acoustic quantities which have been linearized by eliminating higher order terms. Mean swirl is assumed to be zero from the restriction of axi-symmetry.

A linear rectangular 'serendipity' element is formulated from these equations using a Galerkin procedure and assembled in a special purpose computer program in which the matrix map for a rectangular mesh has been specifically coded. Representations of the fluctuating quantities, mean quantities and coordinate transformations are iso-parametric. The global matrix is a block tridiagonal system of which the blocks are themselves banded matrices. The global matrix is held in packed form both in core and on secondary storage and is solved by forward and back substitution following an L-U decomposition with pivoting restricted internally to the blocks.

RESULTS

Although wavelength is a notional factor in multi-modal sound propagation and mesh density may ultimately be determined only by convergence of successive resolution increases, some idea of required mesh density may be gained from the fact that satisfactory results have been obtained with fewer than eight elements per wavelength in unimodal propagation. Computer central processor times of the order of 400 seconds for matrices of order 25,000 have been obtained in typical problems.

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SCHLIEREN OBSERVATIONS OF THE SCATTERING OF UNIPOLAR PRESSURE PULSES

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INTRODUCTION

We are employing Gaussian-like pressure impulses in laboratory scale studies of sound scattering by objects submerged in water. Since they are well defined in both space and time such signals are ideally suited for resolving the various waves that may be generated when an acoustic disturbance is incident on a complex object. In this paper we outline the technique used for generating 0.1 usec wide pressure impulses, discuss in some detail the schlieren visualization system we are using for examining the propagation of such impulses, consider briefly the theory involved in analyzing the observed images, and present some recently obtained results of studies of scattering by metal plates, solid cylinders and hollow shells.

EXPERIMENTAL TECHNIQUES

Gaussian shaped voltage pulses approximately 0.1 usec wide and 2 kilovolts in amplitude are applied at a repetition rate of 1 KHz to a thick lead zirconate-titanate transducer 5 cm in diameter and 3 cm thick. Each voltage pulse gives rise to a series of pressure impulses 0.1 to 0.2 usec wide, spaced in time by the acoustic travel time through the transducer, 5 usec, and in space by 0.75 cm in water. Using various acoustic probe techniques it is possible to show the individual pressure fronts are constant in amplitude over an area closely approximating that of the face of the source. The effect of the finite size of the source is made evident by a second impulsive wave that appears to emanate from a line source defined by the boundary edge of the source. This latter wave represents the diffraction phenomena occurring in this case.

We are employing a versatile schlieren system to visualize these acoustic impulses. An argon ion laser with an acousto-optic modulator is used to produce 0.02 usec light pulses synchronized, at an adjustable delay, with the repetitive voltage impulses applied to the acoustic source. Vidicon-TV monitoring of the optical intensity patterns is employed in both the transform and in the image planes, and a photomultiplier scanning system is used to obtain linear intensity plots in the image plane. By scanning the delay between the voltage pulse applied to the acoustic source and the laser optical pulse it is possible to visually follow the progress of incident and scattered waves on the TV monitor and analyze in detail various optical and acoustic phenomena.

CONCLUSION

A versatile system has been developed to study the scattering of acoustic impulses by objects submerged in water. The usefulness of the technique will be illustrated with motion picture recordings obtained with various scattering configurations.

ACKNOWLEDGEMENTS

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REFERENCES

RANGE DEPENDENT UNDERWATER SOUND PROPAGATION

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For an ocean sound channel whose environmental parameters depend not only on depth, but in a gradual fashion also on range, the wave equation may be separated by the adiabatic range variation method of Pierce (1). This method is used to calculate underwater sound propagation in a channel with a general range dependent sound velocity profile and boundaries (2), and is applied here to a parabolic sound velocity profile that opens up linearly with increasing range (Fig. 1). This profile will focus the sound of a source near the channel axis, as shown by the ray picture of Fig. 2.

Our results illustrate the feasibility of the adiabatic method for range dependent sound velocity profiles, and demonstrate the diminution of focusing power of the parabolic profile as its range dependence becomes stronger (Fig. 3, top to bottom).

We also evaluate the mode coupling coefficients for the case of non-adiabatic range dependence, and solve the coupled range equations in order to introduce mode coupling into the adiabatic method.


Fig. 1 Range dependent parabolic profile
Fig. 2 Focusing in range independent parabolic profile
Fig. 3 Shift of focal point due to range dependence
THE INVERSE FAST FIELD PROGRAM (IFFP): AN APPLICATION TO THE DETERMINATION OF THE ACOUSTIC PARAMETERS OF THE OCEAN BOTTOM

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INTRODUCTION

The problem of estimating the acoustic field produced by a point source located in an environment which is arbitrarily stratified with depth only has received considerable attention. Various models exist which provide reasonable results provided sufficient information about the environment and source characteristics is available. This paper is concerned with solving for the inverse of this problem, i.e., information about the environment, in particular the impedance at the ocean-bottom interface, is sought from a knowledge of the received acoustic field and the source which produced it. The theoretical formulation of the scheme is presented and its numerical implementation is discussed and illustrated for three examples of successive orders of environmental complexity.

THEORETICAL BASES

It is well known that the acoustic field $H(r,z,f)$ produced by a point monochromatic source located at depth $z_0$ in a cylindrically symmetric ocean can be expressed as the Fourier-Bessel transform:

$$H(r,z,f) = \sum_{m=-\infty}^{\infty} G(z,z_0,f,m) J_0(\eta r) e^{im\theta} df$$

where $G$ is the depth dependent Green's function and $J_0$ the zero order Bessel function. If one assumes the sound speed to be stratified with depth a general expression for $G$ is available which satisfies all boundary conditions and permits arbitrary location of the source and receiver. It has been shown (Ref. 1) that an economical and accurate evaluation for the field can be obtained in terms of a finite discrete Fourier transform. The success achieved with this approach has led to the speculation that the inverse solution may also be accurately recast as the finite Fourier transform:

$$2G(z,z_0,f,m) \approx \int H(r,z,f)e^{-im\theta}rdr$$

RESULTS AND CONCLUSIONS

Results have been obtained for the following three cases: (a) infinite constant sound speed ocean where the field consist of a single direct path, (b) semi-infinite constant sound speed ocean where the field consist of bottom reflected and lateral waves, and (c) a representative sound speed profile from the deep North Atlantic where the field is a complicated composition of numerous multipaths of different types.

REFERENCES

INTRODUCTION

Prediction and processing of underwater sound go hand in hand. Prediction of the underwater sound field enables the judicious placement of processing equipment to enhance the received signal in the presence of noise. Processing of the received time function is needed to further combat the noise and to estimate the desired signal. Prediction of the underwater sound and its processing in the presence of noise are performed using a common algorithm that capitalizes on the underlying similarity in both problems when formulated via an integral approach. Prediction of the underwater sound field calls for the solution of an integral equation. An analytical solution is not generally available. A direct numerical solution is not helpful when scaling laws are to be applied or more general results are desired for the purpose of processing the signal. Synthesis of the processor calls for the solution of an integral equation of the same class as the one encountered in the acoustic propagation problem (1-2).

ANALYSIS

On-going investigations aim to develop further solutions to such problems in prediction and processing (3). The approach has distinctive features that make it attractive from an analytical and computational points of view. For application to either prediction or processing, the procedure calls for splitting the total problem into two parts: (a) one part has an analytical solution which is expressible in terms of known functions; (b) the other part has the solution developed by a series of successive approximations which are continued until the contribution of additional iterations are negligible.

Other successive approximation techniques based on the Neumann series converge when the perturbation in part (b) is small. The present series is guaranteed to converge to a unique solution regardless of the form or size of the perturbation in part (b). The rate of convergence is more rapid than a Neumann series. Application of the present approach is formally illustrated by problems encountered in prediction and processing (1-2). Those include: (1) waves in an arbitrary inhomogeneous layer and estimation of a signal corrupted by both white and correlated noise in a finite observation time; (2) propagation modes supported by a given channel and detection of a signal in colored noise by a Karhunen-Loève expansion over a finite interval. In each case, the solution is formulated in a functional form. Then a common algorithm executes the series of mathematical operations. When the mathematical steps are mechanized, they are compatible with efficient computer operation. Discretization of the algorithm and its execution on a general purpose computer has been carried out. Comparative efficiencies have been measured through the Ratio $R$, where $R = \frac{\text{required computer time for techniques of comparable accuracy}}{\text{required time for present technique}}$. The results for problems of type (1) and (2) are respectively: Type (1), $R = 8.5$ for $n = 30$ and $R = 13$ for $n = 50$ where $n =$ number of discrete values; Type (2), $R = 13$ for $m = 10$ and $R = 35$ for $m = 20$ where $m =$ number of modes. The values of $R$ represent approximate lower bounds. It is expected that the values of $R$ will hold or improve in values when implementation is carried out on special purpose machines. This point will be discussed.

The work is supported by ONR and TR/IED Funds.

REFERENCES

Shear wave propagating with disorientation to the acoustical axis of monocrystals is known to have elliptical polarization. If the acoustical axis is axis of 3-fold symmetry, then phenomenon of conical refraction takes place (1). The acoustical birefringence as well as the phenomena mentioned above can be caused artificially by application of mechanical or electrical fields in crystals and isotropic bodies (2).

The propagation of waves with elliptical polarization is investigated by methods of Bragg diffraction and by electrodynamical receiver (3). Bragg diffraction method is based on the fact that diffraction efficiency depends on the polarization of acoustical waves. The rotation of the polarization plane results in periodical change of intensity of diffracted light in probing acoustical waves by laser beam (4). The value of the space period \( \Delta \) depends on disorientation angles of the acoustical axis in respect to the normal to crystal side. For the crystal LiNbO\(_3\) with disorientation angles from 17' to 3° the measured values of \( \Delta \) are in good agreement with theoretical ones (5). By means of modified schlieren technique the deflection of wave beams with orthogonal polarization from direction of wave vector caused by conical refraction effects was observed. The periodical intensity change on the light image was observed in the zone of the ultrasonic beams crossing alone (fig.1).

The measurements were carried out at frequency 470 Mc/s. In crystals with angle of the disorientation 17' the change of the value depending upon the electrical field intensity was measured. With the help of electrodynamical receiver the characteristics of elliptical polarized waves in monocrystals SiO\(_2\) at frequency 10 - 30 Mc/s were measured by two methods: 1) by means of measurements of several projections of the displace vector and 2) by means of measurements of two orthogonal components of the displace vector and difference of phase between them.

The polarization investigations are of great interest because it is necessary to know these phenomena in constructing acoustic devices and research. In addition the characteristics of shear waves polarization gives information concerning properties of crystals.

ULTRASONIC PHASE SPECTROSCOPY AND THE DISPERSION OF ELASTIC WAVES IN SOLIDS

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INTRODUCTION

We show how the dispersion relation and the phase and group velocities of elastic waves in wave guides or solids with microstructures can be determined from the phase spectra of spectral-analyzed broadband pulses.

THEORETICAL BASES

A pulse \( u(x,t) \) propagating in a linear medium \( x > 0 \) can be Fourier synthesized with harmonic waves \( \exp[iu(x-t/v-\xi)] \) of all angular frequencies \( u \), phases \( \xi \) [1]. The phase velocity \( v = \omega/k \) is, in general, a function of \( \omega \) in a dispersed medium. Given an input pulse \( u(0,t) = F(t) \) at \( x = 0 \), the response at \( x = l \) is

\[
u(l,t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \tilde{F}(\omega) e^{i\omega(t-l/v)} d\omega
\]

where \( \tilde{F}(\omega) = \int_{-\infty}^{\infty} F(t) e^{-i\omega t} dt = |\tilde{F}(\omega)| e^{-i\phi_0} \)

and \( \phi_0 = \omega t_0 \), \( t_0 \) being the time at which the input enters a specimen at \( x = 0 \). The Fourier transform of the response \( u(l,t) \) is

\[
\tilde{u}(l,\omega) = |\tilde{F}(\omega)| e^{-i(kt+\phi_0)} = |\tilde{F}(\omega)| e^{-i\phi(\omega)}
\]

It is now clear that from the measured phase spectrum, \( \phi(\omega) \), of a dispersed pulse, one is able to determine the dispersion relation,

\[
k(\omega) = (\phi(\omega) - \phi_0)/t \]

From this relation, one can easily calculate the phase velocity \( v(\omega) = \omega/k = \omega \phi(\omega) / (\phi(\omega) - \phi_0) \) and group velocity \( v_g = \omega/k = 2(\phi' / \omega - \phi_0)' \).

EXPERIMENTS AND RESULTS

Two broadband ultrasonic transducers are attached to both sides of a polished specimen. One transducer is excited by an electric pulse, and the signals received at the second transducer are recorded by a sampling oscilloscope. Through an A/D converter interfaced with a PDP 11/40 digital computer, the signal is digitized and then Fourier transformed to yield both the amplitude (power) spectrum and the phase spectrum. From the phase spectrum of the two transducers directly in contact (no specimen is between), we determine precisely the \( \phi_0 \). The spectral analysis of the signal through the specimen of known length then yields \( \phi(\omega) \).

In Fig. 1 is shown a typical shear wave pulse which has propagated in the filament direction of a 0.546 cm long boron-epoxy fiber reinforced specimen (96 ply). In Fig. 2 the dashed line is the measured and spectral analyzed dispersion relation; and the solid line is the group velocity curve obtained by numerical differentiation.

We note that for waves in dispersive media, the method of continuous wave \( \tau \)-point phase comparison [2] only measures the group velocity. Discrete results obtained by this method are also shown in Fig. 2. The agreement is very close.

TRANSMISSION OF LONGITUDINAL SOUND WAVES THROUGH ELASTIC WAVE GUIDES

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INTRODUCTION

This study investigates the propagation of longitudinal waves along nonhomogeneous straight wave guides made of several parts, see Fig. 1. The fundamental equation for the i'th element is:

$$q(E_{i}(z) \cdot \frac{\partial w(z)}{\partial z}) \cdot \frac{\partial w(z)}{\partial z} = \delta(p_{i}(\omega \cdot \partial z) \cdot \partial z; \omega = \omega(Z, t)).$$

ANALYSIS

The solution uses the method presented in (1), which shows that the Fourier transformed stress in the i'th element can be expressed as follows:

$$E_{i0}, Y_{i0} = f_{i0}(x) B_{i0} \exp(j \omega x - j \omega x) - B_{i2} \exp(j \omega x) \left| E_{i0} = K_{i1}(x) B_{i1} + K_{i2}(x) B_{i2} \right.$$

The transformed velocity is consequently:

$$V_{i0}(x) = j \omega \int \left[ Y_{i0} / \epsilon \theta(x) \right] dy - K_{i1}(x) B_{i1} + K_{i2}(x) B_{i2}.$$  

At cross sections between elements of the wave guide the following conditions should exist:

$$V_{i0}(l_{i}) = V_{i+1,0}(l_{i}); E_{i0}, Y_{i0}(l_{i}) = E_{i+1,0}, Y_{i+1,0}(l_{i}).$$

Using matrix notation, the relations between the coefficients vector $B$ are:

$$B_{i} = \sum_{k=1}^{n} K_{i1}(l_{i}) B_{i1}.$$  

Elimination of $B_{i2}$ is obtained by using the transformed boundary condition $D_{i0}$:

$$B_{i1} \hat{K}^{t} \cdot D_{i0},$$

which is independent of other unknown coefficients.

By inverse Fourier transform we finally get:

$$U_{i0}(l_{i}, t) = \frac{1}{2\pi} \sum_{n=1}^{N} F_{i0}(n) \left[ \exp(j \omega (l_{i} - n \omega)) \cdot \exp(j \omega (l_{i} + n \omega)) \right] - \sum_{k=1}^{n} K_{i1}(l_{i}) K_{i1}(l_{i}) \hat{K}^{t} \cdot D_{i0}.$$  

CONCLUSIONS

Using eq. (6), the parameters that appear in wave propagation in nonhomogeneous media are defined, namely:

$$\hat{K}^{t} = \text{amplitude modulation,}$$

$$\begin{bmatrix} \exp(j \omega (l_{i} - n \omega)) \cdot \exp(j \omega (l_{i} + n \omega)) \end{bmatrix} = \text{phase modulation,}$$

$$K_{i1}(l_{i}) K_{i1}(l_{i}) \hat{K}^{t} \cdot D_{i0} = \text{indication of the i'th region,}$$

$$\hat{K}^{t} = \text{function of elimination of } B_{i2}, \ D_{i0} = \text{actual boundary conditions vector.}$$

REFERENCES

An important problem of acoustics is that of sound transmission through a layer. Much progress has been made in the theoretical study of transmission through a thin layer (1). The model used is that of a panel occupying a cross-section of an infinite waveguide. Exact solutions have been found for transmission through a stretched membrane (2) and for a fixed or simply supported elastic plate (3-4). It has even been possible to find the exact solution for transmission through a double membrane partition (5).

Following this work it is natural to consider the case of transmission through a layer which cannot be treated as a panel, i.e. through a thick layer. Again a two-dimensional waveguide model may be used. A solution has been found for transmission through a layer of elastic solid contained in a waveguide with rigid lubricated walls (6). Each incident mode excites only the same mode on the other side of the layer.

Other boundary conditions do not seem to have been studied. Realistic conditions might be bonding to the walls of the guide, or alternatively free edges of the solid. Following previous methods the displacement in the elastic solid might be represented by an infinite series of fundamental solutions of the elastic equations. Each fundamental solution is made to satisfy the conditions at the walls of the guide. This gives rise to a transcendental frequency equation. The velocity potential in the two regions containing fluid are expanded in Fourier series and the following interface conditions are applied:

(a) continuity of normal displacement at each interface,
(b) continuity of normal stress at each interface. Some progress can be made in the mathematical solution of the resulting equations. Some progress could be made by means of approximations or numerical methods. However, a fundamental difficulty is that the additional condition of continuity of other stress at the interfaces cannot be satisfied. It thus seem doubtful whether approximate solutions of the equations would be worthwhile.

An alternative approach may be attempted using Fourier series expansions but so far the difficulties which appear with this method have also not been resolved. The difficulty appear to be in the assumptions that various series may be differentiated.

In view of the above difficulties, a better approach may be to take a simpler model of the solid layer. For example in the free boundary case the stress distribution may be approximately uniform over a cross-section of the solid. It may be possible to solve the resulting mathematical equations exactly. Thus any discrepancy between theory and experiment can then be clearly attributed to a defect in the model, and may perhaps be overcome by modifying the model.

REFERENCES
1. N. ROMILLY, Shock and Vibration Digest (1975), 7, 71
4. N. ROMILLY, Acustica (1973), 28, 234
5. N. ROMILLY, Journal of Sound and Vibration (1976), 48, 243
AN EXPERIMENTAL STUDY OF THE REFRACTION AND REFLECTION OF AN ELASTIC WAVE AT A SOLID-SOLID INTERFACE

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INTRODUCTION

The calculation of the energy partition of various modes of elastic waves at solid-solid boundaries has been carried out by many authors for numerous combinations of materials. The purpose of this paper is to show the results of an experimental study of the variation of the reflected amplitude of longitudinal and shear ultrasonic pulses at various epoxy-solid interfaces as a function of the angle of incidence.

TECHNIQUE

The solid-solid interface to be examined was formed by casting a hemicylinder of epoxy resin onto a thick plate of the second material under consideration. The combination was then immersed in water beneath the longitudinal wave transducers of an ultrasonic goniometer which are mechanised to move automatically in an arc with centre at the centre of the hemicylinder. For shear waves, water immersion coupling was replaced by viscous epoxy coupling. The frequencies used were variable between 1.25MHz and 20MHz. The reflected pulse amplitude was recorded automatically and displayed on a chart recorder.

RESULTS

In the case of a liquid-solid interface a simple theory, which does not take into account wave attenuation, fails to account for the shape of the reflected longitudinal wave amplitude at angles of incidence for which a surface wave is stimulated. A treatment such as that by Becker and Richardson or by Bertoni and Tamir is required. In the case of a solid-solid interface where mutual adhesion is good, however, we have found that the simple theory due to Nafe describes the experimental observations quite closely.

REFERENCES

(3) J.E. Nafe "Reflection and transmission coefficients at a solid-solid interface of high velocity contrast" Bull Seism. Soc. Am 47, 205, 1957
INTRODUCTION

The acoustical properties of isotropic elastic and viscoelastic materials are generally known and a number of techniques are available to determine them. Isotropic materials, however, offer a relatively narrow spectrum of properties to the acoustical engineer. To meet the increased demands on conventional materials, microinhomogeneous materials offer, through the combination of two or more materials, unique properties that are unattainable in either material alone. Typical examples in which new properties were obtained by introducing various inclusions or voids into a host medium are filled epoxy, syntactic foam, and expanded rubbers.

THEORETICAL BASES

If the inclusions or voids in the host material are distributed uniformly, if the distance between inclusions is such that there is no mutual interaction, and if the inclusion is smaller than the wavelength of sound, then, the elastic and viscoelastic properties of the material can be characterized by the effective values of density, modulus, impedance, or propagation constant.

In the present work we have investigated the acoustical properties of a number of rubbery (host) materials in which small spherical air cavities have been introduced causing a major change in the static and dynamic viscoelastic properties. In this case the host medium is a viscoelastic material and the equations for the effective elastic parameters of microscopically inhomogeneous elastic media were extended to the viscoelastic case. Assuming a Kelvin-Voight model of viscoelasticity the losses were accounted for by simply introducing complex moduli into the previously obtained elastic solutions. As expected, the major energy loss contributions are associated with the dampening factor of the shear modulus.

RESULTS AND CONCLUSIONS

Both analytical calculations and experimental measurements were carried out on rubber samples containing various amounts of air. Good agreement was found between experimental results and analytical predictions. The introduction of a small amount of air produced three dominant effects. The longitudinal sound velocity dropped sharply, the longitudinal loss coefficient rose sharply, and both of these effects were highly frequency dependent. For example, a two percent air content reduced the sound velocity at 10 kHz from 1600 m/s to 600 m/s and increased the absorption coefficient from 0.04 db/cm to 2.6 db/cm. Since the effective modulus is strongly dependent on the shear modulus, it follows that this behavior could be used as an experimental technique for determining the dynamic complex shear modulus or Poisson's ratio of rubber.

REFERENCES

LE CALCUL DE VALEURS MOYENNES DANS LES PROBLEMES NON LINEAIRES

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1. PROPOSITION FONDAMENTALE

Soit un problème dans lequel interviennent des grandeurs liées entre elles par des équations algébriques, différentielles ou aux dérivées partielles non linéaires. Si l'on impose à l'une de ces grandeurs d'être, dans une région, une fonction périodique à valeur moyenne nulle d'une des variables indépendantes, et que le problème admet une solution dans cette hypothèse, les autres grandeurs seront en général des fonctions périodiques à valeur moyenne non nulle. Il est évident que ces valeurs seront fonction du choix effectué pour la grandeur à moyenne nulle. Le calcul de la pression de radiation ou, en général, de $\rho$ dans un fluide parfait soumis à des vibrations constitue une illustration éloquente de cette constatation.

EXEMPLES

2.1. Onde plane progressive produite par un piston dont la vitesse $<v> = 0$

2.1.1. La solution d'Earmes ($\varphi(1859)$) permet d'obtenir très aisément, au droit du piston, en description lagrangienne, $<\rho> = \frac{(y+1)}{4}<\varepsilon>$ où $y = \frac{c_p}{c_v}$ et $\varepsilon$ est la densité d'énergie de l'onde, soit le résultat obtenu par Fubini Ghiron ($1935$) et confirmé par P.J. Westervelt ($1950$) aux termes d'analyses plus complexes.

2.1.2. Dans l'hypothèse où ce serait la dilatation linéaire $\beta$ qui est telle que $<\beta> = 0$, on obtient $<\rho> = \frac{(y+1)}{2}<\varepsilon>$ soit un résultat obtenu par L. Prillouin ($1935$), valable seulement dans cette hypothèse.

2.1.3. En un point fixe de l'espace (description euclidienne) on a la valeur négative connue (Westervelt $1950$) $<\rho> = \frac{(y-3)}{4}<\varepsilon>$

2.1.4. Dans l'hypothèse $<\varphi> = 0$ (variation de densité) on aurait $<\rho> = \frac{(y-1)}{2}<\varepsilon>$

2.2. Formules de King et de Langevin

On montre aisément que, en variables d'Euler, on a $<\rho> = <L_\rho> + <\psi_\rho>$ ($L_\rho$ fonction de Lagrange, $\psi$ potentiel des vitesses). La formule de King (et celle de Langevin qui en est un cas particulier) sont valables dans l'hypothèse où, en tout point de l'espace, on a $<\psi_\rho> = 0$, ce qui ne correspond à aucune hypothèse physique simple autre que celle énoncée par la formule de King elle-même, qui se réduit dès lors à une tautologie.

3. CALCUL DE $<\rho>$ DANS UN CYLINDRE LIMITE PAR UNE PAROI RIGIDE

A ma connaissance, ce calcul n'a pas encore été effectué. On obtient, au 3e ordre près ($V = \omega_0$ amplitude de la vitesse du piston, $L$: longueur du cylindre) $<\rho> = \frac{\rho_0}{2} \left[ 1 + 2 \cos k (x - L) + \frac{k^2}{2} \left( \cos 2k (x - L) + \cos 2k (x - 1) \right) \right]$

ce qui donne notamment, lorsque le cylindre est très court ($kL << 1$) $<\rho> = \frac{\rho_0}{2} V L^2 (y+1) + \frac{2}{L^2}$

qui n'est autre que la formule de Lucas ($1965$) lorsque les petits mouvements sont adiabatiques.

(5) P. Biquard. Revue d'Acoustique 1 p. 93 (1932)
(6) 5e C.I.A. Liège Rapports conf. gén. p. 163 (1965)
INTRODUCTION

Jet aircraft engines and atmospheric acoustic sounders are examples of powerful sources that are capable of generating extended fields of very intense sound outdoors. The question of whether nonlinear effects are important in such sound fields arises. Surprisingly, with the exception of studies of the sonic boom and explosion waves, very few measurements of finite-amplitude effects in open air have been reported. Some measurements have been made in the laboratory, but differences between laboratory and outdoor conditions—propagation distance, homogeneity of the medium, and presence of reflecting surfaces (such as the ground)—are great enough that direct studies in the open air need to be made.

We are interested in the propagation of two kinds of finite-amplitude signals: periodic waves and noise. We elected to investigate periodic waves first because of their comparative simplicity. Furthermore, to keep from introducing too many of the outdoor complications all at once, we selected a vertical propagation path. Thus, ground reflection effects were avoided.

EXPERIMENTS WITH FINITE-AMPLITUDE PERIODIC WAVES

The propagation of intense periodic waves has been studied along a vertical path parallel to an 85 m radio tower. Methods described in Ref. 1 were used to design the experiments. Three different sources were used, a circular array of seven JBL 375 horn drivers attached to short exponential horns, a triangular array of ten of the same elements, and a siren. Data indicating the performance and properties of the three sources are given in the table.

<table>
<thead>
<tr>
<th>Source</th>
<th>SLLm</th>
<th>f</th>
<th>$\theta_{HP}$</th>
<th>$\alpha$</th>
<th>$r_o$</th>
<th>$r_i$</th>
<th>$r_p$</th>
<th>$r_{max}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>7-Elem. Array</td>
<td>144</td>
<td>6.7</td>
<td>7°</td>
<td>0.046</td>
<td>1.5</td>
<td>11</td>
<td>830</td>
<td>33</td>
</tr>
<tr>
<td>10-Elem. Array</td>
<td>146.5</td>
<td>6.6</td>
<td>5°</td>
<td>0.046</td>
<td>1.4</td>
<td>6.1</td>
<td>110</td>
<td>38</td>
</tr>
<tr>
<td>Siren</td>
<td>149</td>
<td>6.3</td>
<td>40°</td>
<td>0.061</td>
<td>0.072</td>
<td>0.15</td>
<td>0.69</td>
<td>22</td>
</tr>
</tbody>
</table>

In this table SLLm is the source level (SPL extrapolated to 1 m); $\theta_{HP}$ is the beamwidth (angular separation between half-power points); $\alpha$ is the atmospheric attenuation factor; $r_o$ is the effective source radius; and $r_i$, $r_p$, and $r_{max}$ are the computed values of the shock formation distance, well-formed sawtooth distance, and distance at which small-signal behavior is restored, respectively. The fields produced by the three sources were subject to weak, moderate, and strong, nonlinear effects, respectively. When sufficient averaging time was used, experimental results generally confirmed predictions for homogeneous media. Interesting waveform changes qualitatively attributed to diffraction effects were observed. Experiments on propagation of finite-amplitude noise are planned. [Work supported by AFOSR, NASA, NOAA, and ONR].

2) L.A. Ostrovsky and A. M. Sutin, ibid, pp. 222-225.
ANALYSE DE LA PROPAGATION DES ONDES NON LINEAIRES PAR LE MOYEN DU PRINCipe DE HUYGENS

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Le principe de Huygens et l'existence des sources secondaires qui en dérivent ont trouvé dans l'holographie optique une confirmation éclatante et des applications spectaculaires. L'holographie acoustique en est l'holographie, base d'une théorie constructive permettant d'analyser et de développer rationnellement les techniques de la reproduction sonore en même temps que celles de l'insonorisation active.

Cependant les formules classiques traduisant mathématiquement le principe de Huygens perdent leur validité en acoustique non linéaire, alors que ce principe lui-même ne présuppose en rien la linéarité. D'autre part la multiplicité et la complexité des équations de propagation non linéaire entraînent le développement de théories particulières dont la généralisation à posteriori serait bien hasardeuse. Aussi conviendrait-il de porter l'effort sur la méthode même de traitement des équations, indépendamment des formes occasionnelles de ces dernières.

C'est pourquoi nous avons rattaché la formalisation du principe de Huygens à un théorème très général connu sous le nom de "lemme d'Urysohn". Nous allons l'appliquer ici à une équation d'onde du type

\[ Lp + Np = K \]

et à un exemple particulier commode :

\[ \left(c^2 \frac{\partial^2}{\partial t^2} - \text{div grad} \right) p + a \text{div grad} p^n = R. \]

Notations : \( p \) = pression acoustique ; \( a \) et \( c \) = constantes ; \( K \) et \( R \) = singularités (sources) ; \( n \) = entier supérieur à 1 ; \( L \) = opérateur linéaire ; \( N \) = opérateur non linéaire.

Or le lemme d'Urysohn affirme l'existence, sous des hypothèses très larges, d'une fonction continue \( s \) nulle dans un domaine choisi \( Y \), et égale à l'unité dans un autre domaine \( V \), \( V \) et \( V \) étant séparés par une zone \( Z \) à travers laquelle \( s \) variera de 0 à 1.

Appliquer le principe de Huygens revient à pondérer par la fonction de "découpage" \( s \) le champ \( p \) et les sources \( K \) ou \( R \). Ceci amène à transformer (1) en

\[ L(sp) + N(sp) = s K + H \]

où \( H \) sera une répartition convenable de sources de Huygens. Ces dernières se séparent en deux groupes :

\[ H^L = (Ls - sL) p \quad \text{et} \quad H^N = (Ns - sN) p \]

calculables explicitement pourvu qu'on sache écrire effectivement les opérateurs \( L \) et \( N \). Appliquées à l'exemple (2) les formules (4) donnent :

\[ H^L = - p \text{div grad} s - 2 \text{grad} s \cdot \text{grad} p \]

\[ H^N = a p^n \text{div grad} s^n + 2a \text{grad} s^n \cdot \text{grad} p^n + (s^n - s) \text{div grad} p^n. \]

Ainsi les non-linéarités des équations (1) ou (2) sont reportées dans les sources \( H^N \) même si l'on ne sait pas résoudre explicitement les équations de départ (1) ou (2), en utilisant les formules de rayonnement classiques du régime linéaire. Les approximations ainsi obtenues peuvent être utilisées pour un calcul de proche en proche du rayonnement d'une source primaire (\( K \) ou \( R \)); pour cela il convient d'utiliser une famille de fonctions \( s \) construites à partir d'une succession de zones \( Z \) emboitées les unes dans les autres et centrées sur la source primaire considérée. La vérification numérique des résultats peut se faire par simple report dans les équations primitives (1) ou (2).
OBSERVATIONS OF THE IMPULSIVE SOUND GENERATED BY THE WIRE EXPLOSION IN AIR

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INTRODUCTION

In order to realize a practical impulsive sound source in air, the wire exploding method is investigated, because stable reproductivity of waveforms and possibility of high intensity were expected.

EXPERIMENTAL METHOD

The experimental arrangement is shown in Fig.1. The waveforms of sound pressure and discharge current were observed under various conditions of wire dimension and discharge circuit parameters (charged voltage, capacitance, wire length and diameter). The microphone (B&K 4135) is fixed at 2m away from the sound source and the material of wire is a copper.

EXPERIMENTAL RESULTS AND CONCLUSIONS

The sound waveforms are considerably influenced by the exploding conditions; the peak level of sound pressure increases with the increments of charged voltage and capacitance, and there are some conditions on the wire length which make the maximum peak level. (see Fig.2) According to above results, the peak level is controlled by the exploding parameters.

The generated sounds are classified in the three types according to their current waveforms; (a) no arc-discharge type, (b) arc-discharge type, (c) "Dwell" type. (see Fig.3.) (a) Discharge current does not flow and the sound is generated by the only wire explosion. In this case, the peak level of sound pressure is small in general, and the pressure waveform is very simple, hence it seems preferable for the measurements of building acoustic. (b) Discharge current flows instantaneously after the wire explosion, and the generation of the sound is made by the overlapping of the wire-explosion and the arc-discharge. In this case, the peak level is very large and rise-time is short, hence it seems preferable for the measurements of sound propagation between long distances or for the source of the finite amplitude sound. In general, type (a) and (b) have a good reproductivity. (c) Discharge current does not flow after the wire explosion, but begins to flow after some period of current pause. The waveform of sound pressure is divided into two parts; one is generated by the wire explosion and the other by the arc-discharge. Being bad to the reproductivity, this type is not good for the acoustic measurements.
Most often acoustic waves are produced by a moving radiating source's surface, while a wave is observed in the constant point of an acoustic field. The approximate solution of hydrodynamic equations with Lagrangian boundary conditions/moving boundary conditions/MBC/ for plane, spherical and cylindrical waves of an optional form, propagating in an ideal gas is presented in this paper. This material is the generalization of results presented in the papers [1-2-3-6-7] concerned with the propagation of sinusoidal or periodic waves exciting in the constant point of an acoustic field (the problem with Eulerian boundary conditions).

In the case of the plane wave, taking into account the well known solution of hydrodynamic equations for simple outgoing plane wave: \( u(t) + \frac{c_0 + \beta u}{u} u_{x0} = 0 \), the approximate solution of hydrodynamic equations is given by, [4]:

\[
\frac{u(t)}{E(t)} = \frac{1}{f} \left[ \frac{1}{E(t)} \sum_{n=1}^{N} A_n \sin \left( \frac{2\pi n t}{T} \right) + \frac{1}{E(t)} \sum_{n=1}^{N} \frac{1}{2} A_n \sin \left( \frac{2\pi n t}{T} \right) \right]
\]

where \( E(t) \) is the function describing the source's surface motion, \( E(t) = E_0 \), etc. The relationship (1) is obtained by the convert Lagrangian boundary conditions to the Eulerian form [5] and then by using Earnshaw's parametric solution of hydrodynamic equations. In this relationship there are only the components which the relative magnitude in relation to the first component is no less than \( 10^{-4} \).

For instance, let \( E(t) = A \sin \omega t \), \( A = 10^{-2} \). Harmonic components of signal \( u(x,t) \) are dependent upon \( \omega x \); nonlinear propagation /NP/ in medium influences or decrease of harmonic with the distance from source, while influence of MBC effect is independent of distance /see Fig. 1/. The point in which both effects are equal is: for second harmonic \( \omega x = 280 \) ms, for third harmonic \( \omega x = 95 \) ms.

For spherical and cylindrical waves, taking into consideration the results presented in the papers [1-2], the approximate solution of hydrodynamic equations for \( r > \lambda_{nm} \) was obtained. Then, using Taylor's series, the description \( u(t) \) of the acoustic field was worked out when the source of wave is a pulsating sphere or a cylinder with a radius given by \( r_{nm} = E(t) \). Since NP effects in spherical and cylindrical waves are less than in a plane wave and since MBC effects are the same in both the plane and spherical or cylindrical waves, the MBC effects relative magnitude is greater in spherical and cylindrical waves.

REFERENCES

The study of linear and nonlinear wave problems may be carried out in the framework of the Hamiltonian formalism in both homogeneous and stratified media from the unitary point of view in the rather general statement. The basic difficulty of the method is determination of the canonical variables \([q, \pi]\), which permit to write the motion equations in the canonical form \( \dot{q_i} = \pi_i \), \( \ddot{q_i} = \partial H / \partial \pi_i \).

Here \((q, \pi)\) are the conjugate pair of canonical variables, functional \(H\) -Hamiltonian of the medium, \(H\) being in agreement with the medium energy in general case and symbol \(\partial / \partial\) means variational derivative. The canonical variables have been introduced for hydrodynamical type of media disposed in the external gravitational and nonhomogeneous magnetic field \(B\) and also for self-gravitated media by taking into account relativistic effects of the general theory of relativity. Finally the media the boundaries of which are moved about can be introduced in this method.

The advantage of this method is that the solution of any concrete problem can be carried out by means of the standard way. As an example we may consider such fine effects as radiation of low frequency internal waves by power acoustic packet \([2]\), phonons acceleration in a turbulent medium \([4]\), build-up of equilibrium wave spectrum on the ocean surface \([5]\) and so on. It should be noted, that asynchronous of the wave interaction in the stratified media can play the considerable role.

References:
INTRODUCTION
For liquid helium below $T_\lambda = 2.17$ K ("He II") the two-fluid model assumes that it is composed of two intermingling but noninteracting liquids: superfluid and normal fluid. This assumption leads directly to two distinct forms of wave motion. When the two fluids move in phase a pressure wave or "first sound" is transmitted. The motion of the two fluids out of phase is a temperature wave or "second sound". This second kind of "sound" is unique to He II since all other substances transmit heat by diffusion. Both kinds of sound are nonlinear in the sense that pressure or temperature perturbations travel slightly faster or slower than the acoustic speed. Consequently, finite-amplitude perturbations will steepen to form shock waves.

THEORETICAL BASIS
Following Khalatnikov (1), one can derive the following governing equation for weak second-sound shock waves transmitted from the surface of a solid heater in He II:

$$C_2 = C_{20} + \tau_2 \frac{Q}{\rho s T},$$

where $C_2$ is the second-sound shock speed, $C_{20}$ is the second-sound wave speed, $Q$ is the heat flux into the liquid, $\rho$ is the density, $s$ is the entropy per unit mass and $T$ is the temperature. The coefficient $\tau_2$, as has been deduced theoretically and partly verified experimentally (2), is not always positive like the corresponding coefficient in the formula describing the speed of first-sound shock waves, but it is negative in the temperature ranges $0.5 K < T < 0.95 K$ and $1.37 K < T < T_\lambda$. As a consequence, the shock travels at the back of the wave in these ranges, this being a striking peculiarity of the second-sound wave transmission in He II. Further, the velocity of the shock will decrease with increasing $Q$ in the ranges of negative $\tau_2$.

EXPERIMENTAL RESULTS
The theoretical predictions have been checked and verified experimentally; the results superseding some only qualitative results obtained 1951 by Osborne (3). Fig. 1 shows an example of the new results. To obtain this picture, single rectangular heat pulses were transmitted from a plane heater into bulk helium and were then detected at a distance of about 10 cm by a fast temperature measuring probe. At a constant helium temperature of $2.09 K$, the heat flux was increased and the horizontal position of the detector signal was lowered on three successive runs. The detector signals shown clearly demonstrate the validity of several predictions: At $T = 2.09 K$ a shock forms at the trailing edge of the heat pulse. The foot of the wave arrives at the detector at the same time for all three runs (it travels at the acoustic velocity $C_{20}$), while the shock arrives later as the heat flux is increased.

REFERENCES
(1) I.M. Khalatnikov, Intr. to the Theory of Superfluidity, Benjamin, N. Y. (1965)
(2) A.J. Deissler, and W.M. Fairbank, Phys. Rev. 104 (1956), 6
SPECTRAL ANALYSIS OF PARTICLE VELOCITY FOR SOUND WAVES OF LARGE AMPLITUDE AND INTERMEDIATE MACH NUMBER IN LOSSLESS FLUIDS

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In recent years problems involving nonlinear propagation of high intensity acoustic noise have been discussed in some papers [1-4-5]. The problems are also connected with outdoor propagation experiments [6] when acoustic signals are described by stochastic processes [2-5].

In this paper some results of the nonlinear transformation of the particle velocity spectrum (PVS) for plane, spherical and cylindrical waves in lossless fluids are presented. The results were obtained in the case of Gaussian excitation, and when the acoustic range is 10 Hz - 20 000 Hz (one can say that is due to the intermediate Mach number).

Let us consider the finite amplitude kinetic excitation (displacement of source surface) \( \{ \xi(t), t \in (-\infty, \infty) \} \) of a plane wave as a stochastic stationary Gaussian process with the mean value equal zero, and \( B(t) = B(\xi(t), \xi(t+\tau)) \) is fixed. Because \( \xi(t) \) is the moving boundary condition (MBC) for the plane wave eq. [7], it is necessary to transform \( \xi(t) \) to \( u(0,t) \) using Earnshaw’s parametric eq. and next, to describe the nonlinear propagation of the particle velocity signal (NP: \( u(0,t) \rightarrow u(x,t) \)). The approximate formula expressing the dependence between \( u(x,t) \) and \( \xi(t) \) is given in [7]. Using the formula and using two new theorems referred to multidimensional moments and correlation functions of Gaussian processes [2] leads to the following result:

\[
S_n(x,w) = S + \frac{1}{2\pi} \int \left[ -2\pi B_y^2(0) \delta(\omega) - 2B_y(0)S + \frac{1}{2\pi} (\omega^2 S)^* (\omega^2 S)^s + \frac{1}{2\pi} S* S^* + \frac{1}{2\pi} S* \right] \bigg( \frac{1}{\omega_0^2} + \frac{1}{\omega^2} \bigg) \bigg( \frac{1}{\omega_0^2} + \frac{1}{\omega^2} \bigg) \bigg[ \bigg( \frac{1}{\omega_0^2} + \frac{1}{\omega^2} \bigg) \bigg], \tag{1}
\]

where \( S_n(x,w) \) is PVS, \( S* \) is the correlation function of the excitation process, \( B_y(t) = B(\xi(t), \xi(t+\tau)) \) is the derivative of \( \{ \xi(t), t \in (-\infty, \infty) \} \) in the Hilbert space, \( x \) is the distance from the source (\( x < X \)), \( X \) is the shock formation distance, \( \gamma \) is the ratio of specific heats, \( c_0 \) is the small-signal speed of sound, and \( * \) is the convolution symbol. The expression (1) is valid for the acoustic frequency range and for the sound intensity less than the intensity corresponding with \( \mathcal{A} = 0 \gamma^2 / c_0^2 \times 10^{-2} \) (\( \mathcal{A} \) is Mach number in a stochastic case). The terms in the first bracket in Eq.1 correspond to MBC and the terms in the second one correspond to effects of nonlinear propagation (NP). Fig.1 shows that the distortions of \( S \), corresponding to MBC (continuous line) are dominant in the lower frequency range and NP distortions (dotted line) are dominant in the higher one. It may be shown that the analogous conclusion is true for spherical and cylindrical waves [2]. Note that the similar result, but for a deterministic excitation, was obtained by Frost and Harper [3].

REFERENCES
INTRODUCTION
The equations of motion become nonlinear for finite amplitude Lamb modes propagating in isotropic plates. Hence, mode interactions may occur. Two such interactions are the Lamb mode three-phonon interaction and Lamb mode second harmonic generation. These two interactions are investigated experimentally.

THEORETICAL BASES
The nonlinear equations of motion for an isotropic plate yield criteria for and characteristics of the generated waves. Specifically, sufficient conditions for generating third phonons are

\[ \omega_3 = \omega_1 + \omega_2 \quad \text{and} \quad k_3 = k_1 + k_2, \]

where \( \omega \) and \( k \) are the angular frequencies and wave vectors of the Lamb modes. The generated third mode has an amplitude

\[ A_3 = A_0 B_0, \]

where \( A_0 \) and \( B_0 \) are the amplitudes of the generating (pump) modes.

A necessary condition for Lamb mode second harmonic generation is

\[ v_2 = v_1, \]

where \( v_2 \) and \( v_1 \) are the velocities of the fundamental and the second harmonic Lamb mode. A generated second harmonic has an amplitude

\[ A_2 = (A_1)^x, \]

where \( A_1 \) is the amplitude of the fundamental and \( x \) is the propagation distance.

EXPERIMENTAL TECHNIQUES
The criteria for generation, eqs. 1 and 3, are used to set up the pump modes for the two interactions. The means of detection of possible generated modes is an optical probe where a laser beam interacts with plate modes to yield a light reflection/diffraction pattern (1). The velocity and frequency of the constituent Lamb modes may be obtained from this pattern. If harmonics are present, the diffraction pattern is asymmetric (2). Further, if a third phonon is generated, then a third diffraction pattern is present.

RESULTS
In the second harmonic experiment, an asymmetric light diffraction pattern is observed. Further, the asymmetry increases with propagation distance, \( x \). This effect is predicted by eq. 4 for a generated Lamb second harmonic. Thus harmonic generation in Lamb modes is observed.

In the three-phonon interaction experiment, a third diffraction pattern is detected. The Lamb mode which creates the pattern has a velocity and frequency which corresponds to the calculated values for a third phonon. Further, the amplitude of this Lamb mode, as given by eq. 2, is measured. Hence the three-phonon interaction is observed for plate modes.

REFERENCES
Due to the ultra-precision requirement of crystal resonators in frequency standards and for frequency control, the effect of external forces on changes in resonant frequencies has become a topic of increasing importance in both experimental and theoretical investigations. The effect of statically applied forces on the frequency changes had been experimentally studied extensively by Ballato (1), Ballato and Bechmann (2), Mingins, Perry, and Marcus (3), and Ratajski (4). Experimental studies on the changes in resonances due to accelerations through body forces were reported by Warner and Smith (5), and also by Valdois, Gagnepain, and Besson (6).

The theoretical aspect of these problems belongs to the general theory of incremental elastic vibrations superposed on initial finite deformations. In a previous paper (7), a system of six two-dimensional equations, accommodating the coupling of the flexure, extension, face-shear, thickness-shear and thickness-twist modes, was derived for waves of small amplitude superimposed on finite deformations due to initial stresses. In these equations, the nonlinear terms associated to the third-order elastic stiffnesses in stress-strain relations were included. These equations were applied to the studies of frequency changes of circular rotated Y-cuts of quartz which were either subjected to a pair of diametrical forces (7) or subjected to initial bending due to transverse, applied forces (8). The predicted results were compared to experimental data (1-4) with close agreement.

In the present paper, it shall be shown that this method of approach can also be applied to the study of frequency changes of circular crystal plates subjected to steady in-plane acceleration of arbitrary direction.

First, a close form solution for a circular plate under acceleration with three or more points of mounting is obtained. From this solution, initial stresses and strains in the plate are computed as function of plate orientation, direction of acceleration, and positions of supports. Secondly, these initial fields are taken into account in the coupled incremental vibrational equations through the second- and third-order elastic stiffness coefficients of the crystal.

Due to the non-uniformity of the initial fields and the smallness of the frequency changes ($\Delta f/f_0$ in the order of $10^{-5}$), a perturbation method is used to calculate the changes in the thickness-shear resonances. The predicted frequency changes are computed for AT-cut of quartz and are compared with experimental data (5).

REFERENCES

INTRODUCTION

The characteristic features of acoustic amplification by combustion were studied in our earlier papers[1,2]. The experimental and theoretical result of this acoustic phenomena explained in linear approximation the feedback noise generation and the associated dipole radiation pattern. This paper analyzes some consequences of the nonlinear amplification response of flames, leading to odd harmonic generation.

THEORETICAL BACKGROUND

A combustion type chain reaction behaves similarly to an acoustically active amplificative medium having basically exponential transfer characteristics. Since this is a function of frequency, it is composed of its spectral constituents. Pressure variation output can lead – theoretically – to rising slope enhancement or even to the formation of shock wave, but for a flame, given by experimental conditions, it is simply measurable.

EXPERIMENTS

The investigation was performed with an acoustically excited burner and spectra were recorded with /upper curve/ and without /lower curve/ flame /see Fig. 1./ This enabled us to analyze the net acoustic amplification with its harmonic content for the thermodynamic conditions of the experiment. This shows high odd harmonic peaks for 250 Hz excitation and 50 Hz residual hum. The high frequency range part contains the quadrupolar turbulence noise, induced by the acoustic excitation.

CONCLUSION

Acoustic perturbation of combustion amplifies nonlinearly the input acoustical signal, and generates high odd harmonic content. This distortion is still, however, a coherent one. This result is in good agreement with the detected large combustion oscillations in resonator interaction [3]. Here this system similarly showed the tendency to oscillate at odd harmonic modes.

REFERENCES

INTRODUCTION

Preliminary measurements are reported on the removal of energy from the fundamental frequency of an underwater sound source due to its nonlinear interaction with noise.

THEORETICAL BASIS

The theory of the absorption of sound by sound has been developed by Westervelt (1), who obtained a formula relating the sound absorption coefficient $\alpha$ to the energy density spectrum of the noise $u(k)$. In the case in which the noise frequency lies below the signal frequency, this formula becomes

$$\alpha = \left( \frac{4\rho_0 c_0^2 v}{\beta^2} \int_0^{k_1} \frac{u(k)}{k^{1-2}} \right) \frac{1}{B}$$

where $k_1$ is the wave vector of the signal and $\beta$ is the coefficient of nonlinearity.

EXPERIMENTAL TECHNIQUE

A continuous sound signal of frequency 2.4 MHz was produced in water and detected by quartz transducers. The receiving transducer was tilted slightly to avoid standing waves and its holder was wrapped with packing foam so that only the transducer face was exposed to radiation.

The noise was provided by a mechanical motor-driven hammer, striking the metallic wall of the tank. This resulted in a noise spectrum that was fairly flat up to about 5 kHz. The signal and transducers were mechanically decoupled from the tank to avoid direct signal interference.

RESULTS AND CONCLUSIONS

The frequency spectrum of the sound signal in the presence and absence of the noise is shown in the drawing. The spectrum is clearly spread out as a result of the interaction, the bandwidth increasing from 40 to 300 Hz. Quantitative comparison with the equation of Westervelt awaits better analysis of the frequency spectrum of both the noise and signal, and such research is underway.

REFERENCES

ON THE THEORY OF THE COLLECTIVE MOTION IN LIQUIDS

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INTRODUCTION
This report presents the selfconsistent perturbation theory of a
density-density Green's function in different regimes and the theory
of propagation of the collective modes in liquids.

THEORETICAL BASES
A sound wave can be generated in a liquid by an external force \( F(t) \)
which compresses the liquid periodically. The longitudinal sound
amplitude \( a(k,t) \) can be calculated in linear response as

\[
a(k,0) = G(k,0)F(0),
\]

where \( G(k,0) \) is the Fourier transform of the retarded density-density Green's
function. We use the standard projection-operator method to derive
the Dyson equation for the Laplace-transformed Green's function

\[
G(k,\Omega) = G(k,\Omega)[Q(k,\Omega) + i\gamma(k,\Omega)]^{-1},
\]

where \( Q(k) \) is the frequency matrix, \( \gamma(k,\Omega) \) is the damping matrix. We find

\[
a(k,t) \sim \exp\{i\varepsilon(k)t\} \exp\{-\Gamma(k)t\},
\]

where \( \varepsilon(k) = Q(k) + \text{Im} \gamma(k,\omega_k) \) and \( \Gamma(k) = \text{Re} \gamma(k,\omega_k) \).

The damping matrix \( Q(k) \) is evaluated trivially and exactly as usual. To evaluate the
damping \( \Gamma \) we derive the set of exact coupled differential equations
for the damping matrix \( \gamma(k,t) \) and its self-energy terms.

The first approximation would be simply to set \( \gamma = 0 \) in the
"collisionless" regime \( \Omega/\gamma \gg 1 \).

RESULTS AND CONCLUSIONS
The collective modes propagate in the "collisionless" regime with the
phase velocity of the "zero sound" and are slightly damped by Landau.
In the "collision-dominated" regime \( \Omega/\gamma \approx 1 \) we have the hydrodynamical
sound modes and strongly damping temperature waves.

The main aspect of the intermediate regime \( \Omega/\gamma \sim 1 \) is the appearance
of the viscous-elasticity. The collective modes are strongly
damped in these regimes.

The poles of the retarded density-density Green's function define
the velocity and absorption of the collective motion in liquids.
To evaluate the Green's function and the damping matrix, one can
generate the approximate kinetic equation for \( G(k,\Omega) \) and \( \gamma(k,\Omega) \) in a
consistent fashion by utilizing various approximations for its self-
energy term.

REFERENCES
INTRODUCTION

The theoretical result /2/ presented in this paper describes the changes of acoustical field due to the motion of source. Intensity of sound generated by the point source at the rest is:

\[ I = \frac{W(\omega) Q(n, \omega)}{4 \pi r^2} \]  

where: \( W(\omega) \) — acoustic power
\( Q(n, \omega) \) — directivity factor in the direction "source-point of observation" given by unity vector \( n \).
\( r \) — distance from source to the point of observation
\( \omega \) — frequency

The motion of source modifies the directivity factor: \( Q \rightarrow \bar{Q} \).

THEORY

Let us suppose that the source undergoes the motion with the speed \( V \) and acceleration \( \ddot{V} \). Then the final result for intensity of sound field is:

\[ I = \frac{W(\omega)}{4 \pi} \left[ \frac{Q(n, \omega)}{r^2} \left\{ 1 + \frac{(V)^2}{\omega^2 c^2 (1 - \frac{n \cdot V}{c})^2} \right\} \right] \]

The square brackets imply evaluation of their contents at retarded time:

\[ t' = t - \frac{r(t)}{c} \]

where: \( c \) — speed of sound.

CONCLUSION

Comparing eqs. 1 and 2 we see that:
- The directivity factor of source in the motion \( \bar{Q} /2/ \) depends not only on the unity vector \( n \) and frequency \( \omega \), but also on speed \( V \) and acceleration \( \ddot{V} \).
- When the motion is accelerated \( V \neq 0 \) and complex sound is generated, the "space patterns" of harmonic components are different from each other: \( \bar{Q}/Q = f(\omega) \).

REFERENCES

RAYONNEMENT D'UN DISQUE PLACE DANS UN BAFFLE DE DIMENSIONS FINIES

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ETUDE THEORIQUE

L'action du disque est caracterisee par la force qu'il exerce sur l'air; celle du baffle est introduite comme conditions aux limites.

La resolution de l'equation d'onde(1), dans le domaine detline par le contour en pointilles ci-contre, permet de separe
les expression de la pression sonore en deux termes: le premier est egal au
champ sonore cree par le disque en
l'absence de baffle(2) et le second est
la contribution du baffle. Ce dernier
term exprime que le baffle exerce
sur chacune de ses faces une force
opposee a la pression au niveau de sa
surface; on peut montrer que cette pression est independante des dimensions du
baffle et qu'elle est ainsi donnee par l' integrale de Rayleigh.(3).

ETUDE EXPERIMENTALE

Le systeme etudi est un disque de polystyrene entraine par un moteur de haut-
parleur. Les pressions sonores sont relevees, depuis le voisinage immediat de la
surface vibrante jusqu'a environ 50 cm de celle-ci, en maintenant l'excitation et la
frequence constante. Resultats theoriques et experimentaux sont alors comparés.

REFERENCES

(2) A.M. BRUNEAU, Acustica (1973) 28 238.
(3) E. SKUDZIKY, The foundations of Acoustica, Springer-Verlog (1971) 496
INTRODUCTION

Les variables d'Euler n'étant pas bien adaptées pour représenter le champ acoustique au voisinage d'une surface de discontinuité du milieu, nous utiliserons les variables de Lagrange.

THEORIE

Notations : \( p = \text{densité} ; \mathbf{u} = \text{vitesse} ; p = \text{pression} ; \)

\((\rho_0, \mathbf{u}_0, p_0)(\mathbf{x}) = \text{état du fluide dans la configuration de référence (le repos) et deux écoulements voisins} : \)

\((p_E, \mathbf{u}_E, p_E)(\mathbf{x}, t) = \text{écoulement d'entraînement} \)

\((\rho, \mathbf{u}, p)(\mathbf{x}, t) = \text{écoulement perturbé} \)

Le champ acoustique \((\rho_A, \mathbf{u}_A, p_A)\) est défini par :

\[ (\rho_A, \mathbf{u}_A, p_A)(\mathbf{x}, t) = (\rho_0, \mathbf{u}_0, p_0)(\mathbf{x}) - (p_E, \mathbf{u}_E, p_E)(\mathbf{x}, t) \]

et vérifie les équations de conservation de la masse et de la quantité de mouvement écrites en coordonnées de Lagrange :

\[ \frac{\partial}{\partial t} \left( \frac{\rho_A}{\rho_E + \rho_A} \right) + \nabla \cdot \mathbf{u}_A = 0 \]

\[ \frac{\partial}{\partial t} (\rho_A \mathbf{u}_A) + \nabla p_A = 0 \]

Soit \( \Phi(\mathbf{x}, t) = 0 \) l'équation d'une surface de discontinuité \( \Sigma \) dans la configuration de référence, de normale \( \mathbf{n} = \nabla \Phi \) : \( |\nabla \Phi| \), de vitesse normale \( \mathbf{W}_n = - \partial \Phi / \partial t : |\nabla \Phi| \).

Considérant les équations (2) au sens des distributions, on obtient les équations aux discontinuités suivantes :

\[ - \frac{\rho_0 - \rho_A}{\rho_E + \rho_A} \mathbf{W}_n + \mathbf{u}_A = 0 \]

\[ - \rho_0 \mathbf{u}_A \mathbf{W}_n + p_A \mathbf{n} = 0 \]

où \( \mathbf{n} \) désigne le saut à travers \( \Sigma \).

Dans le cas particulier du dioptre, \( \Sigma \) (toujours constitué des mêmes particules) a une vitesse normale nulle : \( \mathbf{W}_n = 0 \) : alors les équations (3) deviennent :

\[ \mathbf{u}_A = 0 \]

\[ p_A = 0 \]

CONCLUSION

Les équations en coordonnées lagrangiennes pour le champ acoustique superposé à un écoulement, sont prises au sens des distributions. Après avoir obtenu les conditions de saut qui doivent vérifier les grandeurs du champ acoustique à la traversée d'une surface de discontinuité \( \Sigma \), nous avons montré que, dans la configuration de référence, la vitesse normale et la pression acoustiques sont continues à la traversée d'un dioptre, que l'on soit en acoustique linéaire ou non.
INTRODUCTION

This paper describes both theoretical and experimental work on the transmission of sound through a circular plate clamped at its edges forming a partition across a cylindrical waveguide. The waveguide has a hard termination beyond the plate so that the plate is backed by a finite length cylindrical cavity. The theoretical treatment is based on that of Romilly for a circular membrane (1) and the predictions obtained have been compared with experimental measurements.

THEORY

An expression has been obtained for the complex ratio of the amplitudes of the $n$'th symmetric transmitted mode and the $m$'th symmetric incident mode in terms of the frequency and the parameters of the system. The expression obtained contains an infinite sum, but as the series to be summed is rapidly convergent this presents no computational problem (usually 3 or 4 terms suffice). The result is, however, too lengthy to be reproduced here.

EXPERIMENTAL WORK

Measurements have been made by placing the experimental system at the end of a long cylindrical tube (2) and exciting it by short duration swept sine wave pulses, the time history of the response of the system to the pulse being captured in a digital event recorder for subsequent computation. Only plane waves have been used for the incident wave; the experiment measures the sum of the pressures of all the transmitted modes excited by this one incident mode. To facilitate understanding of the results all measurements have been made at one place, namely the back of the cavity. The results for the complex transmission coefficient have been obtained using the insertion loss technique, the final result being computed via numerical Fourier transformation. Measurements have been made in the frequency range below 1 kHz using various plates all 126 mm in diameter but of various thicknesses and materials, chosen so that some plates had one and some two symmetric modes in the frequency range used. The cavities used ranged up to two wavelengths in depth.

The separate resonances and antiresonances that would be expected for the plate and the cavity alone can be seen to be shifted in frequency by the presence of the other element. The extent of this shift is highly dependent on the proximity of the resonance or antiresonance in one element to a resonance or antiresonance in the other.

REFERENCES

DIE DARSTELLUNG DES NÄHFELEDERS EINER GERADEN STRAHLERGRUPPE UND EINES LINIENSTRAHLERS

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FINLEITUNG


DIE LINEAREN STRAHLERGRUPPEN

Um das Schallfeld darzustellen wurde zunächst die Gleichung für die Schalldruckamplitude in einem beliebigen Aufpunkt in der Nähe der linearen Strahlergruppe aufgestellt. Nach Ausrechnung der Schalldruckamplitude für eine große Anzahl von Aufpunkten, wurden durch Interpolation Punkte gleicher Amplitude ermittelt und miteinander verbunden. Das Ergebnis ist ein Schallfeld, dargestellt durch Kurven konstanter Amplituden.

DIE LINIENFÖRMIGE STRAHLERGRUPPE

Läßt man die Abstände zwischen den einzelnen Strahlern immer kleiner werden und die Anzahl der Strahler beliebig wachsen, so erhalten wir eine Strahlerstrecke. In der Praxis bedeutet es, daß die auf einer Geraden liegenden Strahler einen Abstand voneinander haben, der klein zur Wellenlänge ist. Die Strahler selbst sind ebenfalls klein zur Wellenlänge.


Die Darstellung des Schallfeldes wurde wie schon oben berichtet durchgeführt.

ZUSAMMENFASSUNG


Für den elektrischen Fall ergeben sich analoge Strahlungsfelder, wenn z.B. bei Dipolzeilen für elektromagnetische Wellen die Antenne vertikal angebracht wird und der Aufpunkt auf die horizontale Halbebene beschränkt wird.
SPACE APPLICATIONS OF ACOUSTIC FORCE

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INTRODUCTION
Non-zero time-averaged forces generated by acoustic standing waves may be utilized to position and manipulate samples such as liquid or molten droplets within a resonant chamber. This method has potential utility in the weightlessness environment of space.

THEORETICAL BASES
This simplified expression for the radiation pressure $\langle VP \rangle$ is given by

$$\langle VP \rangle = \left(P^2/2uc^2 \right) - \frac{1}{2} \rho \bar{u}^2,$$

where $P$ is the excess acoustic pressure, $\bar{u}$ is the gas particle velocity, and the bar over a quantity denotes the time average of the quantity. Eq. (1) is the Bernoulli equation, (ref. 1-3) which gives the acoustical perturbation on the ambient pressure from its quiescent value.

The pressure profile in our triaxial acoustic chamber can be derived as follows. The velocity potential $\Phi$ of the wave in the chamber can be expressed as

$$\Phi = \Phi_x \cos (k_x x) + \Phi_y \cos (k_y y) + \Phi_z \cos (k_z z),$$

where $\Phi_x, \Phi_y, \Phi_z$ are the complex velocity potential amplitudes of standing waves of frequency $\omega$ and wave number $k$. The particle velocity $\bar{u}$, by definition, is $\bar{u} = \omega \Phi$. The pressure is given by $P = -\rho \bar{u}^2$. Equation (1) and (2) lead to a radiation pressure nodal point at the center of the chamber.

EXPERIMENTAL TECHNIQUES
The laboratory prototypes of the triaxial acoustical levitation resonance chamber is nearly cubical, with inside dimensions of 11.43 x 11.43 x 12.70 cm, which are the $x$, $y$, and $z$ faces, respectively. During operation of the chamber, there is a tendency for introduced liquids and particles to be driven toward the nodes, where they collect and remain until excitation ceases.

RESULTS AND CONCLUSION
The application of acoustics in space technology has been proved to be successful in our laboratory. At the present time, NASA is considering to incorporate the acoustic chamber into the first United States and European joint space shuttle flight to accommodate not only JPL's experiments, but also the possible experiments proposed by the scientific community of United States and Europe.

REFERENCES
FIRST ORDER ACOUSTIC TORQUES AND RESULTING INDUCED SOLID BODY ROTATION

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INTRODUCTION

Non-zero time-averaged forces, torques and velocities in sound fields have always been associated with second order non-linear effects. We have observed a rather striking phenomenon which, for acoustic displacement amplitudes larger than the viscous penetration depth, is a first order effect.

THEORETICAL BASES

Consider a solid cylinder of radius \( r \), very small compared to \( \lambda \), free to rotate, with its axis in the \( z \) direction and located at \( x = y = 0 \) (see Fig. 1). When surrounded by a gas in which the particles are executing a circular motion of the same sense with velocity \( v \) and radius \( x_0 = \rho v_0 / \omega \), the cylinder will be dragged into rotation also, because of the gas viscosity (to the extent that the quadrupolar scattered acoustic field of the cylinder can be neglected).

One can argue that the torque, \( L \), on the cylinder of radius \( r \) should be given approximately, by

\[
L = \eta \left[ v_0 / \lambda - v_0 / (\lambda + x_0) \right] A \rho \quad (1)
\]

where \( A \) is the surface area of the cylinder inside the acoustic chamber. The physical ideas which prompt us to write Eq. 1 are (1) Appreciable motion of the fluid relative to the cylinder can only occur at distances from its surface which are equal to or exceed \( \lambda \).
(2) Because \( r > x_0 \), one can neglect the curvature of the cylinder in obtaining the viscous stress.
(3) The gradient in velocity normal to the cylindrical surface can be approximated by the fluid velocity divided by the distance from the surface.
(4) If the viscous force per unit area during the positive phase of the orbital fluid motion is \( \eta v_0 / x_0 \), then in this approximation it is \( \eta v_0 / (\lambda + x_0) \) during the opposite phase of the motion.

EXPERIMENTAL TECHNIQUES AND RESULTS

Figure 1 is a diagram of the experiment apparatus. It consists of a vertical cylindrical rod 1 in. in diameter, supported on an air bearing and passing through a box 4 x 4 x 5 in. which has two loudspeakers centered on adjoining 4 x 5 in. vertical sides. The amplitudes of the \( (1, 0, 0) \) and \( (0, 1, 0) \) modes are set equal and the phase difference of the two modes, \( \phi \) in Eq. 1 can be varied with a phase shifter. The frequency of the degenerate acoustic modes is 1.62 kHz and \( \lambda_0 \) is accordingly 5 x 10^{-3} cm.

Eq. 2 is the curve plotted in Fig. 2 and it is clear that a task of theory is to provide a torque which in some approximation yields Eq. 1.
V. Electroacoustics

O. Sound Systems
SOME CONSIDERATION OF ADVANCED STEREOPHONIC SOUND REPRODUCING SYSTEM

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INTRODUCTION

When human recognizes the sound image localization of the real sound which comes from anywhere around the listener, the most contributed factors are the level difference and time difference between listener's both ears, but in this theory, the common information in both earpaths is eliminated. This common transfer function is important especially for front and back discrimination in sound localization.

THEORETICAL BASES

1) To add common transfer function $H_g / H_0$ to block diagram at first.
2) To add circuit making both ears' sound pressure ratio.

First we define that SPL & phase ratio of listener's both ears by 1st speaker $P_1 / P_t$ and by 2nd speaker $P_2 / P_t$ and input level ratio of 2nd two speakers $E_2 / E_1 = H_2 / H_1 = A = a e^{j\theta}$ (1)

$P_2 / P_1 = H_0 / H_0 = B = b e^{-j\phi}$ (2)

$E_2 / E_1 = C = c e^{j\phi}$ (3)

SPL & phase ratio of listener's both ears are given by next equation (4) and

$P_2 / P_1 = (H_0 * H_2 + H_0 * H_1) / (H_0 * H_0 + H_0 * H_0)(1+C/B)/(1+C*B)... (4)$

Where the letter * is a convolution. Conditions (1) is equal to (4) are

$c = (a \cos \theta - b \sin \phi) + (a \sin \theta + b \cos \phi)(1 - a \cos \phi + b \sin \phi)(a + b)(a - b)... \phi... (5)$

$r = \tan (a \sin \theta - b \cos \phi)/(a \cos \theta - b \sin \phi) + \tan (a \sin \theta + b \cos \phi)/(a \cos \theta + b \sin \phi)... (6)$

$\phi = \tan (a \sin \theta - b \cos \phi)/(a \cos \theta - b \sin \phi) + \tan (a \sin \theta + b \cos \phi)/(a \cos \theta + b \sin \phi)... (7)$

These c and r are shown in Fig.2 and Fig.3.

EXPERIMENTAL TECHNIQUES

1. Sound image localization by the ratio of both ears is shown in Fig.4.
2. Sound image localization adding the transfer function difference between back and front characteristics to the experiment 1, is shown in Fig.5.

RESULTS AND CONCLUSIONS

As shown in Fig.2 and 3, characteristics of front and back are very close in lower freq, and we got Fig.4 by subjective experiment, but if we add the transfer function, we got a good result as shown in Fig.5.

REFERENCES

(1) K. Nakabayashi, NHK Technical Report (1975)18, P455 - P461
INVESTIGATION OF QUADRAPHONIC MICROPHONE SYSTEMS

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INTRODUCTION

The problem of the microphone systems to be used for quadraphonic recordings remains up to now open. No optimum performance system has yet been standardized. The here investigation has been based on the following two points:

1) In what kind of rooms will the reproduction take place?
2) What information must be transferred through the used microphone system, in connection to the reproduction room.

EXPERIMENTAL TECHNIQUES

In the consequence, four different microphone systems have been investigated. Two of them are in X-Y; X-Y configuration (coincidence), while the other two are in A-B-C-D configuration. (1)

Two sets of actual subjective tests have been performed. The first set has been made in order to investigate the directional characteristic of the systems the second has been in order to investigate the reproduction fidelity of the systems (with orchestral music). The people used for the test were students with ages varying between 23 to 25 years.

CONCLUSIONS

The test results can be summarized as following:
For reproduction in small to medium sized common rooms the suitable microphone system must be capable to transmit reverberation versus frequency plus sound energy distribution information from the recording room (concert hall). For reproduction in larger, suitably designed rooms the microphone system must be able to transmit information concerning mainly the frequency & directional characteristics of a large volume (orchestra) source. A similar coupled investigation concerning the reproduction loudspeakers will be reported soon in order to cover the subject from both sides.

REFERENCE

(1) Z. Zyszkowski, Podstawy Elektroakustyki (1965), 757
ÜBER DIE HOLOPHONISCHE SCHALLEMPFINDUNG

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EINLEITUNG


THEORIE

Die Schallsignalen werden im Nachbereich des Kopfes modifiziert. Die $S_1$ und $S_2$ Schallefelder bei der Ohren können als Hologramme aufgefasst werden, die sich im Hörsystem addieren / $S_0$.

Die gemischten Glieder der Rekonstruktion / $S_0^+$/ charakterisieren das Quellsignal.

ERGEBNISSE

Die typischen Fällen der Schallübertragung zeigen die Entstehung mehrerer Rekonstruktionen, die mit den Parametern der Übertragungsverhältnissen modifiziert sind. Eine Multiplikation der Übertragungskanäle lässt zusätzliche Rekonstruktionen entstehen, die das Informationsgehalt über die originale Schallquelle erhöhen, und die Störfaktoren der Übertragung Raumakustische Verhältnisse, nichtlineare Verzerrungen, tontechnische Entzerrungen, Übertragungsfehler der Lautsprechern, usw. erniedrigen.

REFERENZEN


CORRELATION BETWEEN SUBJECTIVE AND OBJECTIVE DATA FOR LOUDSPEAKERS

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INTRODUCTION

There is not enough knowledge about what kinds of objective characteristics affect sound quality of loudspeakers. It is necessary to investigate the relation between a subjective evaluation and objective data to reveal the characteristics which have a great influence on sound quality of loudspeakers.

EXPERIMENTS

In the subjective listening tests, seventeen listeners made preference judgments and similarity judgments on all possible pairs of nine hi-fi loudspeakers.

RESULTS AND CONCLUSIONS

The nonmetric factor analysis of preference data yielded the preference space of three factors. The first of them is interpreted as the "consensus preference" factor and the remaining two are "individual difference of preference" factors. As a result of nonmetric multidimensional scaling, similarity data are summed up in three psychological dimensions. Interpretation of these dimensions indicate that they represent "volume and extent", "brightness" and "beauty" respectively.

Measured data of sixteen objective parameters were rated from the viewpoint of high fidelity reproduction (1). Fig.1 shows these parameters which were fitted in the preference space as vectors. (a)-(f) represent scores for sound pressure response in anechoic room [(a) on axis; (b) 30° from axis; (c) sharp peaks and dips; (d) overall flatness; (e) total of (c) and (d); (f) frequency range], (g)-(i) represent diffusion of sound [(g) 30° from axis; (h) 60° from axis; (i) total], (j)-(l) represent harmonic distortion [(j) 2nd; (k) 3rd; (l) total], (m) represents transient distortion and (n)-(p) represent sound pressure response in the listening room [(n) peaks and dips; (o) overall flatness; (p) total]. Rating scores of sound pressure responses in the listening room [(o) and (p)], and in an anechoic room [(b), (d) and (e)], have high correlation with the "consensus preference" factor.

Similarity of response patterns among loudspeakers was calculated for each objective characteristic and it was subjected to multidimensional scaling. The configurations of loudspeakers based on objective similarity were compared with the ones based on subjective similarity. Similarity of sound pressure response in the listening room has close accordance with subjective similarity.

REFERENCE


Fig.1. Correlation of objective parameters with subjective preference.
A SCALE MODEL INVESTIGATION OF REVERBERATION RE-INFORCEMENT BY ELECTRO-ACOUSTIC COUPLING

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INTRODUCTION

A reverberation re-inforcement system utilizing the acoustical properties of two coupled spaces has recently been described by Jones and Fowweather (1). This system was apparently developed empirically and has not been theoretically analysed.

THEORY

The following energy balance equations can be written for two "live" coupled spaces featuring an electro-acoustic feedback loop:

\[ W + \frac{E_{1}SC_{0}}{4} + kE_{2} = \frac{dE_{1}}{dt} + \frac{E_{1}A_{2}C_{0}}{4} + \frac{E_{1}SC_{0}}{4} \]  

\[ \frac{E_{1}SC_{0}}{4} = \frac{dE_{2}}{dt} + \frac{E_{2}A_{2}C_{0}}{4} + \frac{E_{2}SC_{0}}{4} \]  

where \( E \), \( dE/dt \), \( W \) and \( A \) refer to the acoustic energy density, its time derivative, the volume of the enclosure and the total absorption within an enclosure, respectively. The subscript 1 refers to the primary enclosure and 2 to the secondary enclosure. \( W \) is the acoustic power of the source in the primary enclosure, \( C_{0} \) is the velocity of sound and the constant \( k \) is determined by the properties of the feedback loop which consists of a microphone positioned in the secondary enclosure driving a series of loudspeakers positioned in the secondary enclosure via power amplifiers. The degree of reverberation re-inforcement is determined by the magnitude of \( k \), i.e. the gain of the feedback system. These equations are capable of solution by simple numerical methods.

RESULTS AND CONCLUSIONS

Figure 1 shows a plot of calculated values of effective reverberation time versus the values measured in the scale model. It can be seen that the agreement between the two is good.

REFERENCE

(1) M. H. Jones & P. Fowweather, Acustica (1972), 21, 357

Fig. 1 Comparison of measured and calculated values
INTRODUCTION

Most of the experimental work on speech intelligibility and other fields of subjective acoustics has been done using monaural signals as stimuli. In our work we have reached a point where a device capable of synthesizing a realistic sound field with easily controlled parameters has become an indispensable tool. Accordingly we have built an electronic delay line and a mixing console, both of which can be easily programmed using plug-in cards.

DESCRIPTION OF EQUIPMENT

After converting the audio signal into digital form, it is fed to a shift register delay line, which consists of two sections. The first section has one tap with 5 ms delay increments up to 75 ms; the second part has ten taps with 1 ms increments up to a maximum of 127 ms. The amount of delay of each tap is determined by the setting of thumbwheel switches or one of three plug-in program boards. Each of the delayed digital signals is converted into an analog signal to feed a programmable mixing console. The mixers are voltage controlled attenuators, that can be controlled either manually by matrix slide switches or by program boards. There are 10 outputs to feed 10 loudspeakers.

APPLICATIONS

Numerous applications come into mind; in our university it is our intention to investigate the following subjects in the immediate future:

1. The relation between speech intelligibility and modulation transfer function
2. The subjective effect of splitting an echo or late reflection into many small components

In addition it can be pointed out that a much simplified version may find domestic application as a more sensible alternative to quadraphony. Using a simulator, stereophonic hardware and software will not become obsolete, and a more realistic sense of spaciousness can be achieved.
ELEKTRO-AKUSTISCHER ELEKTRET-SCHLITZWANDLER

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EINFUHRUNG

In dem von O. Jefimenko (1) zuerst vorgeschlagenen Elektret-Schlitzwandler besteht ein linearer Zusammenhang zwischen der an dem Tandem-Kondensator angelegten Spannung \( U \) und der auf den Elektreten ausgeübten Kraft \( K = p \cdot S \). Diese Eigenschaft erlaubt es, einen elektrostatischen elektroakustischen Wandler mit streng linearer Zusammenhang zwischen elektrischer und akustischer Amplitude zu entwerfen.

THEORETISCHE GRUNDLAGEN

Der grundsätzliche Aufbau eines elektroakustischen Elektret-Schlitzwandlers besteht aus zwei parallel zueinander angeordneten und entgegengesetzt gepolten Plattenkondensatoren mit einem Elektriten zwischen den Kondensatorplatten, der sich in \( x \)-Richtung parallel zu den Kondensatorplatten bewegen kann. Der Elektret ist mit der Membran fest verbunden. Bei Bewegung des Elektretens in \( x \)-Richtung ändern sich die Ladungen der beiden Kondensatoren und bewirken eine Spannungsänderung bzw. einen Strom an dem angeschlossenen elektrischen Tor. Die Rechnung ergibt für die Zweitor-P-Matrix-Gleichung des elektroakustischen Elektret-Schlitzwandlers

\[
\begin{pmatrix}
U \\
V
\end{pmatrix} =
\begin{pmatrix}
j \omega C_S S G_S & -\psi \\
\psi & Z_m
\end{pmatrix}
\begin{pmatrix}
U \\
V
\end{pmatrix}
\]

mit

\( \psi = 2 \cdot b \cdot q \cdot \sigma \): elektroakustischer Kopplungsfaktor
\( b \): Breite des Elektretens
\( q = (d_1 / e) / (d_2 / e) + (d / e) \): Füllfaktoren der Kondensatoren
\( d \): Dicke des Elektretens
\( \epsilon \): absolute DK des Elektretens
\( C_s \): Kapazität des Schlitzwandlers mit Dielektrikum
\( G_s \): Ohmscher Leitwert des Schlitzwandlers
\( Z_m \): mechanische Impedanz des schwingenden Systems
\( v^m \): Schallschnelle

VERSUCHE

Es werden ein Mikrophon und ein Lautsprecher beschrieben, die mit einem Elektret-Schlitzwandler aufgebaut sind, und die aus der Zweitor-Koeffizientenmatrix berechenbaren Werte für Mikrophon-Empfindlichkeit und Lautsprecher-Wirkungsgrad werden mit den experimentell ermittelten Daten verglichen.

REFERENCES

INTRODUCTION AND THEORETICAL BASES

In this paper, the optimization technique is used in order to obtain a design parameters vector, which realizes a flat response, of a phase inverter speaker system. This technique is superior to the former technique by the network synthesis theory.

Sound pressure transfer function of this speaker system is [1]: (see Fig. 1)

\[
\frac{m_{\text{ap}} w_{\text{ap}}}{F_p} = \beta^3 + \beta^2 \left( \frac{1}{\beta \omega_p} + \frac{1}{\beta \omega_q} \right) + \beta \left( \frac{1}{\beta \omega_p^2} + \frac{1}{\beta \omega_q^2} \right) + \frac{1}{\beta \omega_p^3} + \frac{1}{\beta \omega_q^3}
\]

where \( \beta = j \omega_p / \omega_q \), \( \omega_p = 1/2 \nu \), \( \omega_q = \nu / \nu \), \( \nu = \omega_p / \omega_q \)

\[\beta_0 = M_p / M_0 \quad \beta_1 = \omega_p / \omega_q \quad \beta_2 = 1 / \beta_0 \]

We define a design parameters vector as follows:

\[ X^T = ( \beta_0, \beta_1, \beta_2, \nu, \nu ) \]

Then the sound pressure frequency response \( H(\omega, \alpha) \) is

\[ H(\omega, \alpha) = 20 \log_{10} \left| \frac{F(\omega, \alpha)}{m_{\text{ap}} w_{\text{ap}}} \right| \]

Now we define evaluation function as follows:

\[ F(\omega, \alpha) = \frac{\sum_{i=1}^{L} \alpha_i \left( \beta^3 + \beta^2 \left( \frac{1}{\beta \omega_p} + \frac{1}{\beta \omega_q} \right) + \beta \left( \frac{1}{\beta \omega_p^2} + \frac{1}{\beta \omega_q^2} \right) + \frac{1}{\beta \omega_p^3} + \frac{1}{\beta \omega_q^3} \right) - H_{\text{des}}}{H_{\text{des}}} \]

where \( L \) = number of frequency points evaluated, \( \alpha_i \) = weighting coefficient, \( \alpha \) = positive parameter for realizing the Fig. 1. Mechanical equivalent circuit.

By minimization of \( \frac{\sum_{i=1}^{L} \alpha_i \left( \beta^3 + \beta^2 \left( \frac{1}{\beta \omega_p} + \frac{1}{\beta \omega_q} \right) + \beta \left( \frac{1}{\beta \omega_p^2} + \frac{1}{\beta \omega_q^2} \right) + \frac{1}{\beta \omega_p^3} + \frac{1}{\beta \omega_q^3} \right) - H_{\text{des}}}{H_{\text{des}}} \)

RESULTS AND CONCLUSION

An example of computation result is shown in Fig. 2 with constraints on \( X \) which are determined by physical quantities of speaker, box and absorbing material.

This optimization technique has the following advantages, which are not obtained by the network synthesis technique. (1) This technique is effective even when \( Q_p \) is finite. (2) Constraints on the design parameters can be considered in the design process. (3) This technique is free from the type of flat responses defined by the network synthesis theory, therefore large extension of low frequency response can be obtained.

REFERENCES


INTRODUCTION

A computation method is shown to calculate the sensitivity vs. frequency curve of horn loudspeakers having given geometrical and material data. The simulation of loudspeaker is carried out by means of the running of a FORTRAN IV program describing this calculation method. Most of the trial-and-error work is performed by the computer.

THEORETICAL BASES

The problem may be split into three parts: discussion of the horn, the pressure chamber, and the dynamic transducer.

In the horn, one-parameter wavefront propagation is assumed and Webster's differential equation /1/ is applied taking its validity limits into consideration. Exponential horn profile is supposed but the method is suitable for other profile shapes as well. The mouth of the finite horn is loaded by the radiation impedance, which is modelled either by the rigid round or rectangular piston acting in an infinite baffle or by a rigid piston vibrating in the end of an infinite tube. The type of these approximations depend on the purpose of the loudspeaker. The throat impedance of the horn is determined by Olson's formula /2/.

The transformation effect and the capacitance of the pressure chamber are taken into account, but it is assumed that the placement of the throat hole is optimum to reject vibration modes according to Smith's formula /3/. All calculations are performed at low intensity levels so the nonlinear phenomenon is left out.

The diaphragm of the dynamic transducer is regarded as rigid and the analogous electrical network is applied. At the determination of the sensitivity it is supposed that the loudspeaker is fed by a voltage generator /4/. The directivity index is combined into the equations according to the radiation impedance applied.

RESULTS AND CONCLUSIONS

The input data, for the curve in Fig.1, were as follows: ø mouth=0.08m, flare=44 1/m, area of throat=1.13×10⁻³ m², vol. of pres. chamber=4.2×10⁻³ m³, eff. area of diaphragm=1×10⁻³ m², mass of moving system=4.5×10⁻⁴ kg, compl. of diaphragm=2.1×10⁻¹ m/N, loss resistance=1.9 ohm, B=5.3 Vs/m, E=13 ohm, L=2.1×10⁻⁸ H. This method is suitable to predict the characteristic sensitivity of a horn speaker to be developed. The modification of input data can be done /maybe in several steps/ before completion of the first sample.

REFERENCES

/2/ Olson H.F, J.A.S.A. /1931/April,485.
BANDWIDTH-EFFICIENCY PRODUCT OF HORN LOUDSPEAKERS

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INTRODUCTION

The known expressions of the efficiency of horn loudspeakers /see e.g. /1// are suitable for calculations concerning a particular loudspeaker, however, these are unadapted for drawing general conclusions. This paper presents relationship between the efficiency, bandwidth and some basic design data of moving coil horn loudspeakers.

METHOD

The electrical equivalent circuit of horn loudspeaker driven by a source of constant voltage \( u \) is shown in Fig. 1. \( R_\text{c} \) is the electrical resistance of the voice coil; \( B \) is the magnetic flux density in the air gap; \( l \) is the length of the electrical conductor of the coil; \( m \) is the sum of the masses of the moving coil and the diaphragm; \( c \) is the compliance of the suspension; \( r \) is the sum of the mechanical resistance due to the air loading i. e. to the radiation, and the mechanical loss resistance. Now let us consider the last to be negligible.

The maximum of the efficiency occurs at the antiresonant frequency of the parallel-resonant circuit. Then the electrical impedance of the loudspeaker is pure real and the efficiency is equal to the ratio of the radiated power to the input power i. e.:

\[
e_\text{r}=\frac{1}{1+\left(B_l/R_r\right)^2}
\]

The bandwidth of half power points is given by the product of the capacity and the resultant resistance of the circuit

\[
\Delta\omega=\frac{1}{R_\text{c}+R_l/B_l^2}
\]

This means that one should realize a mechanical loading resistance \( r \) to achieve a prescribed bandwidth as follows

\[
r=\frac{1}{R_\text{c}/\Delta\omega R_\text{c} m}/B_l^2
\]

If we use it in Eq. 1 we obtain bandwidth-efficiency product in a fairly simple form provided that \( m \) is approximately equal to the mass of the voice coil

\[
\Delta\omega \cdot e_{\text{r},\text{max}}^2=\frac{E^2}{\rho f}
\]

where \( \rho \) is the specific resistance and \( f \) is the density of the material of the voice coil.

REFERENCES

Le but du travail est d'obtenir numériquement les variations, en fonction de la fréquence, des impédances mécaniques de haut-parleurs électrodynamiques, impédances intervenant dans un modèle théorique [1] [2], et d'en déduire les valeurs numériques des différentes constantes mécaniques associées à chacun des éléments de ces haut-parleurs.

Cette étude repose sur un modèle précis que nous avons proposé pour les haut-parleurs et qui conduit, dans une analogie électrique de type admittance pour l'impédance motionnelle, à un quadripôle en n faisant intervenir trois impédances mécaniques. Ce modèle, dont le domaine de validité s'étend jusqu'à l'apparition des lignes nodales circulaires, tient compte en particulier de la déformation du diaphragme.

Le nombre important de paramètres qui interviennent dans le problème, trois impédances mécaniques (donc six paramètres) et neuf constantes mécaniques (dont dépendent les trois impédances), et l'obligation de calculer les dernières à partir des variations des précédentes en fonction de la fréquence, a nécessité l'emploi de méthodes mathématiques permettant, à partir d'un nombre important de points expérimentaux, de trouver les paramètres avec la précision qu'autorisent les lois des grands nombres.

Dans ces conditions, à chaque valeur de la fréquence, il était nécessaire de pouvoir faire varier l'impédance de rayonnement pour ainsi accéder aux impédances mécaniques du système par une méthode des moindres carrés; c'est la raison pour laquelle cette première étude a été faite en chargeant le haut-parleur par la colonne d'air (de même section que celle du diaphragme) contenue dans un tuyau sonore fermé par un piston rigide, dont l'impédance d'entrée, outre qu'elle possède une expression analytique bien définie, présente des variations importantes en fonction de la longueur du tuyau.

C'est alors que l'étude des variations des impédances mécaniques ainsi obtenues en fonction de la fréquence par une méthode des moindres carrés donne les valeurs des constantes mécaniques du système.

Les méthodes proposées et les résultats qu'elles procurent présentent un double intérêt : d'une part elles permettent d'avoir une description globale (en termes d'impédances mécaniques) mais précise d'un haut-parleur en tenant compte de l'impédance de rayonnement à l'arrière et sans faire aucune hypothèse particulière sur l'impédance de rayonnement à l'avant, et d'autre part elles donnent une description détaillée du système (en termes de constantes mécaniques) dans des conditions normales de fonctionnement.

REFERENCES

INTRODUCTION

Examining an acoustic field produced by a loudspeaker with a conical membrane, possibilities of separating in the acoustic field areas of a far and a near field were considered.

THEORETICAL BASIS

Acoustic pressure in any point, produced by a flat membrane mounted in a rigid baffle and radiating into the semispace, can be determined on the strength of dependences deduced from the general wave equation. These dependences are subject to substantial simplifications under designated conditions in the acoustic field, the so-called far field. These conditions may be formulated as follows:

\[ R > \frac{\lambda}{2} \]

\[ R > \frac{A}{\lambda} \]

Condition /1/ is to be regarded as essential. Moreover dependence of the far field from /condition 1b/ requires additional precision of definition, when considering the pulse signal radiations for which the wavelength can not be exactly measured. For this purpose a new formula was set up defining the limit value of the correlation factor at the measurement point, separating the areas of the far and near field.

METHODOLOGY

A loudspeaker with the conical membrane mounted in an enclosed cabinet was exposed to vibrations of a sinusoidal signal in a broad frequency band as well as to other different pulse signals in succession. The sound pressure level was recorded at measurement points located along the loudspeaker axis.

RESEARCH RESULTS AND CONCLUSION

On the basis of changes of sound pressure level as a function of the distance of a measurement point from the loudspeaker plane, variations of sound pressure level in an acoustic field of the loudspeaker excited by the different signals, have been determined. As regards the remaining pulse signals, the differences in the distribution of pressure are very clear: they depend on the signal spectrum. The determination of the area of a far and near field in an acoustic field of the loudspeaker, which the loudspeaker radiating the Tone Burst can be based on the formula /1/. For pulses with a complex spectrum the above-mentioned correlating formula should be used. In the problem under consideration the areas of near and far fields had been separated at measurement points, at which the correlation factor \( T_g \geq 68\% \).

REFERENCES

[2] Hojaj E., Analysis and principles of perception of signals with changing structure in a complex field /now printing/
NEW PHYSICAL MEASURES FOR LOUDSPEAKER ASSESSMENT

Tannaka, Y. (Consumer Products Research Center, Hitachi, Ltd, Yokohama, Japan)
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INTRODUCTION

In general, sound pressure frequency responses of loudspeaker systems measured in home listening rooms diverge with the sound pressure frequency response measured in an anechoic chamber or the sound power frequency response in a reverberant chamber.

In order to investigate the correlation between frequency responses in home listening rooms and free field characteristics of the loudspeaker systems, two new physical measures of the loudspeaker system, the partial space power response and the dispersion index, are proposed in this paper.

DEFINITION

The measuring method of the sound power response in an anechoic chamber provides partial response characteristics for a partial region, standing at solid angle $\Omega$, on a hypothetical sphere surrounding a source.

The partial space power response $E_n$ is defined as follows.

$$E_n = 10\log <P>_n + 10\log S_n \quad (1)$$

Where $S_n$ is partial area on the sphere determined by solid angle $\Omega$ and $<P>_n$ is average sound pressure response on $S_n$.

The dispersion index in this study is defined as follows.

$$D_n = E_{4\pi} - E_n \quad (2)$$

Where $E_{4\pi}$ is the total space power response and $E_n$ is the partial space power response on $S_n$.

EXPERIMENTS

$E_n$ and $D_n$ versus solid angle $\Omega$ were measured on three loudspeaker systems which have different dispersion characteristics in middle and high frequency regions. These systems were alternately placed in various kind of home listening rooms and space-average sound pressure responses inside a sphere of which diameter is about 1 m were measured at a typical listening position. $E_n$ was compared with the space-average sound pressure response, taking $\Omega$ and $D_n$ for parameters.

RESULTS AND CONCLUSIONS

The partial space power response $E_n$ proposed corresponds well with the sound pressure frequency response averaged out from all curves obtained for all the rooms. The specific solid angle $\Omega$, where the $E_n$ matches the sound pressure response best, depends on the dispersion index $D_n$ of the pertinent loudspeaker system.

Strong correlations are observed between new physical measures mentioned and values of the subjective sound quality evaluations.

REFERENCES

(1) Arnold P.G. Peterson and Ervin E. Gross, Jr., Handbook of NOISE MEASUREMENT, General Radio Company (1972)
UNE NOUVELLE MÉTHODE POUR L'ÉVALUATION DE LA QUALITÉ DES SOURCES SONORES ARTIFICIELLES

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INTRODUCTION

Normalement, l'on évalue la qualité d'un haut-parleur sur la base de quelques paramètres objectifs, fournis par le constructeur, qui permettent de classifier la source par rapport à une échelle de valeurs purement conventionnelle. La connaissance de ces caractéristiques, telles que l'amplitude de bande, la réponse en fréquence, les relations de phase, la réponse aux transitoires, etc. ne suffit pas, en effet, à définir d'une façon complète, le comportement de l'haut-parleur du point de vue de la qualité de la reproduction du son; cette propriété peut être évaluée seulement à l'aide d'essais d'écoute direct. Le jugement sur la qualité d'une source sonore est lié, en général, à l'écoute de morceaux musicaux de genre différent, diffusés dans une salle acoustique particulière. Nous avons songé qu'on pourrait obtenir des informations plus détaillées sur la fidélité de reproduction d'une source artificielle en adoptant une méthode de mesure plus rigoureuse, basée sur la comparaison directe entre la source naturelle et la source artificielle. Pour ces essais nous avons utilisé, jusqu'à ici, comme signal, la voix humaine; nous songeons, toutefois, d'étendre éventuellement nos essais en utilisant des signaux musicaux.

TECHNIQUE EXPERIMENTALE

Les essais subjectifs, effectués à l'aide de haut-parleurs de genre différent, sont caractérisés par l'écoute d'une liste de phrases prononcées, d'une façon aléatoire, par le locuteur ou par la source artificielle. Les auditeurs, sans voir les sources, doivent juger si le signal perçu est celui naturel ou bien celui reproduit par l'haut-parleur. On effectue ces déterminations en conditions d'écoute différentes, caractérisées par la présence ou l'absence d'un bruit masquant. On pourra, par conséquent, établir une échelle quantitative d'évaluation des sources en fonction du rapport S/B, dont la valeur entraîne une pourcentage donnée de discrimination entre la source naturelle et celle artificielle.

Quelques essais préliminaires, effectués avec trois sources qualitativement très différentes, ont fourni des résultats qui permettent de prévoir de bonnes possibilités d'application de notre méthode pour la solution du problème proposé.
INTRODUCTION

La directivité sonore est particulièrement intéressante en acoustique pour un certain nombre d'applications (1). Nous nous attacherons ici à l'étude de sources bipolaires directives et de leur commande par une analyse fréquentielle à partir du calcul transformationnel.

ETUDE THÉORIQUE

Nous ferons l'hypothèse dans ce qui suit, que les sources sonores utilisées s'identifient à des systèmes linéaires invariants et initialement au repos ce qui revient à s'intéresser uniquement aux parties gouvernables et observables de ces systèmes.

Considérons (Fig.1) deux sources ponctuelles S₁ et S₂ qui constituent un système multivariable dont la fonction du transfert dans le domaine de Laplace (variable s) est donnée par la relation :

\[ [Y(s)] = [H(s)] [U(s)] \]

où [Y(s)] représente la matrice des signaux de sortie (tensions de commande) et [H(s)] la matrice de transfert.

La condition nécessaire et suffisante de directivité idéale s'exprime par l'annulation de la variable d'émission \( Y_1(s) \) (ou \( Y_2(s) \) suivant la direction choisie) ce qui revient à la relation matricielle :

\[ [Y_1(s)] = [H_{11}(s)] [U_1(s)] \]

La gouvernabilité du système se réduit alors à une relation linéaire entre les variables de commande :

\[ 0 = H_{21}(s) . U_1(s) + H_{22}(s) . U_2(s). \]

RESULTATS EXPERIMENTAUX

Le problème de commande des sources S₁ et S₂ revient à l'identification de la matrice de transfert [\( H_{ij}(s) \)].

La Figure 2 montre un exemple concret d'identification dans le cas de sources sonores rigoureusement identiques constituées par des haut-parleurs électrodynamiques (\( H_{ij}(s) = H_{ij}(s) \)).

CONCLUSION

La réalisation de sources sonores directives et en particulier de bipôles pose un problème d'observabilité et de gouvernabilité que le formalisme automatique permet de résoudre. La commande de telles sources peut se faire, dans la majorité des cas, à l'aide de filtres analogiques simples dont la relation (3) permet de faire la synthèse.

REFERENCE

INTRODUCTION

To calculate cardioid transducers, so far, lumped - constant analyses has been used. A new method was developed to carry out these calculations. The components which cause phase shift are taken into consideration by wave guides. A few usual arrangements /e.g.: Fig.1./ has been calculated by this method and by the well known method and results have been compared.

THEORETICAL BASES

The elements which cause phase shift and have large dimension in the direction of wave propagation are treated as wave guides. The solution of wave equation is known at these boundary conditions. These results and those ones which were taken from the network theory have been combined. It has already been proved that the equations describing the performance of cardioid microphones and loudspeakers have a common function interpreting the directivity [3]. It has also been proved that this function is determined by the ratio of volume velocities of the sound gates /I = w1/w2/. Results are shown by ratio I of volume velocities /w2/w1/ belonging to 2 and 1 sound gates.

RESULTS AND CONCLUSIONS

At cardioid directivity total rejection can be achieved if the ratio of volume velocities of the sound gates satisfies the equation as follows [3]:

I' = -exp j88T = -cos88T - j sin88T

where T = d/c is the sound path and c is the sound velocity in air. If lumped - constant RC or LR type phase shifters are assumed and the ratio I' is written according to [3] the result is the first two terms of the series of exp j88T. For more exact approx. of this function equalizing members /additional LC elements/ are suggested in [1], [2]. Writing the ratio of volume vel. with these elements the first three terms of the exp. function is resulted:

I' = -1 -j88T + j88T2. With the described distributed parameters concept and if other required conditions are also fulfilled the ratio of volume velocities can be expressed: I' = -cos88T - j sin88T. By this equation it has been pointed out that it is not necessary to apply additional acoustical elements for a more precise approx. of function exp j88T because the function is automatically satisfied e.g. by the classical arrangement of Fig. 1 too.

REFERENCES

ELECTROMAGNETIC TRANSDUCER FOR ELECTRONIC TELEPHONE
RINGER USE

Ceruti, R.  
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INTRODUCTION

The use of integrated circuits in telephone sets allows the conventional electromechanic ringers to be replaced by electronic ones. So it's possible to reduce the calling signal level and to obtain better acoustic signals with respect to pleasantness and perceptibility.

Electronic ringers, at present implemented, generate sounds with over 2 kHz fundamental frequencies. In fact they use telephone transducers not specifically designed but with resonance frequency at about 2 kHz.

On the contrary the transducer in the following description was designed by taking into account the characteristics of the sound to be generated.

TRANSDUCER DESCRIPTION

The transducer, shown in fig. 1 and fig.2, meets the following requirements: 1) high electric impedance (2.5 ± 3 kΩ); 2) response curve suitably shaped with the view of getting the maximum efficiency with the associated circuit excitation (fig.3); 3) suitable size with respect to the telephone dimensions.

As shown in fig. 3, 500 Hz and 1000 Hz exciting signal characteristic frequencies have been magnified.

At the same time three different easily integrable electronic circuits were designed. Their calling signal detection characteristics are: a) \( \geq 30 \, V_{RMS} \), 25 Hz; b) \(-10 \, \text{dBm}, \, 450 \leq 500 \, \text{Hz} \); c) both a) and b) types.

The produced acoustic signal, consisting of a 20 Hz amplitude modulated 500 Hz frequency, was determined on the ground of purposely made subjective tests.

REFERENCES

(1) F.L.Crutchfield, J.R.Power; Bell Lab. Record, Febr. 1957  
(2) Attia

(2) Atia, Gale, Seneer; IEEE Trans. on Communications, vol. COM-20, No. 1, Febr. 1972
INTRODUCTION

The increasingly visible need in recent years for teleconferencing (TC) has stimulated an effort at BNR to design an audio-TC model terminal. A survey of potential TC locations gave useful information about the acoustic environment in which a terminal will operate (1). Accordingly, a model terminal has been designed to provide an efficient interface between the telephone network on one side and the talker and acoustic environment on the other.

DESIGN BACKGROUND

The parameters of the modal size of a conference room (1), the sound levels required by a listener and produced by a talker for satisfactory communication, as well as the specified ranges of transmitted and received signals on a telephone line, were used together with previous experience in TC (2) to define the required sensitivities of the TC-terminal in both directions. Appropriate transducers were then chosen and mounted, around the same axis of symmetry, in one cylindrical housing of dimensions fulfilling human factors as well as acoustical stability requirements. The electronics are a modification of the circuitry in the "Companion" or "Logic-Handsfree" loud-speaking attachments, which is characterised by a voice-switch (3), unnoticeable in a disciplined conversation.

RESULTS AND CONCLUSIONS

The BNR audio-TC-model terminal has been designed to operate efficiently with groups of up to 20 participants or teleclasses of up to 30 students. It gives adequate sound power and transmitted signal with listeners or talkers at a distance of up to 3 m, irrespective of the angle of incidence (no dead zones). A tone control (treble boost or cut) assures a good receive quality for a variety of subscriber loops. For a noisy environment or for the terminal connected to a public-address system in an auditorium, external close-talking microphones are provided. An ear-phone with a dedicated volume control connected between the telephone line and the voice switch enables the chairman to listen to all signals on the line. A tape-recorder jack provides a documentation and editing facility. These features boost the efficiency of interaction between teleconferencing groups.

The terminal is aesthetically pleasing and portable (less than 9 Kg). Experience, so far, has indicated favourable user reaction.

REFERENCES

(3) W.E. Clarke and J. Gale, Telesis, Fall 1973, pp. 79.
V. Electroacoustics

P. MUSICAL ACOUSTICS
INTRODUCTION

This paper describes a bibliographic project whose aim is to provide a broad index to the literature and invention, both the well-known and the obscure, arising from the collaboration between science and music.

It will include references to monographs, to passages in monographs, to periodical articles, to patent specifications, to passages in works of speculative fiction and to a few works of surreal art. For the most part, these references will be to literature originally written in English, French, German, Italian or Latin from the 16th to the 20th centuries. For the early period, because of the manageable size of the literature, the index will be fairly inclusive, embracing as well much peripherally related material in both music theory and general acoustics.

TOPICS SURVEYED

The project surveys the collaboration between music and almost every major branch of science. Music has to do with MATHEMATICS, for example, in tuning and temperament, in probabilistic techniques of composition, and in musical cryptography; with ASTRONOMY, in the harmony of the spheres; with PHYSICS, in sound production from vibrating strings, air-columns, plates and membranes, in the analysis of musical tones, and in the design of musical instruments (including unusual ones, e.g. those of glass); with CHEMISTRY, in the chemical harmonica (or flame organ, etc.), in the law of octaves, and in the analysis of materials such as violin varnish, and the metals of gongs; with GEOLOGY, in the singing sands and stones, and in instruments like Till's "rockophone"; with METEOROLOGY, in the weather harp, in prediction of the weather from humming telegraph and telephone wires, and in the effect of ringing bells on thunderclouds; with BIOLOGY, in the study of the 'music' of insects, birds, whales, porpoises, etc.; with PHYSIOLOGY, in the playing of musical instruments, and in singing; with MEDICINE, in music therapy; with PSYCHOLOGY, in musical perception, including the mechanisms of pitch discrimination, beats, consonance and synaesthesia; with ARCHITECTURE, in echoes, whispering galleries, and the acoustics of concert halls; and with TECHNOLOGY, in electronic musical instruments, and in the recording and reproduction of music, by the early mechanical methods as well as the later electronic ones.

SCHEDULE

The project was begun in the summer of 1972. By early 1974, 2000 20th century references had been recorded. By late 1974, the 19th century literature had been surveyed in outline. In early 1975, work was begun on an intensive survey of the literature from the 16th to the 18th centuries. By late 1976, most inquiries about the project had so far concerned only a few of the topics being surveyed. It was then decided that the first publication resulting from the project should be a preliminary check-list on those few topics (tuning and temperament and also the scientific study of musical instruments) from the 16th to the 20th centuries. This check-list, which is being prepared in machine-readable form, will contain an estimated 1500 references and should be ready for publication as a book in 1978. Eventually, probably in 1980, the first volume of a detailed survey of all topics, covering the 16th to 18th centuries, will be published. It will have annotations in English for each item, and there will be a series of essays tracing the history for each topic surveyed. Later volumes will provide similar surveys for the 19th and 20th centuries.
SUR LA NATURE DU PHÉNOMÈNE MUSICAL

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En toute matière scientifique, la première condition à remplir est de bien connaître, au moins en première analyse, la nature du phénomène dont on s'occupe. Est-ce le cas du phénomène musical ?

NATURE COMMUNE APPARENTE DES DIFFÉRENTS ARTS DES sons

Historiquement nous avons connu, depuis la musique traditionnelle, successivement ce qu'on a appelé la musique polytonale, puis atonale, puis la musique concrète, plus récemment d'autres catégories d'arts sonores plus ou moins comparables. Si un musicologue analyse des œuvres appartenant à deux ou plusieurs de ces catégories, il peut trouver, sous les différences, des similitudes remarquables telles que reprise des thèmes, construction analogue de la composition, et on peut croire trouver là la nature commune, musicale, de ces différents arts des sons. Cependant, on peut constater que ces points communs appartiennent aussi aux autres arts. Il ne s'agit donc pas de la nature de la musique, mais de celle de l'art.

LA NATURE DE LA MUSIQUE TRADITIONNELLE

Au lieu de chercher la nature commune des différents arts des sons avant de savoir si elle existe, mieux vaut d'abord bien dégager la nature ou les caractères essentiels de la musique traditionnelle.

La nature de la musique traditionnelle est liée à l'utilisation, parmi tous les sons ou bruits existants, d'une échelle de fréquences discontinue, et à l'irrégularité de l'échelle de base ou diatonique ; et elle réside dans l'ensemble des liens qui rattachent les sons dans l'harmonie musicale, cet ensemble se synthétisant dans la tonalité. (Plus précisément ces liens sont les psycharithmes. Ils sont mis par le cerveau entre les sons reçus de l'oreille, et c'est par ce phénomène que les sons deviennent musique.)

NATURE DIFFÉRENTE DES AUTRES ARTS DES sons

Indépendamment de la théorie des psycharithmes, il est dès lors net que ni la musique atonale, ni la musique concrète, ni les musiques plus récentes plus ou moins assimilables, n'ont la nature de la musique traditionnelle. (Ceci n'est pas un jugement de valeur, mais seulement la constatation d'une différence de nature en dépit de l'usage commun des sons.)

LES TROIS ATTITUDES POSSIBLES VIS-À-VIS DE LA MUSIQUE ET DE L'ART

On peut avoir trois attitudes : 1) Ou bien on ne se pose pas de question d'ordre scientifique. Le seul guide alors est l'intuition créatrice. Cette attitude est en général celle de tous les grands créateurs du passé. 2) Ou bien on réfléchit - mais sans aller au fond sur la forme ou les règles de l'art qu'on pratique, et on imagine de les modifier. 3) Ou bien on adopte la stricte attitude scientifique. Elle comporte : l'approfondissement du phénomène ; l'absence d'une attitude normative devant l'art comme devant tous les phénomènes de la nature.

Du point de vue de l'art, la deuxième attitude est aléatoire ; du point de vue de la science, elle est éminemment insatisfaisante.
SYNTHESE RAPIDE DE SONS MUSICAUX

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INTRODUCTION

La musique par ordinateur a encore une diffusion très restreinte. Une des raisons principales est le coût élevé des calculs à effectuer, qui interdit une quantité d'essais suffisante. Les méthodes décrites ici sont très performantes et utilisent des fonctions calculables par récurrence.

BASES THÉORIQUES

La relation de récurrence (R)

\[ U_{n+1} = 2 \cos \phi U_n - U_{n-1} \]

admet pour terme général

\[ U_{n+1} = A \sin (n\phi + \varphi) \]

Si on pose \( \phi = 2\pi f/F \), où \( F \) est la fréquence d'échantillonnage et \( f \) la fréquence de l'onde acoustique à reproduire, le terme général \( U_{n+1} \) représente la valeur sinusoidale à la date \( n+1/F \).

Ainsi connaissant deux points \( U_n \) et \( U_{n-1} \) on pourra calculer le suivant avec une seule multiplication. De plus, il suffit de trois zones de mémoire pour effectuer cette succession de calculs.

Considérons maintenant les fonctions du type

\[ f(a, b, U) = \frac{A \sin (a + \varphi_1)}{1 - U \sin (a + \varphi_2)} \]

[de préférence à d'autres non récurrentes (1)]

où \( \varphi_1 \) et \( \varphi_2 \) sont des phases et \( U \) un paramètre discret inférieur à 1.

Cette fonction possède un spectre fonction de \( U \) dont l'harmonicité dépend du rapport \( a/b \). Il est facile de constater qu'une telle fonction se calcule de proche en proche avec, en gros, trois multiplications. \( a \) et \( b \) étant des fonctions en escalier, variant de 20 à 100 fois par seconde de son réel.

ESSAIS EFFECTUÉS

Un programme de synthèse de sons (2) écrit spécialement pour traiter de telles fonctions calcule en temps réel des partitions complexes à une fréquence \( F = 26000 \) Hz.

Plusieurs heures de musique ont été obtenues de cette façon avec des timbres très divers.

CONCLUSION

Ces techniques récurrentes associées à un traitement discret des paramètres moins significatifs pour l'oreille permettent la production d'œuvres de grande complexité et ceci pour un coût très raisonnable. En regard du prix considérable de la musique traditionnelle, la musique par ordinateur est susceptible de prendre une place importante dans les prochaines années.

(1) J.M. CHOWNING. J. Of the Audio Engineering Society. Sept 1973
(2) F. BROWN, R. LEHMANN, G. KLEIN. Revue d'Acoustique n° 30, p 206 à 215
Die realitätsbezogenen FO/Formant-Relationen und die Bildung von Klangmaterialsystemen

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Eine realitätsbezogene Werktechnik für Musikinstrumente sowie eine künstlerisch flexible, optimale Musikrealisation mittels EDV (elektron.

\[1\] V. Nösselt: Das Klingen; Proc. 7. ICA, Budapest (1971), 20 5 16

786
INTRODUCTION

Theories explaining the tonal consonance of complex tones with many frequency components were proposed by Plomp, Levelt and Kameoka, Kuriyagawa. These theories are based on their experimental results that the consonance of a complex tone produced by two frequency components increases with increasing frequency difference independent of frequency ratio if the difference exceeds about a quarter of critical bandwidth. This paper showed that the theories cannot be applied in case of some particular combination of frequency components.

EXPERIMENT AND DISCUSSION

Five kinds of complex tones consisting of 4 frequency components \( (f_1, f_2, f_3, f_4) \) were synthesized on a digital computer as shown in Table 1. Each complex tone had a duration of 400msec and a rise and fall time of 50msec. Five subjects compared pairs of tone stimuli randomly chosen from the five kinds of stimuli and were asked to decide which of two stimuli, presented successively during 300msec intervals, was more consonant. According to the conventional theories, the stimulus of a larger number in Table 1 is expected to be more consonant than the stimulus of a smaller number, since both frequency ratios and differences of adjacent constituent frequencies are larger with increasing stimulus number.

Fig.1 shows the result of the experiment. In most of the cases, the stimulus became more consonant with increasing stimulus number but the stimulus No.3 was most consonant of five stimuli. The former result supports the conventional theories but the latter result cannot be explained by the theories.

Although frequency differences of constituent frequencies can be considered one of factors for the consonance sensation, periodicity of the waveform of tone stimulus seems to be another important factor for the consonance sensation.

<table>
<thead>
<tr>
<th>Tone No.</th>
<th>( f_1 )</th>
<th>( f_2 )</th>
<th>( f_3 )</th>
<th>( f_4 )</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>800</td>
<td>1100</td>
<td>1350</td>
<td>1550</td>
</tr>
<tr>
<td>2</td>
<td>800</td>
<td>1150</td>
<td>1450</td>
<td>1700</td>
</tr>
<tr>
<td>3</td>
<td>800</td>
<td>1200</td>
<td>1600</td>
<td>2000</td>
</tr>
<tr>
<td>4</td>
<td>800</td>
<td>1250</td>
<td>1750</td>
<td>2500</td>
</tr>
<tr>
<td>5</td>
<td>800</td>
<td>1300</td>
<td>1850</td>
<td>2450</td>
</tr>
</tbody>
</table>

Table 1: Constituent frequencies of five kinds of complex tones used in this experiment.

Fig.1: Percentage of responses, averaged over five subjects, in which the consonance of a tone stimulus was judged as more consonant than other four stimuli.

References
THE ACOUSTICS OF THE TENORA (Catalan Tenor Shawm)

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INTRODUCTION

An introductory study of the acoustics of the tenora has been made.

The tenora is a characteristic instrument of the "cobla"-outdoor Catalan folk ensemble-. This instrument evolves from the shawms in use in Europe till the 17th century, and since then has undertaken an isolated evolution in Catalonia (1).

It is a double-reed conical tube instrument 80 cm long. It is a B-flat instrument with lower fundamental of 164.9 Hz (F-sharp, sounding E).

EXPERIMENTAL TECHNIQUES

Passive resonances of the tube have been experimentally determined (Fig. 1). And an introductory spectral analysis has been carried out by means of a 400 point real-time spectral analyser.

RESULTS AND CONCLUSIONS

Partials of the tube are in good agreement with the harmonics of the fundamental, as can be seen in the harmonic response shown in Fig. 2, and resonance peaks are fairly wide.

These facts explain the timbral richness of the tenora sound. The spectral analysis has shown more than 25 partials for its lower register.

Due to this spread harmonic composition the tenora is very suitable for outdoor playing.

Fundamental tracking mechanism can trace weakened low frequency components (2).

The obtained results have encouraged the authors starting a wider research concerning mathematical modeling and accurate acoustic impedance measurements.

REFERENCES

Il est actuellement bien établi que les oscillations de la colonne d'air des instruments à vent sont entretenues grâce à un mécanisme non-linéaire (A.H. Benade).

La colonne d'air d'un instrument tel que la clarinette peut être caractérisée par les valeurs de son impédance d'entrée; cette impédance présente en général une série de N maxima qui sont les modes propres de la colonne. Dans ces conditions on peut considérer cette colonne comme étant constituée d'une série de N oscillateurs harmoniques. D'autre part, on considère qu'aux petites amplitudes l'anche se comporte comme un oscillateur harmonique unique. Par son mouvement l'anche détermine alors, au niveau du bec, des variations de vitesse volumique non-proportionnelles à son élongation et cette non-linéarité introduit un couplage entre les N modes propres de la colonne (W.E. Worman).

Des expériences de visualisation du mouvement de l'anche combinées à des mesures de pressions acoustiques et statiques nous ont permis de constater que le fonctionnement normal (musical) de l'instrument correspond au batttemeent de l'anche sur le bec. Ces observations nous ont conduit à substituer au modèle d'oscillateur harmonique de l'anche un modèle d'oscillateur à relaxation qui permet de décrire le battement et ceci indépendamment de l'amplitude. Le fonctionnement non-battant de l'anche (non-musical) permet aussi de constater que l'on peut raisonnablement séparer le rôle de l'anche en une contribution purement hydrodynamique et une contribution à l'augmentation de la pression acoustique au niveau du bec. La loi non-linéaire d'ouverture de l'anche établit ainsi le couplage entre les N modes de la colonne et on aboutit à un modèle de fonctionnement de l'instrument où les auto-oscillations résultent du couplage de N oscillateurs harmoniques par un oscillateur à relaxation.


W.E. Worman : Self-sustained nonlinear oscillations of medium amplitude in clarinet-like systems, Cax Western Reserve University, Ph. D., 1971
INTRODUCTION

This is a continuation of a paper given at the 7th Congress 1971, "Study of the Timbre of Wet and Dry Japanese Bamboo Pipe, Shakuhachi." Using the same material, this study is extended to the attack transients, extracting its spectrum, and comparing to that of the steady state to investigate its behavior.

MATERIAL USED

Based on a D 286Hz, seven sounds, strong and weak, were recorded by one player using two pipes, one "good," one "unfinished" which was later placed in water or lacquered. Each material was digitalized at a rate of 10kHz sampling.

PROCEDURES

The input data was processed in 3 steps.

1. Pitch-synchronous analysis was performed.
2. As for the spectrum of the steady state, it was calculated from averaged values of its harmonics lasting approximately 100 msec. and was plotted.
3. This averaged spectrum of (2) was subtracted from that of (1).

Refer to Fig. 2.

RESULTS

Although the attack shapes differ from one to another, the low-ordered harmonics are weak while the 4th is always strong as shown in Figs. 1 and 2. There is a dip on the 5th. The attack time to reach the half-power amplitude takes place in

1. 175 msec. unlac dry, weak;
2. 100 msec. unlac wet, weak;
3. 100 msec. unlac wet, strong;
4. 125 msec. good, weak;
5. 125 msec. lac wet, weak;
6. 125 msec. lac wet, strong;
7. 125 msec. good, strong.

REFERENCES

LE RÔLE DU TUYAU DANS LE TIMBRE DE CERTAINS INSTRUMENTS À TROUS LATERAUX

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INTRODUCTION

Les théories de la ligne de transmission et des filtres (1) appliquées à l'acoustique et aux instruments à vent sont bien connues. Nous avons cherché à les appliquer à certains instruments à trous latéraux, notamment dans des domaines de fréquence où le système excitéur ne joue pas un rôle trop grand sur le timbre.

CALCULS THÉORIQUES

On calcule le coefficient de transmission dans le tuyau à travers diverses discontinuités (trois latéraux, appelés cheminées, ouvertes ou fermées, considérées comme des dérivations; changement de conicité; discontinuités de section). Les calculs ont été faits sur ordinateur d'abord dans les approximations les plus grossières, sans corrections, pour le mode (0,0). Tous les tuyaux constitués d'éléments de cônes recouverts peuvent ainsi être étudiés (le cylindre étant un cas particulier du cône).

Nous considérons un tuyau avec n cheminées ouvertes comme (n+1) tuyaux dans lesquels nous calculons le coefficient de transmission. Nous cherchons ainsi le spectre du son rayonné par chaque cheminée ouverte, ainsi que par le pavillon.

VERIFICATION EXPERIMENTALE

On sépare par des écrans les trous ouverts pour pouvoir analyser le spectre du son rayonné par chacun d'eux. Un microphone est placé à la sortie de chaque trou, et l'analyse est faite au sonographie. On voit sur la figure (1) le résultat schématisé pour le fa2 (174,6 Hz), à la sortie du trou de fa2 (a), du trou de do2 (b) et du pavillon (c) pour le basson système Buffet.

CONCLUSION

Les résultats théoriques corroborent bien l'expérience. Pour les différents systèmes de bassons (notamment Buffet et Heckel), on retrouve bien la place des formants et leur largeur (en-dessous de 2000 Hz, domaine au-dessous duquel l'onde a un rôle prépondérant) (2). La place et la largeur des formants (par exemple celui situé entre 440 Hz et 550 Hz environ) est déterminée notamment par l'écartement des cheminées et leur intensité par la section et la hauteur des cheminées.

L'étude du son rayonné par chaque trou rend d'autre part bien compte de la complexité du problème de la directionnalité, qui dépend de la note jouée ainsi que de la hauteur de l'harmonique. Cette étude, théorique et expérimentale, permet d'aborder le timbre des instruments à trous latéraux de manière plus précise et plus complète.

REFERENCES

(2) J. Kergomard et J.M. Heinrich, bulletin du G.A.M., 1976, 82-83, " Le basson ".

791
THEORETICAL SOLUTION OF THE STEADY STATE OSCILLATIONS OF ORGAN PIPE BY AN INTEGRAL EQUATION METHOD

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INTRODUCTION

A non-linear integral equation has been developed that describes the steady state autonomous oscillations of a class of systems that includes musical instruments. It has been applied to the model of the organ pipe described by Fletcher (1), which is his synthesis of the recent work by Cremer and Ising, Elder, Coltman, and Fletcher. The calculations described here obtain the spectrum of the particle velocity in the mouth of the organ pipe to an arbitrary number of harmonics—typically 31, but sometimes 63 to check the accuracy of the calculations.

THEORETICAL BASIS

The excitation mechanisms described in reference (1) depend on the position of the exciting air jet as it strikes the edge or lip of the pipe. That position is determined by the particle velocity of the oscillating air column in the mouth of the pipe, and on the air jet dynamics. The dispersive model of jet dynamics from reference (1) is used here. For the steady state oscillations, the particle velocity $V(t)$ in the mouth can be written, with time measured in units of the period, as

$$V(t) = \int_0^1 K(t-t') Q_j(X(t')) \, dt'$$

where the jet current $Q_j$ depends on the transverse displacement of the jet at the lip, $X(t)$. The $k^{th}$ Fourier component of that displacement is determined by the $k$ component of $V(t)$:

$$X_k = i S_p \exp(-i \delta_k) v_k / S_m = a_k v_k$$

where $S_p$ and $S_m$ are the pipe and mouth areas, and the dynamics of the dispersive jet are described by $C_k$ and $\delta_k$ (See reference (1)). The kernel of equation (1) is

$$K(t) = \sum_{k=-N}^{N} \frac{H_k}{S_p(Z_k + Z_m)} e^{2\pi i k t} = \sum_{k=-N}^{N} H_k e^{2\pi i k t}$$

$Z_k$ and $Z_m$ are the pipe impedances as seen at the lip looking away from and towards the mouth respectively, and $H_k = p V_0 / S_p + Z_m$, where $p$ is the air density and $V_0$ the jet velocity.

The equations (1) and (3) are solved analytically for $N=2$ and 3, and yields the requirement on the solutions that

\[ \sum_{k=1}^{N} \frac{k^2}{k} \, a_k = 0 \]  

where $a_k = |v_k| / v_k$, and $b_k = \Im(1/a_k v_k)$. These approximate solutions give approximate oscillation frequencies and amplitudes that serve as adequate initial approximations to numerical, iterative solutions of the full integral equation, Eq.(1). Those results show that the upper limit of (4) extends to arbitrary harmonic number.

RESULTS AND CONCLUSIONS

Results will be shown for $V(t)$ in the normal and overblown regimes. The results are so rich in predictive detail that available data are inadequate to test by comparison the model used in these calculations.

REFERENCES

INTRODUCTION.
Le réglage de l'orientation du jet par rapport au biseau est un des paramètres importants de l'excitation des tuyaux à bouche. Lors de l'étude, on distingue d'une part le rôle de l'angle (θ) que fait la direction moyenne du jet avec l'arête du biseau, et d'autre part la déviation latérale du plan du biseau par rapport à l'axe du jet (e). Ce dernier paramètre est de loin le plus sensible. Y. ANUO(*) a montré, à propos de l'étude de la flûte traversière, que (e) contrôle à la fois les pressions de passage d'un régime au suivant, donc l'octavation, et le timbre des sons rayonnés par l'instrument.
L'étude de Y. ANUO ne porte que sur 3 notes en régime stationnaire. A l'aide d'une flûte à bec dont on peut régler la position de la lumière par rapport au biseau, nous montrons l'incidence du paramètre (e) sur le timbre des sons de l'instrument, dans des conditions normales de jeu.

EXPERIENCE ET RESULTATS.

FLUTE À BEC ALTO
L'analyse au sonagraph montre d'importantes modifications du spectre et des transitoires d'attaque. L'image photographique de la lumière vue depuis l'entrée du bec permet de repérer la position du biseau.

a) Jet très extérieur - Harmoniques de tous rangs, dont le 2ème intense émission facile dans l'aigu, mais beaucoup de bruit au transitoire d'attaque.
b) Jet médian - Fréquence très nette des harmoniques impairs ; attaque moins bruyante, instrument mieux équilibré en intensité.
c) Jet très intérieur - Réapparition des harmoniques pairs, mais le 2ème est plus faible qu'en a) ; retard des harmoniques aigus à l'attaque.

CONCLUSIONS.
En conditionnant en grande partie la forme de l'oscillation du jet au niveau de la bouche, le réglage de l'orientation du jet par rapport au biseau joue un rôle déterminant dans le timbre des tuyaux à bouche. Il permet d'expliquer le rôle des chanfreins des bords de la lumière et est responsable en grande partie des caractéristiques du timbre que les facteurs attribuent généralement au matériau.

REFERENCES.
(*) ANUO Y.-Influence of the air beam direction upon acoustical proper-
ties of flute ones - C.R. 6° ICA , Tokyo (1968)
(**) CASTELLENGO M.- Contribution à l'étude expérimentale des tuyaux à bouche - Thèse , Paris (1976)
SOUND PATTERNS FROM THE CHIME

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INTRODUCTION

Although any blow on an orchestral chime is recognizable as a chime sound, the careful observer can hear a definite difference in the sound depending on his position with respect to the chime, how it is mounted, and how the blow is struck.

The chime, unless very carefully made, exhibits a warbling or beating tone (1), similar to that of some bells. The most prominent partials of the tone can be shown to be doublets (2-3). If the chime is perfectly symmetrical about its longitudinal axis, these two patterns will have the same frequency. With most chimes, there are small perturbations in the symmetry, causing one of the pair to be slightly out of tune with the other; this results in a slow beating of the sound. Furthermore, the sound pattern in the region around the chime changes with time and location, and may be different from blow to blow. In this experiment we investigated this sound field for several blows of an A4 chime.

EXPERIMENTAL TECHNIQUES

The A(440Hz) tube from a set of chimes was suspended in a sound absorbent room near an array of thirteen microphones. A row of eight microphones faced the tube in the direction the blows were to be applied. These were at a distance of 12 cm. and spaced along the length of the tube. Two others were placed 12 cm. from the side of the tube, at right angles to the front-facing row. Two more microphones were set at a distance of one meter from the center of the tube, one in front and one at the side. The last microphone was placed in the far field, at a distance of four meters in front of the chime.

The signal from each of the 13 microphones was fed to a separate channel of a multi-channel tape recorder. Each time a blow was struck, all 13 signals were recorded simultaneously. The individual recordings were fed through band-pass filters and plots of sound pressure level as a function of time were made for all of the prominent components. The resulting display allows us to study the sound pattern in both space and time.

RESULTS AND CONCLUSIONS

For the particular tube studied, the periods of the beats are found to be 8.8 sec., 8.2 sec., 6.1 sec., and 3.3 sec. for the 540-, 880-, 1289-, and 1758-Hz components respectively. The lowest (91.2 Hz) component was not plotted and the second (277 Hz) did not exhibit a regular beating pattern. Generally, the maximums for one frequency occur at about the same time for all the microphones facing one side of the tube. When there is a maximum in one set of microphones, there is a corresponding minimum in the set that is a quarter of the way around the tube.

The superposition of the waves in the air emanating from the various antinodes of the tube's vibration should set up a pattern in the air around the tube consisting of regions of strong vibration and regions of relative quiet (4). The records from the microphones demonstrate clearly the existence of such a pattern.

REFERENCES

(1) T.D. Rossing, The Physics Teacher (1976) 14, 566
(2) A.T. Jones, JASA (1949) 21, 315
(4) J.J. Schroeder, Acustica (1963) 106
INTRODUCTION:
In this paper the ancient musical instruments will be analyzed, and through experiments the data on frequency response and acoustical power for one of these instruments called Setar (see fig 1) will be presented.

A more detailed data on remaining instruments is available through a separate paper.

THEORETICAL BASES
Sound power levels defines the acoustic output power of the source. Envision a hypothetical sphere of radius R which surrounds a source; the sound power generated by that source is proportional to the average pressure at the surface of the sphere and is usually expressed as sound power level in decibels referred to a reference power:

\[ PWL = 10 \log \frac{W}{W_{REF}} \]

Where PWL is the sound power level, W is the sound power in Watts, W_{REF} is the reference power, usually 10^{-12} Watt or 1 picowatt. The sound power level is determined in an anechoic environment. The output of each microphone is measured in octave bands, PWL is calculated from the following equation:

\[ PWL = \text{spl} + 20 \log R + 10 \text{db} \]

\[ R = 0.80 \text{ Meter} \]

\[ \text{spl}_n = 10 \log \left( \frac{\text{antilog} \text{spl}_{n-1} + \text{antilog} \text{spl}_{n-2} + \ldots + \text{antilog} \text{spl}_1}{10} \right) \]

EXPERIMENTAL TECHNIQUES
All measurements of acoustical powers and frequency responses of musical instruments are done in an anechoic chamber of an acoustic lab in the school of Television and Cinema. Acoustical power measurements are done according to the theory described above. All of the measurements are carried out at the maximum power output of a played instrument at various frequencies.

The sound pressure measurements are done at eight different locations on hypothetical sphere. (see fig 2)

Frequency response is measured with real time analyzer and recorded on paper. (see fig 3)

RESULTS AND CONCLUSION
This is the first time that research has been done on the acoustic properties of ancient Iranian musical instruments, and the work has presented difficulties due to the lack of standard shapes and frequency responses. The results of this experiment have been obtained using a common Setar produced by a well known craftsman, Mr. Esghghi. In this way, it is hoped that the criteria suggested could be used as a way of standardization of these kind of instruments.

REFERENCES:
General Radio Company
INTRODUCTION

As we noticed the relation of harmonics as one of important factors in frequency component of music tone, we made a system in which harmonics produced by nonlinear circuit can be added to any level, and found that the quality of violin sound through telephone line was improved in the system.

SYSTEM

The system consists of a high pass filter, a nonlinear circuit and delay circuits. The harmonics corresponding to the lost frequency component are produced from the signal through the high pass filter using nonlinear circuit, and are added to the band-limited sound, delayed or advanced, in the time domain by the delay circuits; in the nonlinear circuit, output power level is directly proportional to input power level.

SUBJECTIVE ASSESSMENT TEST AND RESULTS

Four samples were used in assessment test: A [band-limited sound (150-3400 Hz)]; B [band-limited sound (delay 0 ms) + harmonics (delay 0 ms)]; C [band-limited sound (delay 6.25 ms) + harmonics (delay 0 ms)]; D [band-limited sound (delay 0 ms) + harmonics (delay 6.25 ms)].

These samples, paired at random, were subjectively assessed by 56 students, under training of hearing, by method of seven-gradation pair-comparison in a studio.

Scale of main effect figured as a result of analyzing variance are indicated below:

<table>
<thead>
<tr>
<th>Factors</th>
<th>SS</th>
<th>DOF</th>
<th>MSE</th>
<th>F</th>
</tr>
</thead>
<tbody>
<tr>
<td>M</td>
<td>211.67</td>
<td>2</td>
<td>105.84</td>
<td>71.29</td>
</tr>
<tr>
<td>M X F</td>
<td>535.31</td>
<td>2</td>
<td>267.66</td>
<td>97.66</td>
</tr>
<tr>
<td>C</td>
<td>0.06</td>
<td>3</td>
<td>0.02</td>
<td>0.13</td>
</tr>
<tr>
<td>D</td>
<td>6.10</td>
<td>1</td>
<td>6.10</td>
<td>6.10</td>
</tr>
<tr>
<td>D X F</td>
<td>424.44</td>
<td>2</td>
<td>212.22</td>
<td>0.17</td>
</tr>
<tr>
<td>L</td>
<td>203.41</td>
<td>445</td>
<td>0.45</td>
<td></td>
</tr>
<tr>
<td>TSSX</td>
<td>2782</td>
<td>445</td>
<td>0.62</td>
<td></td>
</tr>
</tbody>
</table>

Decision is that main effect is especially significant: that subjects were influenced by exhibit of order of samples; and that B, C and D were all preferable to A, and also D was preferable to B when each sample’s significance was tested in 99% confidence interval.

Also all subjects clearly pointed out that there was 200 Hz of difference between each high cut-off frequency of band-limited sound, from 2800 Hz to 3400 Hz.

Thus, it was proved (Fig. 2) that the quality of the band-limited violin sound improved by adding harmonics and that the quality improved best by adding delayed harmonics.

CONCLUSION

We observe that this method can make effective sound-compensation in band-limited transmission such as in telephone lines, and that its application in this area can be much expected.

(The authors wish to thank Prof. Kitamura of Kyushu Institute of Design for his advice in this test.)
LA PORTEE DU SON DU VIOLON DANS L'ESPACE LIMITE

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Ecole Normale Supérieure
Zielona Góra - Pologne

INTRODUCTION

La portée du son est universellement considérée comme une de principales valeurs de son des instruments à cordes /1-2/.

Les recherches présentes ont pour but de trouver un modèle acoustique du violon, le meilleur possible. Pour y arriver on a pris à tâche de définir les changements de principaux paramètres du son en tenant en considération les conditions différentes, mais toujours en divers locaux fermés.

TECHNIQUE DE L'EXPÉRIENCE

Suivant les appréciations subjectives des experts on a choisi les instruments ayant la portée différente. Dans plusieurs coins de la salle on a enregistré des sons fondamentaux /3/ venant du jeu continu /détaché mf/ ainsi que des sons isolés /martelé ff/. On a pris les mesures dans une salle de concert, dans un local pour la musique de chambre et dans un studio. On a comparé les résultats de l'analyse oscillographique et spectrale concernant les changements de l'intensité et du timbre du son avec ceux de l'appréciation subjective faite en même temps.

RÉSULTATS

Il faut distinguer la portée du son en tenant compte de son intensité et de son timbre. L'étendue des changements est fort différenciée suivant les instruments qu'on en étudie.

Changements maximum des paramètres du son dans limites de la salle

<table>
<thead>
<tr>
<th>Paramètre</th>
<th>G Son continu</th>
<th>D Son continu</th>
<th>A Son continu</th>
<th>E Son continu</th>
<th>concert</th>
<th>G D A E</th>
<th>de concert</th>
<th>n</th>
<th>1-5 0-3 0-5 1-4 1-6</th>
<th>14,0</th>
</tr>
</thead>
<tbody>
<tr>
<td>A /dB/</td>
<td>2,0-</td>
<td>5,0-</td>
<td>2,5-</td>
<td>1,0-</td>
<td>1,5-</td>
<td>1,0-</td>
<td>3,0-</td>
<td>2,0-1,5</td>
<td>14,0 18,0 20,0 14,0 14,0 16,0 6,5 19,0 14,0</td>
<td></td>
</tr>
<tr>
<td>pour la</td>
<td>7,0-</td>
<td>9,0-</td>
<td>10,0-</td>
<td>8,0-</td>
<td>4,0-</td>
<td>2,0-</td>
<td>6,0-</td>
<td>9,0</td>
<td>12,0</td>
<td></td>
</tr>
<tr>
<td>musique de</td>
<td>0-3</td>
<td>0-1</td>
<td>1-3</td>
<td>1-3</td>
<td>1-4</td>
<td>1-3</td>
<td>1-4</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

REFERENCES

3. L'étude a été faite en coopération avec SPAL-Varsovie, L'Ecole Sup. de Musique de Poznań, Le Musée des Instr. de Musique à Poznań
INTRODUCTION

Two methods of studying the Eigenmodes of free violin plates have been reported (1); by hologram interferometry and by vibration patterns known as Chladni patterns. The latter method, mentioned just briefly at the ICA 7 in 1971, is not only simple and relatively inexpensive, but is proving very effective in checking some of the stiffness characteristics of arched violin plates as well as those of braced guitar tops and backs for good tone production in finished instruments.

THEORETICAL BASES

Theoretical considerations indicate that the higher the Q of mode # 5, the so called tap tone or ring mode, the more likely this vibration will be effective in the coupled system when the top and back plates are glued on the ribs.

EXPERIMENTAL TECHNIQUES

The test plate is activated by sympathetic vibration from a loudspeaker using a sine-sweep input from an audiogenerator and appropriate amplification, with the plate mounted horizontally over the speaker cone on soft foam pads, and its inside surface sprinkled with a lightweight granular substance such as sawdust or aluminum flake. (2, 3) As the sine wave is moved through the frequency range from about 50 to 800 Hz, the various plate Eigenmodes can be observed by watching the flake bounce out of the active antinodal areas and pile up along the nodal lines when the frequency of the input signal matches the frequency of one of the plate resonances.

RESULTS AND CONCLUSIONS

For free violin plates it has been found that when Mode # 5 shows both high amplitude and high Q in each plate and the frequencies of this mode in a top and back pair are not more than a tone apart, a good sounding instrument results. If these two frequencies match, tests indicate that the Q's should be different. Studies of other modes in violin plates and their effect on tone quality and playing characteristics are under way. This method in combination with response curves of free violin plates has been used in the development of the eight new instruments of the violin family (5) and in fine sounding conventional violins, violas and cellos. Free top and back guitar plates exhibit the ring mode, # 5, when the struts are properly placed and their stiffnesses adjusted (4). Resultant tone and playing qualities seem most promising. Several violins and guitars based on these findings will be available for trial and discussion.

REFERENCES

The frequency response of complete violins and of isolated bridges has been measured at very small amplitudes by harmonic excitation of the damped strings and recording of lateral bridge top movement. Bridge response curves have been obtained by mounting the bridge on a "mute violin" where the bridge base was completely rigid.

It was found that below 1000Hz the elastic properties of the bridge have little influence on total vibration whereas between 1 and 3 KHz the bridge exhibits pronounced resonance peaks with a strong influence on violin timbre. Bridge natural frequencies are between 2, 1 and 3 KHz. Fig. 1 shows typical bridge frequency responses. The measured decrease of bridge natural frequency with additional masses at the bridge top agrees well with theoretical prediction for a simple mass-spring model. Bridge and violin corpus at the small amplitudes of the experiment can be considered as linear systems, therefore relative changes of bridge frequency response are directly reflected in total instrument timbre.

By measuring bridge stiffness for the fundamental vibration mode and bridge frequency response on different test rigs, correlations have been established between the geometric and elastic properties of the bridge and their influence on violin timbre.

Because maple varies considerably in stiffness and changes in bridge geometry usually affect both stiffness and mass, the net acoustic effect cannot be predicted by the luthier without quantitative information. Some ground rules have been derived from the test series for practical application.

The method of cutting bridges such as to obtain specific response curves has been applied to a large number of instruments with consistent results of violin timbre changes.

**Fig. 1**

- **FORMANT RANGES**
- **FREQUENCY RESPONSE OF BRIDGE WITH 4 STRINGS BASE FIXED**
- **BASIC BRIDGE**
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9 INTERNATIONAL CONGRESS ON ACOUSTICS
MADRID 4/9-VIl-1977
HELMOHOLTZIAN

MULTIPLE

WAVES

ON A BOWED

Gakushuin University,

Kondo, M.

STRING

Tokyo —

Japan

Kuni, A.
Kubota, H.

The envelopes of a bowed string of a violin have been taken with a double cylindri-

cal lens camera(l) at various bowing epeeds and pressures, even in the unusual

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the motion of a bowed string, whatever complex and unmusicel its sounds are, is
composed exclusively of a number of pure Helmholtzian waves.

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ranges of playing.(Fig. 1)
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Express in solid make
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The shapes of these envelopes lead us to conclude that

statement ©

oryincing,let us take the method of synthesis.

X-axis in Chart 1, and thoseof Wy on Che Y-AR1g"oltrian wayos (W1) located on the

one of Wy and one of Wy, occur simultaneously, the envelope will be complex in shape
—except in the cases of ¢=0, where ¥ is the phase difference between Wy end Wg.
When $50, each Helmholtzian wave makes its own envelope, as shown in Chart l,with This distinction we can not make in the photo
solid and dotted lines respectively.
But with the charts of composition like Chart 1, we records of the experiments.
can find out its componente and make identification.
The envelope composed of =~

three Helmholtzian waves is much more complicated than of the two, and only a simple
example is given ati the right end frame in Fig. 1.

Fig, 1

Some envelopes of a bowed string

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1

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Chart 1

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P=21X ä'

!!:21[)(%_

Envelope synthesis of two Helmholtzian waves

o>

P=21X A


800


A NEW METHOD OF STUDYING THE MOTION OF A BOWED STRING

Kubota, H. Gakushuin University, Tokyo - Japan
Kondo, M.

(1) VISUALIZATION OF "STICK AND SLIP"

As shown in Fig.1, a 35mm camera is fixed with a bow so as to take the picture of "stick and slip" at the points of contact, and the shutter is made open during the time interval in which the string image travels on the film surface from one end to the other (streak photo). Fig.2 is the result when the bowing is normal, while Fig.3 shows two stickings during the one whole cycle. (A vertical center zone in each figure is the back of the bow.)

(2) A NEW RUNNING FILM METHOD IN A SPACE FIXED CAMERA AIDED WITH "THE HETERO-MAGNIFYING DEVICE" (Fig.4)

The camera catches the whole length of the bowed string with the magnified amplitude of vibration. Thus the more extended pictures of the bowed string than (1) can be obtained (Fig.5 and 6). Fig.7 and 8 are the enlarged pictures of Fig.5 and 6, and correspond to Fig.2 and 3 respectively. It is interesting to note that for analysing the string motion the speed of the film is not necessarily the same as that of the bow.

(3) APPLICATION

In the photographic records of the string, taken by the above mentioned method, we can easily find out the amplitudes and phases of all Helmholtzian waves on the string, thus enabling to investigate the transient behaviour of a bowed string in music playing.
THE ATTACK AND DECAY TRANSIENTS OF GUITARS AND PIANOS

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INTRODUCTION

The perceived pitch and timbre of musical sounds is very largely determined by the "steady-state" spectrum, but the dynamic and interpretative aspects are almost entirely contained in the changing or transient portions. These include effects intentionally created by the musician, such as vibrato, staccato and glissando, and also transients which are characteristic of the instrument and over which the musician has little or no control. These include attack, decay and release transients and "steady-state" instability. An earlier study investigated the attack transients of some organ pipes (1). The present study concerns the attack and decay transients of a variety of guitar and piano sounds.

TRANSIENT ANALYSIS

It has been shown (2) that within certain constraints the transient waveforms of musical sounds which have an almost-periodic "steady-state" can be accurately represented by the changing Fourier series

\[ f(t) = \sum_{n=1}^{k} A_n(t) \cos(nut + \phi_n(t)) \]

where \( A_n(t) \) is a function of time describing the amplitude of the \( n \)th harmonic and \( \phi_n(t) \) is another function of time describing its phase. The accuracy of such a representation is dependent on the rate of change of \( A \) and \( \phi \) and on the inharmonicity of the sound being studied.

EXPERIMENTAL TECHNIQUES

To determine the functions \( A_n(t) \) and \( \phi_n(t) \) the musical transients under study were first digitized and then were broken into segments each one period long. Each of these segments was individually analysed by a numerical simulation of Fourier techniques. The results for each harmonic amplitude, \( A_n \), and phase, \( \phi_n \), were plotted against time and a curve fitted to them. The result is a graphical description of the functions \( A_n(t) \) and \( \phi_n(t) \).

RESULTS AND CONCLUSIONS

The study has shown a strong correlation between the transient and steady-state frequency response of guitars, especially at low pitches. It has also shown that both responses are more uniform in better quality guitars. The method of plucking a guitar whether with a plectrum or the thumb has some effect on the transient response at low frequencies.

In the piano the striking force affects the spectrum of lower pitched notes but has little effect on spectrum in the middle and upper part of the range. The attack and decay rates are nearly independent of the striking force.

The transient response of pianos and guitars are very similar and in each the duration of the attack and the decay are approximately proportional to the period of the fundamental.

REFERENCES

ON INSTRUMENT AIDED PIANO TUNING

Agulló-Batlle, J.  
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Universidad Politécnica de Barcelona

INTRODUCTION

Aural tuning of pianos results in a "stretched" scale. As has been reported by numerous authors, this fact comes from the inharmonicity of piano strings (1)(2)(3).

There is no "absolute stretched scale" for piano tuning. Each model, and even each instrument, requires its own characteristic stretching which is obtained by the tuner by means of aural evaluation. This has prevented straightforward electronic counter tuning.

A method is envisaged for instrument aided tuning, claimed to provide characteristic stretched scale for each individual instrument.

BASES OF THE METHOD

Due to the inharmonicity of piano strings, partials of one string cannot match exactly corresponding partials of the string one octave below. A compromise must be settled.

Aural optimisation relies on intensity and duration of each pair of corresponding partials.

Analytical optimisation methods can be envisaged to minimize aural effect of partials mismatching, for a given audiometric curve.

EXPERIMENTAL SUPPORT

To trace pairs of partials governing piano tuning measures have been made of partial departures from equally tempered scale harmonics, on a SCHIMMEL-174E medium "grand piano" (Fig.1).

Spectral analysis of sounds produced have been carried out to trace the tuning pairs actually used by aural tuners.

RESULTS AND CONCLUSIONS

Experimental measurements have allowed setting a "tuning layout" defined by the set of partials to be tuned in pairs.

This "tuning layout", which provides the key to piano tuning, is claimed to be far more invariant from one piano to another than the characteristic stretched scale is.

To test practical suitability of the method, special purpose electronic instrumentation is under development.

REFERENCES

On a déterminé, par la méthode électroacoustique, les réponses sonores résonnantes du corps de la guitare, dont la table supérieure était sans petites barres, et on a déterminé les oscillations propres et leurs fréquences des caractéristiques enregistrées amplitude/fréquence. À l'excitation nouvelle du corps de la guitare par les oscillations sinusoidales, dont les fréquences correspondent aux celles des oscillations propres du corps, par l'exciteur électromagnétique, les figures d'interférence sur la table de résonance étaient enregistrées par la méthode de l'infréometrie holographique (Figs 1 et 2). On a observé les changements des figures d'interférence apparaues, en dépendance de l'intensité de la force coercitive, de la fréquence d'excitation et de l'influence de l'accroissement du nombre de petites barres collées sur la table de résonance.

Refernces:
2. J. Jovčič Acustica Vo. 18 No 6, 356-360
Il est connu que le spectre tonique de la guitare change le plus si le nombre et la distribution de barres sur la table de résonance changent. Les meilleures conditions pour la résonance sont possédées par les corps ayant le grand nombre d'oscillations propres dans le domaine de fréquence englobant tous les harmoniques des cordes. Dans cet article on a examiné les changements des oscillations propres du corps à travers des caractéristiques amplitude/fréquence des réponses résonnantes, advenant à cause des changements du nombre et de la distribution de barres sur la table de résonance de la guitare (Diagramme). Le modèle sans barres a été examiné et ses oscillations propres étaient déterminées. Puis, les changements de ces oscillations propres étaient suivis quand trois et puis cinq et sept barres étaient collées sur la table de résonance. En analysant des diagrammes obtenus on a observé des changements dans la forme des courbes des réponses de fréquence, l'établissement des nouvelles oscillations propres à cause de l'accroissement du nombre de barres, l'accroissement du niveau relatif général du son et l'élargissement du domaine pour la résonance des oscillations propres particulières.

Références
2. Jansson B.V. ACUSTICA 25, 95-100
Although percussion instruments have witnessed a surge of popularity in recent years, little research has been done on their acoustics. [This is in contrast to the rather extensive literature on string instruments and the recent interest in the acoustics of wind instruments (1).]

Percussion instruments frequently employ vibrators such as bars, membranes and plates, whose modes of vibration are not harmonic. These inharmonic overtones give percussion instruments a distinctive timbre. In this paper, we describe the modes of vibration and other acoustical properties of the glockenspiel, marimba, xylophone, vibes, and chimes.

**GLOCKENSPIEL**

Glockenspiel or orchestra bells employ rectangular steel bars 1 to 1 1/4 inches wide and 3/8 to 5/16 inches thick. The frequencies of the transverse vibrations are quite well described by the equation for a thin bar: \( f = \frac{K}{L^2}\sqrt{\frac{Y}{\rho}} \), where \( K \) is the radius of gyration, \( L \) is the length, \( Y \) is Young’s modulus, and \( \rho \) is the density. In addition to the transverse vibrations, torsional and longitudinal modes can be identified. Because the overtones have very high frequencies and die out quickly they are of relatively less importance in determining the timbre of the glockenspiel than are the overtones of a marimba or xylophone, and therefore no effort is made to tune them.

**MARIMBA, XYLOPHONE, VIBES**

The marimba typically includes 3 to 4 1/3 octaves of tuned rosewood or synthetic bars. Beneath each bar is a tubular resonator tuned to the fundamental frequency of that bar. A deep arch is cut in the underside of the bars in the low register so that the first overtone has 4 times the frequency of the fundamental (2).

The xylophone has 3 to 3 1/2 octaves of tuned bars and usually includes tubular resonators, but the overtones are tuned to 3 times the fundamental frequency. Thus the resonator reinforces the overtone as well as the fundamental. Xylophones are played with hard mallets and have a crisp, bright sound.

The vibraphone or vibraharp usually has 3 octaves of aluminum bars with overtones tuned similar to the marimba. The aluminum bars tend to have a long decay time, and so pedal-operated dampers are included. The most distinctive feature of vibes, however, are the motor-driven discs at the top of the resonators which open and close the tubes. These discs or pulsators typically generate an amplitude modulation of about 6 dB (3).

**CHIMES**

One of the most interesting characteristics of chimes or tubular bells is that there is no mode of vibration with a frequency near the pitch of the strike tone. Modes 4, 5 and 6, whose frequencies have approximately the ratios 2:3:4, appear to determine the strike tone. Judicious choice of dimensions and the addition of a plug to one end of the chime tune the overtones to give the chime a bell-like timbre.

**REFERENCES**

V. Electroacoustics

Q. INSTRUMENTATION
TWO METHODS OF MEASURING ACOUSTIC IMPEDANCES

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INTRODUCTION Many methods of measuring acoustic impedances have already been described till now. The two methods this contribution concerns are therefore only the partly modified known methods: (1) the method of the sound source of the known volume velocity, with an auxiliary acoustic resistor and (2) the single pulse method, using a digital processing.

THEORETICAL BASES Method 1 - The source of a known acoustic volume velocity can be e.g. a rigid piston, if its front area is known and its velocity can be measured. This principle has been used e.g. in the TESLA MAI-1 impedance meter. Another possibility to obtain a source of known volume velocity is shown schematically in Fig. 1. Here 1 and 2 are the measuring microphones, 3 is a sound source and 4 is a tube with fixed acoustic impedance $Z_n$. If the dimensions of the device are small against the wave length of the measuring signal the following simple relation will hold:

$$Z_{ax} = \frac{Z_{no}}{\Pi_{(P_1 - P_2)}}$$

(1)

Evidently it will be advantageous to use the two microphones with the same transmission properties. Processing of the output signals of the two microphones can be done in an analog device or digitally. The impedance $Z_n$ dependence on the frequency can be simply compensated for or involved in the processing if it is made digitally.

Method 2 - The basic theory for this method is given e.g. in /2/. The arrangement that has been investigated is shown schematically in Fig. 2. Here 1 is a microphone, 2 is a sound duct and 3 is a sound source. Suppose $l_1$, $l_2$, and the transmitted pulse duration $T_x \ll \lambda/c$. Then the time function $p(t)$ of the microphone output voltage can be divided as it is shown in Fig. 2. The time shift of the part II of the original time function should be $T = 2\cdot l_2/c$. It is simply possible to carry out this procedure on the digitized signal. The FFT and further processing is then made on the functions $f_1(t)$ and $f_2(t)$.

EXPERIMENTAL TECHNIQUE AND RESULTS For method 1 different acoustic "impedance heads" were tried out. Digital instrumentation were used followed by the numeric processing on a programmable calculator. For method 2 the waveguide of 18mm dia (inner) was used, with $l_1 = 60$ cm and $l_2 = 3$ cm. The microphone ( B&K 4134 ) output signal was being digitised and captured by the Digital Event Recorder B&K for the following processing was made on the TESLA ZCG computer using the standard FFT routine. For checking and calibrating rigid stoppers and calibrated cylindrical cavities were used with both method 1 and 2. Very good accuracy was obtained with method 1. With method 2 high S/N ratio is essential, which especially on high frequencies depends first of all on the transmission properties of the driver.

Since 1961, the sensitivity as well as the distortion of insert earphones have been measured using the IEC standardized 2 cm² coupler. It is important, however, that the artificial ear, made up of a coupler and a microphone, loads the earphone with the same acoustic impedance as the human ear, because the sound pressure produced by the earphone depends on the acoustic impedance loading it. This is because most earphones have a relatively high acoustic output impedance. The existing IEC Recommendation R 126 has therefore rightly been criticized, since the 2 cm² volume does not in the least simulate the impedance of the human ear. The reason why a proposal for a more accurate artificial ear for insert earphones has not been suggested until now is simply the inaccuracy involved in impedance measurement of the human ear, which to the present date could not be measured above 8 kHz.

A method is described in the paper which involves insertion of two miniature condenser microphones into the ear canal, one used as a high-impedance transmitter, the other as a receiver, whereby the impedance can be measured rather accurately up to 20 kHz. The measurement method is described, and the results obtained from measurements on different human ears are illustrated.

Comparison of the results with those available up to date shows good correlation at the low frequencies, while new information is yielded at the higher frequencies.

Based on this and all other information about the impedance of the average human ear, Working Group 6 of IEC/TC 29C in March 1976 suggested an impedance curve which is shown in the figure. The equivalent volume at 500 Hz is 1.4 cm³, and the volume at 2000 Hz is 0.7 cm³. The length resonance is 14 kHz, and the volume is 0.07 cm³.
INTRODUCTION

As long ago as 1947 Nichols et al. demonstrated that the acoustic effect of placing a hearing aid on the body was considerably diminished when the measurements were made in a reverberant rather than an anechoic room. However, in spite of this early beginning which showed that the acoustic behavior of hearing aids as a function of body baffle in an anechoic room was not necessarily the same under more normal reverberant conditions, performance aspects such as directionality and head shadow effects continue to be measured almost exclusively in anechoic rooms. A considerable deterrent to the measurement of such characteristics under more commonly encountered reverberant acoustic conditions is the presence of standing waves in these environments. The present paper reports on a series of studies of azimuth effects with directional and nondirectional hearing aids with different microphone locations in various environments. The measurement technique used is one in which a broadband thermal noise serves as the test signal and frequency response is obtained using a real-time analyzer (RTA).

METHOD

The frequency responses of "nondirectional" hearing aids with downward and forward facing microphones were measured in isolation, on KEMAR and on real heads as a function of azimuth in different environments. In another study "directional" hearing aids were placed on KEMAR in several different environments including an anechoic room. The outputs of three directional hearing aids were observed as a function of azimuth and reverberation time at the test location. A signal "averager," of the type used in evoked response work was used to compare the frequency response curves obtained under different conditions by subtraction or to average the curves obtained from several subjects. The use of a broad band noise test signal and the RTA produced a controlled degree of smoothing of the standing wave patterns in the reverberant rooms while retaining adequate detail in the response curves.

RESULTS

Results at 0°, 90°, 180°, and 270° azimuths revealed good agreement between KEMAR and the mean results from four subjects with both downward facing and forward facing hearing aid microphones. The differences between the frequency responses obtained with the hearing aid on the head and with the hearing aid alone at each azimuth in four environments revealed the effects of head baffle and head and pinna shadow effects with these microphone locations. Results with three "directional" hearing aids at eight azimuths in an anechoic room, an audiometric test room and other more reverberant environments revealed a considerable reduction in the directivity of head worn directional hearing aids as reverberation time was increased.

REFERENCE

A NEW ARTIFICIAL EAR FOR TELEPHONE USE

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INTRODUCTION

Till today the CCITT recommends provisionally to use the IEC artificial ear as an acoustic load to measure the sensitivity/frequency curve for telephone receivers.

This artificial ear was obtained for audiometric requirements. In the telephone use the acoustic leak greatly influences the acoustic load.

In this paper we describe an automatic system for the impedance measurement in module and phase versus frequency of the human ear measured during the telephone use. The theoretical bases of the measurement were described in a foregoing paper (1).

MEASUREMENT CONDITIONS

The measurements were obtained with a telephone handset in a normal use. The telephone handset was the acoustic source. Measurements were made on 50 subjects at two different sound levels.

After obtaining the electric simulation of the acoustic impedance shape we built a prototype of a new artificial ear for telephone use.

A microphone probe at the handset auricle center drew the acoustic pressure.

RESULTS

Fig. 1 shows the median shape of the obtained acoustic impedance in comparison with the one of the IEC artificial ear.

Fig. 2 shows the electric simulation of the acoustic impedance of the human ear illustrated in fig. 1.

From this electric simulation we obtained a new artificial ear for telephone use.

REFERENCES

(1) G. Modena, A. Reolon, F.A.S.E. '75 - Coll. 1 - 1.1
MÉTHODE ET APPAREILAGE POUR DÉPISTER LES MALENTENDANTS

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INTRODUCTION
L'audiogramme classique mesure la sensibilité de l'oreille à un nombre fini et limité de fréquences sinusoidales, généralement entre 250 Hz et 8 ou 10,000 Hz. Comme nous l'a montré une longue expérience, il s'avère insuffisant pour apprécier les propriétés de l'oreille relativement à l'intelligibilité de la parole et à la perception de la musique. L'audiométrie néglige en particulier deux aspects importants en ces domaines la fréquence de coupure aiguë et le pouvoir séparateur temporel de l'oreille. Nous avons en effet vérifié qu'une audition déficiente sur ces deux points explique de nombreux échecs scolaires chez de jeunes enfants, et justifie souvent la mauvaise qualité des prestations de certains musiciens. Nous avons donc fait une recherche sur ces questions et imaginé une méthode de test de l'oreille, ayant abouti à la construction d'un appareillage spécifique.

APPAREILAGE ET MÉTHODE
L'appareil de test fréquence-temps (TFT), de dimensions réduites (21 x 14 x 8 cm), porte un petit écouteur permettant de tester une oreille après l'autre et de faire :

- le relevé d'un audiogramme simplifié entre 200 Hz et 10 000 Hz, les fréquences sont continuément variables, mais on se contente généralement de relever la sensibilité de chaque oreille pour 200 - 1000 - 3000 - et 6000 Hz,
- le relevé de la fréquence de coupure aiguë pour chaque oreille, entre 6000 et 20 000 Hz,
- le relevé du pouvoir séparateur temporel grâce à un générateur de rafales de clics, où les distances inter-clics sont réglables à loisir entre 2 et 250 millisecondes, on demande au sujet de compter le nombre de coups perçus.

RÉSULTATS ET CONCLUSIONS
On a fait ces tests systématiquement avec 350 enfants d'âge scolaire, parmi ceux-ci près de 10% avaient des difficultés scolaires d'apprentissage de la lecture et de l'orthographe. Personne n'avait remarqué que ces sujets étaient des malentendants. On a fait les mêmes tests avec trois groupes de 30 étudiants, jeunes, musiciens. On a vérifié que les "goûts" particuliers du point de vue musical se justifiaient largement en regard des relevés de tests fréquence-temps. Le pouvoir séparateur temporel semble jouer ici un rôle particulièrement important, tant en ce qui concerne la perception des timbres (transitoires) et celle de la hauteur des sons (justesse des instruments). Ce test fréquence-temps permet donc de dépister les enfants malentendants ayant des problèmes scolaires, et de définir si tel sujet possède ou non une "oreille musicienne".

RÉFÉRENCES
LEIPP (E) et CHIRON (D) : Dépistage des enfants malentendants à l'école élémentaire. Publication interne du Laboratoire de Mécanique de l'Université de Paris VI à paraître dans "Revue d'Acoustique" ; Paris.
LEIPP (E) La machine à écouter. Essai de psycho-acoustique Masson, Paris (1977)
MESURE DES FUITES ACOUSTIQUES EXISTANT LORS DE L'EMPLOI NORMAL D'UN ECOUTEUR TELEPHONIQUE

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INTRODUCTION : Le but de l'expérience était de déterminer les fuites acoustiques liées à la position du combiné téléphonique par rapport à l'oreille humaine, lors d'une conversation téléphonique ordinaire. Elle doit permettre d'apporter une contribution aux travaux du CCITT dans ce domaine. Nous avons donc considéré comme "fuites acoustiques" la différence des niveaux de pression mesurés : l'un avec l'écouteur posé sur l'oreille artificielle normalisée CEI/CCITT (1), l'autre, relevé au "point de référence oreille", le combiné étant utilisé normalement. Cet affaiblissement de couplage entre l'écouteur et l'oreille est nommé $L_p$ dans le tome V du Livre Vert du CCITT.

PLAN D'EXPERIENCE : Au point de référence "oreille" (centre du cercle de tangente à l'écouteur) la courbe efficacité-fréquence de l'écouteur a été relevée. Afin de se placer dans des conditions proches de la réalité, le sujet écoutait les signaux suivants : - deux minutes de parole à faible dynamique afin que le sujet règle sa position en fonction du niveau d'écoute. - un signal sinusoïdal à fréquence glissante de 200 à 4500 Hz servant de signal de mesure. Les mesures ont été réalisées avec 8 sujets (4 hommes et 4 femmes), pour 3 niveaux d'écoute de la parole (52-62-72 dB(A)) et 3 passages par niveau et sujet. Le signal de mesure était recueilli au point de référence "oreille" par un microphone de 3 mm, filtré continuellement, puis numérisé afin d'en faciliter le traitement statistique. Le signal de mesure était relevé de la même façon lorsque l'écouteur était placé sur oreille artificielle. Par différence, on en a déduit la valeur $L_g$.

RESULTATS : La figure montre que les valeurs obtenues sont en assez bon accord, à partir de 400 Hz avec celles de G. MODENA et A. REOLON (2) (condition "normale"). Elles s'écartent notablement des mesures relevées par D.L. RICHARDS (3). Ces écarts soulèvent la question suivante : "Est-il possible de déterminer une courbe universelle de fuites acoustiques, indépendante du niveau d'écoute et du type d'écouteur ?". En effet, l'observation de la dispersion de nos résultats fait apparaître une distribution non normale mais plutôt rectangulaire, ainsi que l'a mentionné N. GLEISS (4). Pour ce qui concerne le "niveau d'écoute", les résultats obtenus permettent d'affirmer qu'il n'influence que faiblement (1-2 dB) sur l'allure de la courbe de fuite. Quant à l'influence des différentes formes d'écouteurs, une étude complémentaire portant sur un certain nombre de types différents est nécessaire.

CONCLUSION : A condition que les études complémentaires en cours mettent en évidence l'existence d'une fuite "moyenne", ces résultats permettront de progresser dans l'étude d'une méthode de mesure objective des équivalents de référence (e.r) téléphoniques. Des travaux au CNET à Lannion ont montré que le signal de parole normalisé (Paris, Bordeaux, le Mans...) pouvait être remplacé, lors de la mesure des e.r à la réception, par un signal de bruit stationnaire dont le spectre est identique au spectre moyen de la phrase normalisée. Il s'agit maintenant de vérifier si les méthodes normalisées de détermination de la sonie des bruits stationnaires conduisent à un bon accord avec la détermination subjective des e.r, lorsque le facteur $L_g$ est pris en compte.

STANDARDIZATION OF AN EAR SIMULATOR FOR THE CALIBRATION OF INSERT EARPHONES

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INTRODUCTION

The Working Group 6 of IEC SC 29 C worked on the Standardization of an ear simulator for insert earphones since 1971. This simulator should cover the frequency range from 20 Hz to 10000 Hz and allow a more accurate exchange of specifications and physical data of earphones than is possible with the well-known IEC reference coupler (1).

The main requirements for the simulator are

a) material to be stable in time
b) geometrical similarity with the real ear
c) simulation of the acoustic load presented to the earphone by the real ear.

Several authors proposed devices which fulfill those requirements (Zwislocki (2), Diestel (3), Shaw (4), Brüel (5), and Zuercher and Burkhard (6)). However a standardization is desirable, to make data exchangeable.

PROPOSAL

In March 1976 the Working Group met in Washington, D.C., and agreed on the following principles:

a) The properties of the simulator should be described in a simple way, if possible by volumes.

b) The damping should be described by the ratio between the maximum and the minimum sound pressure in a standing wave.

c) The simulator should be usable also with the full-length "ear canal". Therefore damping material in the outer part of the "ear canal" should be avoided.

d) The same simulator should be used both for insert earphones and with some extension for supraaural and circumaural earphones.

e) If possible, more than one specific example should be given in the appendix.

Preliminary data for the simulator were proposed at the Working Group meeting. The quotient of the sound pressure at the microphone to the volume velocity at the entrance is described in terms of a generalized volume. These data are presented in detail and discussed.

REFERENCES

TACTILE AIDS FOR THE DEAF

Jongens, A. W. D.  Central Acoustics Laboratory,  University of Cape Town

INTRODUCTION

People with normal hearing are able to conduct a conversation by employing two sensory channels: the auditory channel being the main one, and the visual channel, in the form of lip reading, supplementing it.

When speech information via the auditory channel is reduced, as is the case for the aurally handicapped person, more information via the visual channel is used and, for the severely handicapped, this becomes the main information channel.

The profoundly deaf, however, experience great difficulty in lip reading and thus in speech communication as a whole.

It has been shown that the severely hard of hearing can be proficient lip readers provided that a minimal amount of aural information is received by one of the senses (Jongens, to be published).

It is felt that the tactile sense can be used to provide this information to the profoundly deaf person.

DESCRIPTION OF APPARATUS

A tactile instrument was developed at U.C.T. covering the frequency range of 400 Hz to 3000 Hz. The input from a microphonic is amplified before being applied to eight parallel 300 Hz bandpass filters. Each 300 Hz frequency band is converted to the most sensitive tactile frequency band of 100 Hz to 400 Hz and then applied to an electromagnet driving 20 closely spaced tuned steel reed vibrators. These serve as 15 Hz narrow band analysers and as transducers which are applied to all fingers of the left hand.

RESULTS

Experiments have been conducted with this instrument on a profoundly deaf young adult and a profoundly deaf child.

The adult underwent a formal learning program and could distinguish vowels and diphthongs with little training and consonant-vowel combinations quite easily after a few hours practice using only the sense of touch.

The child used the instrument at home and during school classes without formal training. Within two weeks a marked improvement in the child's overall school progress, and especially in his reading ability, was noted.

It soon became apparent, however, that a more portable device was necessary as the original device restricted the physical movement of the child who, in spite of its usefulness in communication, tended to reject it after a few months use.

Development of smaller, portable devices necessitated a reduction in the number of vibrators to forty, and the upper frequency limit to 1000 Hz.

The modified devices, which are applied to the palm of the hand, are being used by children at a school for the hard of hearing.

Initial results indicate the same degree of progress is being made as with the original device.
Within the past years acoustic imaging systems based on the use of arrays of transducer elements have undergone a rapid stage of development. With such arrays, large focusing aperture may be synthesized. This paper presents a theoretical study of the focusing properties obtained with different focusing techniques.

For each of them lateral and axial resolution, depth of field and side lobe level will be discussed.

Two main techniques are used.

I. The use of delay lines allow a converging one dimensional phase lens to be simulated (I).

When the aperture is sampled by a sufficient number of transducers a very good resolution with a low side lobe level is obtained using a short emitted signal.

However these devices give a short depth of field.

II. Focusing property of Fresnel zone plate may be used with linear array. In the aperture each transducer is commuted in phase or in antiphase to simulate a one dimensional holographic lens (II).

The resolution is then as good as the one given by the precedent technique, but side lobe level is much higher. Various apodization process are presented. The importance of the directivity of the elementary transducers is emphasized. Great improvements may be obtained by using different apertures at emission and at reception.

Moreover it will be shown that for such a focusing technique, a large depth of field may be obtained when using a short emitted pulse.


(II) P. ALAIS and M. FINK, Fresnel zone focusing of linear arrays applied to B and C echography.
INTRODUCTION
The increasing demand for action regarding community noise and the new legislation mandating the measurement of noise, both with respect to annoyance and hearing damage potential, has required the development of new, advanced statistical processing equipment that still is easy to use for the inexperienced operator. In this paper, a battery-operated statistical processor will be described.

GENERAL REQUIREMENTS
Since community noise measurements must be made over long time periods, it is important that the instrument has a wide dynamic range to accommodate the widely varying levels. In addition, since measurements must be made in the field, portability, battery operation, and protection against inclement weather must also be considered. For operating convenience and to remove the possibility of human calculation error, the instrument should automatically process the data and calculate the final results, without any need for manual calculation or post-processing. Finally, provision for automatic documentation of the results, both in numerical and graphical form should be made.

DESCRIPTION OF THE ANALYZER
The analyzer consists of a precision sound level meter with digital display along with two digital processors operating in parallel. One processor handles the statistical functions such as the probability distribution and cumulative distribution calculations while the other processor determines the equivalent energy level, $L_E$. Input to the instrument may be from a precision condenser microphone with preamplifier or a direct electrical source, while output is on the built-in digital display. Additional outputs are provided for a battery-operated graphic level recorder and alphanumeric printer.
INTRODUCTION

The recent upsurge in noise legislation has led to the need for reliable estimates of precision likely to be achieved in measurements of noise level. One aspect is the inherent precision of sound level meters (SLMs) and how adequate traceability to laboratory standards can be achieved. Reference is usually made to the use of a precision SLM complying with IEC 179 (2nd Edition):1973 but this is a type-test not a calibration procedure. In the case of the associated acoustical calibrator there is no recognized specification at all. A study was therefore undertaken to establish the spread in SLM/calculator-calibrator performance and then to produce minimum acceptance tests and calibration procedures required to produce a stated measurement accuracy when using one particular type of instrument (ie B & K precision SLM type 2206 and sound level calibrator type 4230).

EXPERIMENTAL DATA

Electrical tests were carried out on twenty-five instruments which had been supplied to the UK Health and Safety Executive but which had not yet been issued to their inspectors. The following electrical tests were carried out: A-weighting characteristic at octave frequencies over the range 31.5 Hz to 8 kHz; accuracy of the nominal 10 dB range-change steps over the range 70 to 120 dB for test frequencies of 31.5 Hz, 1 kHz and 8 kHz; scale markings of the meter at 31.5 Hz, 1 kHz and 8 kHz; dynamic characteristics of the indicating instrument on FAST and SLOW using a 1 kHz tone-burst technique; overload performance using a peak-clipping criterion; square-law performance using non-harmonic frequency-pairs. Overall acoustical calibration of each SLM was also carried out at a level of 70 dB(A), using pure tones at octave frequencies – from 31.5 Hz to 250 Hz in a duct, and from 500 Hz to 8 kHz in a free-field room. The SPL generated by each of the acoustic calibrators was also determined. For each of the tests a number of repeats were made over a period of several weeks to produce estimates of replication variance.

CONCLUSIONS

Deviations from IEC requirements almost all lay within the permitted tolerances. However, it is not obvious what effect these deviations may have on accuracy when using the meters for their intended purpose of measuring A-weighted levels of industrial noise. Results for each meter were therefore combined with octave band levels of a wide range of industrial noises and the resulting A-weighted sound levels derived by calculation. Whereas meters complying with IEC 179 could give readings differing from the true value by ±2 dB(A), the sample of meters tested would lead to a spread of only ±0.41, ±0.55 dB(A) over the range 82 to 110 dB(A). Thus calibration errors will probably be smaller than those inherent in most field samples of time-varying noise. A direct check of SLM performance by exposing them to a broad-band 'pink' noise confirmed the above conclusions and indicated an rms error of 0.24 dB(A). It has been possible to recommend minimum acceptance tests and periodical (annual) recalibration tests which should take less than 15 min per instrument; further, a minimum user check procedure has been recommended (before and after each critical noise measurement). Whilst certain of the test specifications and tolerances are specific to the type 2206 SLM, the general principles have more general application and can be extended to cover other types of precision SLM.

A detailed account of this work is available in NPL Acoustics Report Ac 75:1976.
INTRODUCTION

The part in a sound level measurement equipment for the conversion of sound ac signal into dc output is a critical element because it determines both steady state and transient responses as well as dynamic range which is required in data processing or wide range indication. In IEC or other standards, no mention is made about specific circuitry for the above conversion as in the CISPR standard for the EM interference measurements. In the following, various new methods of ac-dc conversion techniques are discussed.

METHODS OF AC-DC CONVERSION

(1) Computing circuit of rms value

The square and square-root operations are usually made by analog or digital ICs. The latter operation is omitted in case of linear dB output by means of logarithmic conversion. The required dynamic range of the squared quantity is twice of that of input and the achievable accuracy and dynamic range depend on the employed ICs.

(2) Proportional piece-wise rms approximation rectifier

In this method, the absolute value of input and its divided voltages are fed into a common averaging condenser with a discharge resistor through rectifier-resistor series circuits. The approximation for rms value by the condenser voltage depends on the number of divisions, division ratios and resistor values. The squared quantity does not appear apparently and the square-root operation is not required. This circuit can be called as a proportional piece-wise rms approximation rectifier in which break points shift with the size of input. By the use of precise rectifying devices composed of analog ICs and/or transistors, the dynamic range has been increased.

(3) Quasi-rms rectifier

When only one charging circuit exists in the preceding case, it is usually called as a quasi-rms rectifier. This has the similar circuit configuration as the quasi-peak detector in the EM interference measurements except time constant. Because of its simplicity, this rectifier has been popularly employed in sound level meters. Its initial step response is slower than the previous cases. By the use of analog ICs and FETs instead of diodes, a dynamic range of 70 dB has been obtained.

Conclusions

Besides the above methods, the rectification of compressed ac signal is used to obtain a wide dynamic range. But in this, the expected approximation for rms value is less than the above cases. The choice of ac-dc conversion method must be made according to the classification of sound level measurements considering the requirements for small power consumption and other factors.

REFERENCES

Inspecting the tolerance levels in IEC's standards for sound level meters, it is seen that, especially for medium high and high frequencies, very large deviations may occur between results from measurements carried out with different sound level meters, which each fulfills the given requirements. This is partly due to the allowed tolerances on the frequency and the directional sensitivity characteristics and partly due to the permission of different microphone features such as flat 0° incidence free-field frequency response or flat random incidence frequency response.

The poorly defined impulse response requirements for the "Fast" and "Slow" detector/indicator modes may also result in appreciable deviations in the results from measurements of noise with some impulse content.

The increased legislation in the noise pollution area and the often very great costs connected to a certain noise reduction, will in the future give rise to increased accuracy requirements and thereby to a sound level meter type with improved performance.

In this paper, the extreme theoretical deviations between measurements with different microphone sizes/configurations are compared to the deviations noted in relation to measuring situations found in practice.

Deviations in measuring results in consequence of different detector/indicator systems are also discussed.

Fig.1. IEC 179 Tolerances for frequency response and directional sensitivity
In this paper, it will be demonstrated that a Digital Real-Time Frequency Analyzer operating with digital filtering and RMS detection techniques is especially well suited to acoustical measurements where measurement with constant percentage bandwidth filtering is required. We are here primarily thinking about the 1/3 octave filters which are used in many standardized measurements. However, the great flexibility of digital filtering makes it possible and relatively simple to change the filtering characteristic to, e.g., 1/1 octave, or, if the requirement for real-time operation is renounced, even to 1/6, 1/12, or 1/24 octave.

A generalized block diagram of a recursive 2-pole digital filter is shown in Fig. 1. The filter properties are a function of the multiplier coefficients $A_0$, $A_1$, $A_2$, $B_1$, and $B_2$. Changes in the transfer function of the filter, e.g. from 1/3 octave to 1/1 octave bandwidth, only require that these coefficients be changed. The transfer function for the filter, can be written as, using $z$-transform notation:

$$H(z) = \frac{A_0 + A_1 z^{-1} + A_2 z^{-2}}{1 - B_1 z^{-1} - B_2 z^{-2}}$$

With digital RMS detectors, it is possible to overcome many of the limitations of equivalent analog systems, e.g., dynamic range and crest factor. Possibly the most important feature is that linear averaging becomes feasible. This form of averaging is carried out over a fixed period of time, and the process may be represented by the following equation:

$$A_r = A_{r-1} + \frac{T}{K}$$

where $A_r$ is the averaged value, $T_r$ is the r'th sample, and $K$ is the total number of samples.

It will be demonstrated that with this system, an ideal transient response is obtained, that is, measurements of transients and impulses will be quite repeatable and give results exactly as theory predicts.

Further, the flexibility of the system when operating with a control unit, e.g., a Desk-Top Calculator, will also be demonstrated.
Bei der Spektalanalyse akustischer Vorgänge mittels eines durchstimmbaren Schmalbandfilters können unerwünschte große Messzeiten entstehen. Für die überwiegend benutzten Filter mit konstanter relativer Bandbreite wird deshalb die Funktion $f = f(t)$ gesucht ($f_1 = $Filtermittenfrequenz), bei der die Analysierzeit auf einen optimalen Wert verkürzbar ist.

Die Bandbreite $\Delta f$ eines Frequenzfilters ist ein Maß für die Einschwingzeit $\Delta T$ dieses Filters auf den Meßwert:

$$\Delta T = k \cdot 1/\Delta f$$

Der Proportionalitätsfaktor $k$ hängt von verschiedenen Einflußgrößen ab wie Flankensteilheit, Niegengenaugigkeit, Form und Determiniertheit des Spektrums und soll zusätzlich die Eigenschaften des nachgeschalteten Gleichrichters erfassen, wenn dessen Einschwingzeit optimal an die Filtereinschwingzeit angepaßt ist. Für Filter mit konstanter relativer Bandbreite gilt

$$\Delta f = f_m \cdot 1/Q$$

Bei Verwendung des üblichen Gleichrichters mit konstantem Trägheitsglied richtet sich die Meßzeit für jede Filtermittenfrequenz nach der größten auftretenden Filtereinschwingzeit, also der Einschwingzeit $\Delta T$ für das unterste Frequenzband mit der Mittenfrequenz $f_m = f_1$.

Die Zahl $n$ der aneinandergereihten vorstellbaren Schmalbandfilter zur Erfassung des Frequenzbereiches $f_u \ldots f_m$ aus den Ansätzen

$$f_m/f_u = (1+1/Q)^n$$

ergibt, ermittelt man für die Funktion $t = t(f_m)$ aus (1), (2) und (3)

$$t_1(f_m) = t_2(f_m) \Delta T = kQ \cdot n(f_m) \cdot 1/f_u = kQ \cdot \ln(f_m/f_u) \cdot 1/[f_u \ln(1+1/Q)]$$

Paßt man dagegen das Einschwingverhalten des Gleichrichters der jeweiligen Filtermittenfrequenz an, dann erhält man aus (1) und (2)

$$\Delta f/\Delta t = f_m^2/2kQ \approx df_m/\Delta t$$

Mit $f_m(t=0) = f_u$ ergibt sich in der Form $t = t(f_m)$:

$$t_1(f_m) = kQ^2 (1/f_u - 1/f_m)$$

Bei einer Frequenzanalyse ergibt sich das Verhältnis beider Meßzeiten aus (4) und (6) für $Q > 1$ und $f_1 > f_u$ zu

$$t_1(f_m = f_0)/t_2(f_m = f_0) \approx \ln(f_0/f_u)$$

Für den akustischen Frequenzbereich mit $f_u = 20\,kHz$, $f_1 = 20\,Hz$ ergibt sich ein Verhältnis von 6,91. Anhand dieses Beispiels zeigt die Tabelle den Vorteil des zeitoptimalen Betriebs. Der gewählte Zahlenwert für $k$ wird hier nicht näher begründet.

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A TECHNIQUE FOR MEASURING SOUND INTENSITY WITH A SOUND LEVEL METER

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A technique is being developed whereby the acoustic intensity vector at a point in an arbitrary sound field may be measured in octave frequency bands by using two commercially available 6 mm diameter condenser microphones, a special purpose summing, differencing and integrating circuit and a standard portable sound level meter. The well known principle of using the integrated difference signal from two closely separated microphones to obtain an approximation to the acoustic particle velocity is employed: the innovation, if such, consists in the need for only one filter, thereby eliminating filter matching errors, and the cost of twin, matched filters. The primary disadvantage of the technique is the restriction to the measurement of sources which are continuous and steady state, at least in a time average sense.

The circuit is shown schematically in the figure. The outputs from the two microphones separated by a distance $2h$ are applied to sum and difference circuits. Either the sum, or the difference, is integrated and the two signals are again applied to sum and difference circuits after appropriate gains have been applied. The two resulting signals are fed separately through the octave filter of a sound level meter which indicates approximately their time average (mean) square values. The difference between these values is proportional to the acoustic intensity vector in a direction determined by the line joining the acoustic centres of the two microphones.

The corresponding equations are as follows:

$$V_1^2 = A^2 \left( \left( p_1 - p_2 \right) \right)^2 + 2AB \left( \left( p_1 - p_2 \right) \left( p_1 + p_2 \right) \right) + B^2 \left( p_1 + p_2 \right)^2$$

$$V_2^2 = A^2 \left( \left( p_1 - p_2 \right) \right)^2 - 2AB \left( \left( p_1 - p_2 \right) \left( p_1 + p_2 \right) \right) + B^2 \left( p_1 + p_2 \right)^2$$

$$V_1^2 - V_2^2 = 4AB \left( \left( p_1 - p_2 \right) \left( p_1 + p_2 \right) \right) = 16AB \rho h I$$

where $A$ and $B$ are the products of the circuit gains and microphone sensitivities and $I$ is the intensity.

At the time of writing, preliminary measurements in a standing wave tube using both pure tone and random noise have indicated reasonably satisfactory performance. Later developments will be reported at the Congress, in which machinery noise measurements using this technique are compared with the results of digital signal processing, using the relationship $I = \int (Q(p_1, p_2)/2)df$, where $Q$ is the imaginary part of the cross spectral density of pressure signals. In principle any degree of frequency resolution can be obtained by using a single swept band pass filter squaring and averaging circuit and recording system. However source conditions would have to remain steady over the total period of recording, which, for analogue systems would increase in inverse proportion to filter bandwidth for adequate statistical accuracy.
THE ACCUMULATING EQUIVALENT SOUND LEVEL METER

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Fukuchi, Y.
Maune, Y.
Rikow, Y.

INTRODUCTION

There are many kinds of instruments which can get the noise exposure, so called the noise dose meter. These meters make only available a continuous measurement of the total exposure, but they can not get the successive variation of the noise exposure. However, for the fundamental data to investigate the influence of the noise pollution to human life in detail, it is important to know the variation of the noise exposure in relation to time and activity in daily life.

The newly developed instrument described here has faculty to measure, accumulate automatically and read out the equivalent sound level \( L_{eq} \) in dB every 10 minutes through whole day (24 hours, 144 data).

BLOCKDIAGRAM and FEATURE

The instrument consists of two parts as shown in Fig. 1. One is the portable accumulating equivalent sound level meter (Leq meter) and the other is the specialized data processor. The Leq meter conforms to equal energy principle (ISO 1999). The "A" weighting network and detector are similar to those used in Sound Level Meters (JIS-C1502, IEC R 123). The features of this instrument are as follows. 1) portable size (Leq meter), 2) measuring 24 hours and holding data 72 hours (4 AA size batteries), 3) wide dynamic range 40-110dB, 4) printing out Leq (10 minutes, a day or arbitrary time zone) in dB.

MEASURING EXAMPLE

The variation of a press worker's Leq is shown in Fig. 2. In this case, the total Leq (24 hours) was 81 dB.

CONCLUSION

This instrument can be used not only studying the variation of the noise exposure every 10 minutes but evaluating conveniently the general environmental noise.

We'll be able to get the data for the conservation of hearing and environment.
INTRODUCTION

La dosimétrie des bruits continus ou lentement variables est de nos jours parfaitement au point (1-2). En revanche pour les bruits impulsionnels, aucune technique spécifique n'a encore été proposée.

Dans cette étude nous exposons les fondements physiques et physiologiques d'une technique adaptée à ces bruits, ses caractéristiques et une synthèse portant sur une expérience menée sur de nombreux bruits industriels.

BASES THÉORIQUES

La dosimétrie est liée directement au niveau continu équivalent :

$$\text{Leq} = -10 \log \frac{1}{T} \int_0^T \left[ \frac{p(t)}{p_0} \right]^q \, \text{dt}$$

$p(t)$ : pression acoustique instantanée  
$p_0$ : pression de référence $2 \times 10^{-5}$ Pa  
$T$ période de référence : 8 heures par jour ou 40 heures par semaine  
$q$ : coefficient que nous prenons égal à 3 pour satisfaire au principe d'isoénergie.

Dans ce cas on peut également exprimer la dose en Z.

PRINCIPE

En dosimétrie classique, on utilise la courbe de pondération A. Or pour les bruits impulsionnels souvent riches en fréquences graves, cette pondération affaiblit considérablement l'incidence de ces fréquences. Or ce sont elles, nous l'avons souligné dans une étude précédente (3), qui sont souvent à l'origine de la nocivité spécifique de ces bruits. D'où la nécessité de conserver dans ces cas, les énergies spectrales réelles.

La technique que nous proposons consiste à supprimer l'insertion du filtre A dès qu'un bruit impulsionnel a été identifié. Cette identification se fait à partir de trois critères : intensité, durée et soudaineté.

SYNTHÈSE

L'étude faite à l'aide de notre dosimètre expérimental nous permet de retenir les valeurs suivantes :  
- durée de commutation : 100 ms  
- critère de dépassement : 10 dB dans les ambiances à faible niveau, 5 dB dans celles à fort niveau  
- pente de discrimination : 0,18 Pa/us

CONCLUSION

Cette technique constitue une première tentative visant à inclure quantitativement grâce à une mesure directe l'influence spécifique des bruits impulsionnels. On a pu ainsi mettre en évidence pour certains bruits des différences notoires pouvant aller du simple au triple entre la dose classique et la dose impulsionnelle. Pour la plupart des bruits, les écarts sont faibles à cause d'une compensation entre fréquences graves et fréquences aiguës.

(1) Recommandations ISO R 1996 et 1999  
(2) Arrêté du 12 août 1972  
(3) R. UNTERREINER, Acustica 1974, Vol 30, 100
DYNAMIC CALIBRATION OF TEMPERATURE WIRES BY MEANS OF A SOUND FIELD

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INTRODUCTION

Measurements of fast temperature fluctuations in gases are usually performed by measuring the temperature-induced resistance changes in small wire probes. The time response characteristics of the probes can be theoretically determined from the heat balance equation. Experimentally, the most used technique is to study the response of the probes to an internally generated electrical heating. A problem with this method is that while the probe in a measuring situation responds to temperature fluctuations arising from the surrounding medium, its characteristics are determined from temperature fluctuations generated within the wire. In frequency ranges where boundary layer effects are important for the response, their influence may be different in the two situations.

To circumvent this problem the authors of the present work have chosen the direct method of studying the response of the probes by exposing them to a sinusoidally oscillating temperature field of continuously variable frequency. The temperature oscillations are generated by a strong sound field as a result of the relation between temperature and pressure as stated by the equation of state of a gas.

MEASUREMENTS

In still air the measurements were performed in a small cavity (volume 6.5 cm$^3$) attached directly to the opening of a horndriver unit. By means of a compressor microphone a constant sound pressure level of 134 dB re 20 μPa was obtained in the frequency range from 3 Hz - 10 kHz corresponding to a temperature amplitude of approx. 0.1 K. At low frequencies the influence of heat conduction effects on the temperature field shall be taken into account [1]. A typical response of a wire probe is shown in the figure below.

![Graph showing typical response of a wire probe](image)

DISA Pt-wire: diameter 1 μm, length 0.7 mm, current 1 μA.

Correspondingly, measurements with a superimposed air flow were carried out in a restricted frequency range (40 Hz-8 kHz) using a tube of length 1 m with a termination.

RESULTS

Measurements have been carried out on a number of probe wires, in the diameter range 0.25-5 μm and the length range 0.2-3 mm. [1].

Two unexpected results were obtained:

a) the prongs holding the active wire attenuated the temperature oscillations considerably at low frequencies,
b) the upper-limit frequency of the probes did not increase with decreasing diameter of the wire as foreseen theoretically for diameters smaller than about 0.5 μm.

Point a) is explained by a refinement of the already existing theories [1], and the effect vanishes for a superimposed air flow of a velocity higher than 1 m/s.

Point b) is unexplained so far.

REFERENCES

SIMPLE, PORTABLE INSTRUMENTATION FOR STATISTICAL NOISE MEASUREMENTS

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INTRODUCTION

Instrumentation for the measurement of statistical characteristics of noise has been available on the market for many years now. However it has been generally bulky instrumentation and is by no means portable. The more recent digital sampling and processing has solved the portability problem but has done nothing to cut the price of a statistical analysis. This paper describes a simple, inexpensive, portable system for the measurement of statistical parameters. Although basically applied to the determination of $L_{10}$ and $L_{90}$ in dBA, there is no reason why the technique should not be applied to $L_N$ of statistical distribution analysers of broad band (weighted or linear) measurements. Two forms of the equipment are commercially available, each with the alternative of battery or mains power; the first consists of a precision sound level meter and separate recorder; the second is an integrated package enclosed in a weatherproof case designed for regular outdoor use. Alternatives of 20dBA logarithmic or 50dBA linear display are available in both cases.

THEORETICAL BASIS

Although many alternative measurements can be considered, the case described in this paper is measurement of $L_{10}^{10}$ and $L_{90}^{90}$ as required in the British "Land Compensation Act" legislation. $L_{10}$ is defined as the level (in dBA) which is exceeded for 10% of the measurement time. Similarly $L_{90}$ is the level which is exceeded for 90% of the measurement time.

MEASUREMENT

In principle, the measurement of $L_{10}$ can be made by storing the reading of a sound level meter at regular (say 10 second) intervals, writing down the result, and after say 100 readings crossing out the 10 highest, the 11th highest being $L_{10}$ over 16.67 minute measurement period. Clearly for the long term measurement this is terribly tiring and impractical for all but very occasional use. Consequently a sampling chart recorder is used to produce a dotted trace. The recorder samples the sound level every 3.75 seconds and chart speed can be selected as required for convenience over the measurement period. Usually a 15 minute measurement period is used which consequently contains 240 samples. In the record, $L_{10}$ is therefore the 25th sample from the top of the chart.

Practicality and portability of the system will be demonstrated at the Congress and it is hoped that several examples on its use to the date of the Congress will be available. Whilst the system is by no means perfect it achieves with little extra labour any statistical analysis or distribution which can be obtained with equipment of 3 to 5 times its price.
Les problèmes liés à la mesure des phénomènes acoustiques sont loin d'être parfaitement résolus, en particulier lorsqu'on espère relier l'indication de l'appareil de mesure à la sensation auditive. Faute de certitudes dans ce domaine, il a fallu adopter des conventions aussi proches que possible de la réalité supposée des mécanismes perceptifs. La détection quadratique qui a été adoptée puis normalisée pour assurer la comparativité des mesures répond à cet aspect du problème. Il en est de même des constantes de temps normalisées.

Une étude critique de la chaîne de mesure acoustique traditionnelle montre clairement que nous ne pouvons pas espérer disposer de circuits électroniques permettant de réaliser un sonomètre dont la dynamique de mesure serait égale à celle de l'oreille. Ceci n'est envisageable qu'à l'aide d'un changement de calibre incompatible avec l'utilisation de cet appareil pour le contrôle de l'environnement.

L'étude de principe d'un circuit compresseur de dynamique réalisant la fonction logarithmique imposée par l'utilisation du décibel montre qu'un tel dispositif est apte à effectuer la détection des niveaux sonores sur une dynamique supérieure à 80 dB.

Une étude théorique montre que sous certaines réserves, la réponse d'un compresseur de dynamique peut donner des résultats comparables à ceux d'un sonomètre traditionnel.

Suivent quelques exemples de réalisations industrielles permettant d'utiliser ce type de dispositif soit dans des sonomètres portatifs donnant une indication linéaire en décibel, soit dans des centrales de mesure des grandeurs d'environnement, le résultat de la mesure pouvant être analogique ou numérique.

De par leur simplicité et leur commodité d'emploi, ces dispositifs semblent devoir donner naissance à une nouvelle génération d'appareils de mesure des bruits.
A CALIBRATED CAPACITANCE TRANSDUCER FOR USE IN ACOUSTIC EMISSION EXPERIMENTS

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INTRODUCTION

A current trend in acoustic emission technology is towards calibration of transducers such that data can be suitably quantified for universal interpretation. With this in mind this paper introduces a form of capacitance transducer which directly yields the absolute displacement of a specimen surface when an acoustic emission waveform reaches it. By calibrating this device with a laser interferometer it is further possible to obtain absolute surface displacement as a function of frequency.

PRINCIPLE OF PERFORMANCE

The transducer is constructed by a combination of electroless, electrolytic and vacuum deposition metallising. A circular thin glass optical flat is metallised with three nickel 'feet' such that when placed upon a flat polished surface the gap formed is of the order of 5 x 10^{-3} mm thick. Between these feet at the centre of the plate, an area 10mm in diameter is thinly vacuum metallised. The metallising is brought to the edge of the disc in a thin strip where it overlays a thicker strip of metallising which rounds the edge of the plate so that an electrical connection can be made to it.

The flat polished specimen surface forms the earth plate of an air capacitor, the active plate of which is the thinly metallised central disc. If this capacitor is connected to a charge amplifier, and is provided with a dc voltage of 1kV, then any stress wave pulse, of amplitude \( d \sin \omega t \), striking the surface beneath the active plate will cause a voltage pulse, \( e_\alpha \sin \omega t \), to flow from the charge amplifier according to:

\[
e_\alpha \sin \omega t = \frac{C_\alpha E}{C_t d_\alpha} d \sin \omega t
\]

where \( \omega = \) angular frequency; \( d_\alpha = \) thickness of the air gap; \( C_\alpha = \) input capacitance of the charge amplifier; \( C_t = \) capacitance of the transducer; \( E = \) d.c. voltage applied to the transducer.

(This principle has been used previously with flexible, self adherent, capacitance transducers). By determining the electrical parameters of the transducer and amplifier the absolute displacement waveform can be calculated if the thickness of the air gap is known.

To determine the air gap thickness all that is required is that parallel white light be directed normally through the transducer surface and the reflected light observed through a spectroscope. A white light spectrum is derived containing a regularly spaced array of dark fringes, (Edser-Butler fringes). The spacing of these fringes yields the air gap thickness according to:

\[
d_\alpha = \frac{A_n}{2(\lambda_1 - \lambda_2)}
\]

where \( A_n \) is the number of fringes between wavelengths \( \lambda_1 ; \lambda_2 \).

CONCLUSIONS

The transducer is currently undergoing trials on a tensile specimen specifically designed to permit observation of the direct emission received from fracture in the gauge length.
INTRODUCTION

In this paper, acoustic load dependency of electroacoustic transduction efficiency of electrostrictive transducer is derived theoretically and is further measured experimentally. It is resulted that there is an optimum acoustic load resistance which maximizes the efficiency.

THEORETICAL BASES

The fundamental formula of electroacoustic transduction (at A type resonance) is expressed as follows:

\[ v = \frac{I}{Z_q + j\omega} = \frac{V}{Z_q + j\omega} \]  

where \( V \) is the voltage of the transducer, \( I \) the current, \( Z_q (=\sigma + j\omega) \) the mechanical internal impedance, \( \omega \) the vibrational velocity and \( F \) the external force at the acoustic terminal. Now, in case when \( F = 0 \), it becomes \( V = \frac{I}{Z_q + j\omega} \) and \( I = \left( \frac{\sigma}{\omega} + j\omega \right) \) \( \omega \). Then, the input admittance of the transducer becomes \( \frac{V}{I} = \frac{\omega}{\sigma + j\omega} \). The electroacoustic transduction efficiency (at A type resonance) is expressed as follows:

\[ \eta_{ea} = \frac{\omega}{\sigma + j\omega} \]  

From eq. (2), \( \eta_{ea} \) comes to maximum when the acoustic load resistance is

\[ r_a = \frac{\omega}{\sigma + j\omega} \]

EXPERIMENTAL TECHNIQUES

The electroacoustic transduction efficiency, i.e. the ratio of acoustic output power to input electrical power, is measured with acoustic dummy load using high frequency wattmeter. The acoustic output power is calculated according to the formula

\[ P_a = \left( P_e - P_{el} \right) \]  

where \( P_{el} \) and \( P_{en} \) are the value of input electrical power in loaded condition and at no load respectively, measured at resonance frequency and at the same vibration velocity, and \( P_e \) and \( P_{en} \) the value of dielectric loss power also at load and at no load. The electroacoustic transduction efficiency and the acoustic load resistance is further calculated according to the formula

\[ \eta_{ea} = \frac{P_a}{P_{el}} \]  

where \( v_{eff} \) is the effective value of the vibration velocity of the transducer.

RESULT AND CONCLUSIONS

Experiments above are carried out using the 28 MHz transducer. And the result is shown in Fig. 1. The measured values agree well with the theoretical curves.

REFERENCE

INTRODUCTION

Several different methods have been used to calibrate AE-transducers. Transient pulse methods (using electrical sparks, breaking of pencil leads etc.) have been used along with continuous signal calibrations like ultrasonic face-to-face calibration.

Some of these methods have been studied, especially a reciprocity calibration technique proposed by Hatano [1].

Normal commercial transducers are using a rather big (~20 mm) piezoelectric disc, coupled to the structure through a wearplate and glue or grease. This kind of coupling implies many problems, especially at small wavelengths and/or rough surfaces. How their response depends on the vibration mode and coupling is not known.

By using a special construction with a very small piezoelectric disc, suspended by means of a thin membrane, and with a rounded probe-like coupling to the structure, most of the difficulties can be overcome.

EXPERIMENTAL TECHNIQUES

The reciprocity calibration of AE-transducers requires Rayleigh waves on the surface of a semi infinite medium. This is fulfilled by using a thick plane-ground steel plate with damping over as much of the surfaces as possible. One transducer is used as sender and excited by a swept, warbled sine signal.

To get the maximum output from the special low capacitance transducer, a special low noise preamplifier is built into the transducer.

RESULTS AND CONCLUSIONS

A comparison between reciprocity, glass capillary breaking and spark bar calibration on the same transducer is shown in fig. 1. The reciprocity method fulfills the requirements to an absolute, repeatable, broad frequency calibration.

Fig. 2 shows a comparison between the new transducer type and some other transducers made by the reciprocity method. It is seen that it has a high sensitivity and a broad frequency range which makes it useful for many AE-measurement purposes.

---

Fig. 1: Reciprocity, Glass capillary, Spark bar
Fig. 2: A: B&K, type N, no 2.1
B: Commercial, flat
C: B&K, type D, no 2
D: Commercial, resonance

(1) H. Hatano and E. Mori, JASA (1976), 59, (2), 344
THEORETICAL BASIS

In earlier works we have indicated the existence of anamorphical relationships among different states of a signal in its propagation, and how these relationships allowed an analysis of the characteristic magnitudes of the source. Generally these relationships are also function of the propagation media an receiver.

In this paper we have considered linear anamorphical relationships such as \[ L(x,\omega, t) = A(x,\omega) + L_1(x,\omega, t) \] between the instantaneous levels of the acoustical signals \( s_1(x, t) \) and \( s_2(x, t) \). These relationships have special interest in the study of propagation phenomena where the influence of the remaining acoustic magnitudes can be considered negligible because of physical reasons. It has permitted us to study:

a) receiver properties, establishing a new secondary method for TRANSDUCER CALIBRATION,
b) properties of the medium i.e. the VIBRATION TRANSMISSIBILITY

Nevertheless the joint effect of several magnitudes are of interest in some cases as in the EQUALIZATION CURVE of personal microphones also studied in this work.

EXPERIMENTS AND CONCLUSIONS

Secondary free field calibration of microphones.

Fig. 1 shows the relative response of two microphones (B&K, 1") placed in two identical points of an acoustic field produced by a 4" loudspeaker radiating pure tones in anechoic room. Compared with the classical feedback method it is worth pointing out that the anamorphical method (AM) presents a noticeable increase of both, frequency band and dynamic range (>50 dB). It is also worth to note that this procedure does not involve additional distortion as it occurs in classical methods, specially in the low frequency range, due to the low efficiency of the loudspeaker.

Transmissibility of a mass-spring system. Figure 2 summarizes the experimental results. The gain in quality and dynamic range achieved with AM is noticeable. Moreover it permits a better excitation conditions and a sweep frequency velocity up to ten times greater.

Frequency response of personal microphones. (Fig. 3)

Under normal speech conditions and successive automatic filter sweep in 1/3 octave bandwidth steps, the equalization curve for a Lavalier microphone was obtained. The basic experimental conditions involved a standing speaker in one anechoic room and two microphones (same to Fig. 1) placed 27 cm apart from the speaker's mouth, one in front and the other fastened on the speaker's chest.

In the above mentioned cases, the AM methods are specially efficient and well suited even if the nature of the signals is intermittent or impulsive, instances in which traditional methods became inadequate.
INTRODUCTION The acoustic impedance of laboratory microphones is usually expressed by the equivalent volume of the air \( V_e \). The volume of the air in the microphone body in front of the diaphragm, usually called the front volume \( V_f \), is not included into the equivalent volume in this case. Often, however, the equivalent volume as measured by some methods, can involve so called difference volume \( V_d \), which is the difference of the actual and nominal front volume values. With condenser microphones it is possible to compute the equivalent volume with good accuracy for an arbitrary frequency on the basis of a few simple measurements.

THEORETICAL BASES From the basic theory of condenser microphones the following relation can be derived:

\[
\frac{\nu}{\nu_e} = \frac{V_f}{V_e} + \frac{V_d}{V_e} = \frac{\nu^2_0 c_a (1 - \omega^2 c_m a_m)}{1 - \omega^2 c_a a_m}^2 + \omega^2 \frac{r_c c_a}{\omega^2 c_a a_m} + \omega^2 \frac{r_c c_a}{\omega^2 c_a a_m}^2 \tag{1}
\]

In Eq. 1 \( c_a, r_c \) are the basic elements of the acoustic equivalent circuit of the microphone, \( \beta \) is the ratio of the specific heats, \( \nu_0 \) is the static pressure. It is possible to ascertain the values of \( a_m, c_m, r_c \) from the electrical impedance measurements on the electrical terminal of the microphone \( / \) or from the measurement of the acoustic impedance at the resonance frequency and on a very low frequency.

EXPERIMENTAL TECHNIQUE Following measuring methods were used for measurements of the equivalent volume: (1) the electrical impedance method, (2) a comparative method using a source of the constant volume velocity, (3) a reciprocity method (for details see /1/), and a single pulse method (4) (see second contribution of this author). The values of \( a_m, c_m, r_c \) were being computed from measurements by methods (1), (2) taken at \( f_r \) and at \( f_a \). For computations a programmable calculator was used.

RESULTS AND CONCLUSIONS As an example in Fig. 1 are shown the values of \( V_f \) and \( V_e \) measured by method (1) on a microphone \( \& k \) 4160 and computed in the proposed way (by the same method). In the Fig. 2 the plot is for the same microphone (the same \( a_m, c_m, r_c \)) with added \( V_d \) of 22 mm³; in the graph shown are the values measured by method (2). In Fig. 3 shown are the results of measurements by methods (2) and (3) on a microphone with \( V_f = 55 \) mm³ for the plot added is approx. 50 mm³. Good agreement is obtained especially for \( V_e \). The \( V_d \) component usually is less important.

REFERENCE Measuring the Acoustic Impedance of Standard Laboratory Microphones, Report of the Laboratory of Acoustics at DTH, Lyngby, Danmark
L'usage du transducteur électrostatique comme capteur du débit a été décrit en [1]. Sur Fig. 1 on voit un dispositif à mesurer l'impédance acoustique \( Z_{ac} \), constitué d'une membrane mince (tendue par une tension mécanique \( T \)), divisée en deux parties, une partie circulaire (1) et une partie annulaire (2) qui servent respectivement de capteur de pression et de déplacement volumique. L'onde d'électrode annulaire (4) est perforée de façon à ce que la transparence acoustique soit suffisante et permet l'excitation de la membrane annulaire (2) par la pression \( P \). La tension de sortie de la partie annulaire est \( U_a \), la tension de la partie circulaire est \( U_b \). Entre la membrane et l'échantillon de l'impédance, est une cavité dont l'influence est négligeable.

Les deux capteurs ont la même réponse (en phase et en module).

La méthode a été appliquée à la mesure des divers systèmes acoustiques. La Fig. 2 présente un exemple comportant la compliance \( c_a = 4 \times 10^{-4} \text{ m}^2 \text{s} \text{kg}^{-1} \text{Hz}^{-1} \), la masse acoustique \( m_a = 280 \text{ kg m}^{-4} \) et la résistance \( \rho = 175 \times 10^2 \text{ kg s} \text{m}^{-1} \). La Fig. 3 porte un résultat de la mesure automatique comportant le module \( Z_{ac} \) et la phase. Les valeurs calculées (les petits cercles représentent la phase, les petits triangles le module) sont en bon accord avec les courbes obtenues par la mesure automatique.

La limite de l'utilisation en basses fréquences est donnée par des paramètres électriques, la limite d'utilisation en fréquences hautes est donnée par les dimensions des échantillons en comparaison avec la longueur d'onde et par la réponse des capteurs.

La méthode a une large application. Il est possible de réaliser des dispositifs de dimensions externes très différentes (de diamètre) et de chercher les réponses de toutes sortes d'échantillons.

Référence

ON TRANSIENT RESPONSE OF ELECTROSTATIC LOUDSPEAKERS

Merhaut, J. Technical University, Prague

It is obvious to start the theoretical considerations of horn type loudspeakers with an analogue network, in which the cavity below the diaphragm is considered to be a pure compliance. This is good enough for calculation of efficiency and frequency response, but it proved to fail in case of transient response. The author has derived a wave equation for the case of soft, uniformly driven diaphragm (electrostatic forces) by a force \( f_g \) per unit area:

\[
\frac{\partial^2 p(x)}{\partial x^2} + \left( \frac{\omega^2}{c_0^2} - \frac{\gamma}{\gamma D h} \right) p(x) = -\frac{\gamma D h}{\gamma D h} f_g \tag{1}
\]

In eq. (1) \( p(x) \) is the sound pressure in the air chamber below the diaphragm, \( \omega \) the angular frequency, \( \gamma_0 \) the density of the air, \( c_0 \) the sound velocity, \( h \) the thickness of the air chamber and \( \gamma D \) the unit area density of the diaphragm.

By solving the eq. (1) and after insertion \( x = 1 \) (the width of the diaphragm from the support to the entrance of the waveguide) a complex transmission coefficient \( K(s) = \frac{p(x)}{p_g} \) is obtained:

\[
K(s) = \frac{1}{(1 + \frac{1}{\alpha} s^2 + \beta s \sqrt{a^2 + s^2}) \cdot \coth\left( \frac{1}{\gamma} \sqrt{a^2 + s^2} \right)} \tag{2}
\]

The poles of the expression (2) were found by a special procedure, from the curves \( K(j\omega) \) plotted by a table calculator.

The transient response to a unit step (time) is then

\[
R(t) = 1 + D \exp(-\gamma t) + \exp(-\alpha t)(E \sin \beta t + F \cos \beta t), \tag{3}
\]

where \( D, E, F \) are constants and \( -\gamma \) and \( -\alpha \) the poles of (2).

The theoretical response to a rectangular pulse having a width 20 and 40 \( \mu \)s calculated from the eq. (3) corresponds very well with the measured ones.

ELEKTRET-MESSMIKROFONE

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EINLEITUNG:
Für Schallpegelmesser, Schalldosimeter, Schallindikatoren und sonstige elektroakustische Messungen werden in steigendem Maße preisgünstige Mikrofonkapseln mit guten elektroakustischen Eigenschaften und ausreichender Konstanz benötigt.

Elektret-Mikrofone bieten sich für diese Zwecke an. Die bisher auf dem Markt erhältlichen Kapseln entsprechen aber nur teilweise den Anforderungen. Ihre Membranen müssen meist eine Doppelfunktion übernehmen:

AUFBAU DER KAPSELN:

EIGENSCHAFTEN DER KAPSELN:

<table>
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<td>Übertragungsfaktor</td>
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<td>Temperaturbereich</td>
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</tr>
</tbody>
</table>
MODERN POLYMER TRANSDUCERS

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INTRODUCTION

Transducers with prepolarized thin-film polymers have recently found extensive use in a variety of applications. This can be attributed to the favorable properties of the polymer materials, such as long-term charge-storage capability, mechanical flexibility, and availability as thin films. These features are directly responsible for the electrical stability and the excellent electroacoustical performance of the transducers.

POLYMER MATERIALS

Two groups of charged polymers are used in transducer applications, namely electrets and piezoelectric substances. The former group consists of relatively nonpolar, extremely low-conductivity materials, such as the fluorocarbons TFE (Fig. 1a) and FEP. The piezoelectric polymers are highly polar materials, for example polyvinylidene fluoride (PVF₂, Fig. 1b).

Charging of these materials is performed with thermal methods (application of heat and an electric field) or with other procedures and results in real-charge distributions (TFE, FEP, Fig. 1c) and alignment of polar crystalline domains (PVF₂, Fig. 1d).

TRANSUDERS AND APPLICATIONS

Electret transducers (example Fig. 1e) utilize the air-gap field caused by the charged polymer. Such transducers are extensively used as audio-frequency microphones with various directional characteristics, as infrasonic and ultrasonic receivers, and as transmitters in the upper audio and ultrasonic range. Recent transducers to be discussed include touch-sensitive and impact-sensitive devices, electromechanical switches, and miniature directional microphones.

Piezoelectric transducers utilize the longitudinal and transverse piezoelectric effects caused by the polarization of the polymer material. Coupling between the transverse vibrations and the surrounding medium can be improved with dome-shaped diaphragms (Fig. 1f) or bimorph arrangements. Piezoelectric polymer transducers have found wide use in earphones and tweeters, while microphones, phonograph cartridges, ultrasonic and hypersonic transducers and acousto-optic devices are being studied.
A REMOTE CHECKING METHOD FOR NOISE MONITORING MICROPHONE USING Si BULK PELLETS

Miura, K. Matsushita Communication Ind. Co., Yokohama - Japan

INTRODUCTION
A noise monitoring microphone using a piezo-resistive effect of Si bulk has been developed. The long term outdoor installation is suitable for this microphone because of features; stability for humidity, no corrosion on any parts, low output impedance, and small sensitivity deviation for temperature.

A method to check operation or sensitivity of Si bulk microphone is proposed, and its principle and basic characteristics are described.

TRANSDUCER ELEMENT AND DRIVING METHOD
The basic construction of the transducer element is proposed with two Si bulk pellets fixed on both side of a substratum, and the element is used as a cantilever. (See Fig. 1)

When the signal current is supplied to a Si bulk pellet on one side of substratum, the pellet is driven mechanically to bent because of thermal stress with the 2nd harmonics of driving current frequency.

The pellet on the other side produces output signal with its piezo-resistive effect proportional to the applied driving force.

DRIVING CHARACTERISTICS WITH THERMAL STRESS
A practical microphone using two Si bulk pellets was constructed with basic structure as shown in Fig. 1.

As the result of an experiment, the relation between a driving current of one pellet and the sound pressure is shown in Fig. 2.

In Fig. 2 the driving current frequency is 500 Hz, then the output signal frequency is 1000 Hz.

The driving force is given enough to be able to use the checking of the operation or calibration of sensitivity in a noisy circumstance. As in Fig. 2, 10mA of the driving current is equal to 92dB SPL of sound pressure.

CONCLUSIONS
A remote checking method of operation of the noise monitoring microphone, which uses the Si bulk elements is proposed.

This method has single construction, no influence on sound field and easy operation. The basic characteristics were given, and it is confirmed that this method is applied to the practical use for checking or calibration of Si bulk microphone.

REFERENCES
EINLEITUNG

Die messtechnische Trennung dicht nebeneinander liegenden Geräuschquellen ist die schwierigste Aufgabe der Schallmessung. Am meisten ist diese Problem unlösbar. Bei Maschinen Diagnostik wäre aber zweckmäßig die nebeneinander liegenden Quellen messtechnisch so zerteilen, dass die Messwerte und die Energiedichte von einander unabhängig Bewertet werden können. Im Freifeld erreichen wir kein Ziel, wenn die Quellen so dicht nebeneinander liegen, dass die Energiegleichgewichtsschichte sich einander umarmen.

EXPERIMENTALTECHNIK


ERGEBNISSE UND SCHLUSSFOLGERUNGEN


Im Vortrag wird von diesem Messkopf und von der Anwendungsmöglichkeit diskutiert.

LITERATUR

/1/ Dr. T. Kélya, Absonderung von Geräuschelementen mit Hilfe eines Messkopfes, JÁRKÜVEK MEZÓGAZDASÁGI GÉPEK 1972/12.
FINITE PLANE BAFFLES OF IRREGULAR SHAPE

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INTRODUCTION

The effects of plane baffles of irregular shape on the response of a loudspeaker in it are studied using the doublet model.

THEORY

The shape of an irregular baffle is given by the radius vector $R(\theta)$. (fig.1). Suppose that $R$ is $n$-valued, $R_i (i=1, \ldots, n)$, or the baffle is made of $n$ sectors with angle $\theta_i$. Then the back source in the doublet model may be considered to be composed of $n$ virtual sources in effect. (fig.2). Hence the response is given by Eq.(1).

$$ g(\omega) = 1 - \sum_{i=1}^{n} \frac{Z_o}{L(\theta)} e^{-j\omega \frac{\Delta_i}{c} \frac{\theta_i}{2\pi}} \tag{1} $$

When $R(\theta)$ is continuous, summation in Eq.(1) is replaced by integration Eq.(2), and the discrete virtual sources become a line source.

Let one radius vector be drawn in an unite angle. There may be several angular ranges $d\theta_j$ in which the lengths of the radius vectors are between $R$ and $R+dR$. We put the summation of $d\theta_j$ be equal to $d\Theta$.

We change the integration variable of Eq.(2), $\Theta$, to $\Theta'$, on the other hand, $d\Theta/2\pi$ is the rate of numbers of these radius vectors to their total number $2\pi$. We also equate this rate to $p(R)dR$. In Eq.(2), $\Delta$ tends to $R$ and $Z_o/L$ to unity as $Z_o$ tends to infinity. Thus and $p(R)dR$ finally change the integration variable to $R$. Thus Eq.(3) is obtained.

$$ g(\omega) = 1 - \int_{0}^{\Theta} \frac{Z_o}{L(\theta)} e^{-j\omega \frac{R}{c} } p(R)dR \tag{3} $$

Inversely, the $p(R)$, which should realize the given response, is obtained from the inverse transform of the response.

The low frequency limit is given by $R = \int_{0}^{\Theta} p(R)dR$.

---

**fig.1**

**fig.2**
VERGLEICH VON VERSCHIEDENEN TESTMETHODEN ZUR SUBJEKTIVEN BEWERTUNG DER ÜBERTRAGUNGSQUALITÄT VON LAUTSPRECHERBOXEN

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EINLEITUNG

Die Ergebnisse jeder subjektiven Messung der Wiedergabequalität sind u.a. auch von der benutzten Testmethode abhängig. Es wurde versucht, fünf verschiedene Bewertungsmethoden hinsichtlich der Reproduzierbarkeit und Gültigkeit, der Sicherheit von Urteilen, des Zeitaufwands, der subjektiven Schwierigkeit u.a. experimentell zu vergleichen (1).

VERSUCHSDURCHFÜHRUNG

10 Versuchspersonen (Vpn), die auf Grund von Resultaten eines vorhergehenden Auswahlversuchs (2) ausgewählt wurden, beurteilten an 4 kurzen Musikauschnitten die Gesamtqualität der Wiedergabe von 5 verschiedenen Lautsprecherboxen durch die 5 folgenden Bewertungsmethoden: klassischer Paarvergleich, wahlfreier Paarvergleich, Triadenvergleich, Rangordnung und Benotung. Bei den 4 letztgenannten Methoden wurde den Vpn die Möglichkeit gegeben, in den Verlauf des Versuchs aktiv einzumischen. Die Hörtests wurden mit jeder Vpn einzeln in einem akustisch geeigneten Hörraum durchgeführt. Der gesamte Versuchsplan sowie auch die Planung der Einzeltests sollten die am häufigsten vorkommenden Messfehler (wie z.B. den Fechnerschen Zeit- und Raumfehler, die Einflüsse der Einarbeitung und der Ermüdung der Vpn u.a.) in höchstem Maße vermindern, bzw. kompensieren.

ERGEBNISSE

Die Vpn-Urteile wurden in die Rangwerte überführt und durch nichtparametrische statistische Verfahren bearbeitet. Statistisch bearbeitet wurden auch die durch Fragebogen ermittelten Vpn-Angaben über die subjektive Schwierigkeit der Testverfahren und die gemessenen Werte der Zeitaufwände von einzelnen Versuchen. Die Ergebnisse zeigten, dass die subjektiven Bewertungen der Wiedergabequalität bei der Benutzung einer qualifizierten Hö rergruppe mittels jeder der fünf untersuchten Methoden mit ziemlich grosser Reproduzierbarkeit durchführbar sind (die Test- Retest-Reliabilität erreichte die Werte r = 0,9 + 1,0 ). Die rezeitierenden Rangordnungen der Lautsprecherboxen nach ihrer Qualität stimmen gut mit ihrer "wahren" Rangordnung überein, die auf Grund der physikalischen Messungen ermittelt wurde. Daraus lässt sich auf hohe Aussagefähigkeit der subjektiven Bewertungen schliessen. Die durch den Konkordanzkoeffizienten W ausgedrückte Sicherheit der Urteile lag im Intervall W = 0,70 + 0,92.


LITERATUR

(1) A. Melka, Forschungsbericht Nr. 72045/3, TESLA-VUŠT Prag (1976)
(2) A. Melka, 14. tschechoslowakische akustische Konferenz, Tatranská Lomícia (1976), 127
The electret condenser microphone has many inherent advantages in measurement applications over the more conventional air condenser and ceramic types. Considerable effort has been expended in perfecting the design of this microphone over the past several years. Performance is now such that it challenges the other types for nearly all measurement applications. The electret condenser microphone operates on the condenser microphone principle but does not require an external polarizing voltage. The advantages of low sensitivity to humidity, low vibration sensitivity, mechanical ruggedness, and low cost are becoming well known.

The early electret microphone used a diaphragm material that could be formed in a stable electret. The diaphragm was supported by many raised points on a backplate giving the microphone an outstanding resistance against mechanical shocks. The performance at high humidity was also improved with respect to the air condenser microphones. Because of the bound nature of electret charge, voltage breakdown across the air gap cannot occur. Of course, as with any high impedance device, the insulation resistance between the output terminals is subject to the effects of contamination. Early electret microphones show somewhat larger sensitivity shifts with temperature changes than most of the air condenser microphone types when measured at frequencies below 1 kHz. However, in contrast to the air condenser microphone whose sensitivity increases markedly at high frequencies, the electret microphone has a fairly constant sensitivity to temperature as frequency rises.

The temperature characteristics of the electret microphone have now been further improved by separating the electret phenomenon from the diaphragm function. The latest design electret condenser microphone has an electret layer bonded to the backplate while the diaphragm is made of material with best mechanical and thermal properties. Mechanical ruggedness is ensured by diaphragm supporting members bonded to the diaphragm. The temperature coefficient of sensitivity is typically less than +0.01 dB/°C in the temperature range between -20°C and +60°C. Temperature coefficient is independent of frequency throughout the frequency range. A new insulating material is used between the output terminals that is less susceptible to contamination. Long-term stability has been improved.
V. Electroacoustics

R. SIGNAL PROCESSING
SECTIONAL DPCM AND ITS APPLICATION TO THE SPEECH REPRODUCTION SYSTEMS

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INTRODUCTION

Sectional DPCM is a newly developed speech coding method. Digitized speech signals (12-bit PCM) are divided into the sections of approximately 25ms-length. At the sections where signal levels are less than half of the full scale, signal levels are multiplied by $2^k$ ($k = 0, 1, \ldots, 4$), but do not exceed the full scale. Also the selective quantizer with 15 steps is adopted.

THEORETICAL BASES

Reconstructed signal, $\hat{S}_r$, is:

$$\hat{S}_r = \frac{1}{r} \left\{ \sum_{j=1}^{a_j} E_{ri} + \sum_{j=1}^{n} a_{rij} S_j r_i - j \right\} \quad (1)$$

Signal-to-noise ratio, SNR, is:

$$SNR = 10 \log \left\{ \frac{E[E(S_r)^2]}{E[(S_r - \hat{S}_r)^2]} \right\} \quad (2)$$

where $r$ is the scaling factor for the $r$th section, $a_{rij}$ is the linear predication coefficient, $E_{ri}$ is the quantized prediction error, $S_r$ is the digitized speech signal, and $\hat{S}_r$ is the approximation of $S_r$.

EXPERIMENTAL RESULTS

The selective quantizer had three quantizer functions and each function was selected according to the standard deviation of prediction errors for each section. Fig. 1 shows SNR obtained by computer simulation.

APPLICATION AND CONCLUSIONS

This new coding method was applied to the speech reproduction system [1]. Consequentially, without any reduction in sound quality, memory capacity was reduced by 50%. Frequency band-width was 4.5kHz.

REFERENCE


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FIG. 1 SIGNAL-TO-NOISE RATIO
INTRODUCTION

For the information theoretical and experimental study of the transmission system especially with non-linear characteristics, an appropriate model of information source as input signal is necessary.

THEORY AND EXPERIMENTAL RESULT

It is natural to give a model signal $U(t)$ in a form of $U(t) = V(t)W(t)$, where $V(t)$ is the audible random signal with the pdf $f(v) = (1/\sqrt{2\pi})\exp(-v^2/2)$ of a constant rms, and $W(t)$ is the rms signal with the $M-$distribution, $h(w) = (2m^m/g(m))w^{m-1}\exp(-m/w^2)$, which is well known as an intensity distribution of fading (Fig.2).

The pdf of $U(t)$ is then given by

$$f(u) = \frac{2m^m}{\sqrt{2\pi}g(m)} \int_0^\infty \frac{2(m-1)w^{2(m-1)}\exp(-u^2/2w^2)}{w} dw$$ (1)

Since $f(u)$ fits well to the pdfs of program sounds (Fig.1), the $U$-process is an appropriate model for both level-controlled and uncontrolled program signals.

GENERATION OF THE MODEL SIGNAL

The $V(t)$ can be made of the sum of the narrow band Gaussian processes, non-synchronized sine waves, or the author's process,

$$V(t) = \sum_{i=1}^{M} c_i \cos(\omega_t + \Delta \theta_i(t) + \phi_i)$$ (2)

where $c_i$ and $\phi_i$ are constants, $\Delta \theta_i(t)$ is a uniformly distributed random variable, range of $\Delta \theta_i >> \pi$ for orchestra and $<< \pi$ for speech. A method to generate $W(t)$ is shown in Fig.3, and its parameter $m$ can be varied by $C$.

CONCLUSION

The $U$-process consisting of almost line spectral components is suitable for the measurements of realistic non-linear distortion or cross talk, or for the comparison of limiting amplifier and monitor meter with non-linear gain or response.

Fig.1 The pdfs of $f(u)$ and program sounds.

Fig.2 The pdf of the $M$-distribution.
A MONOSYLLABIC VOICE TYPEWRITER

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INTRODUCTION

Every Japanese spoken word can be pronounced as a series of open monosyllables, each of which corresponds to a phonetic sign. This enabled us the design of a voice typewriter quite easy. For the purpose of a blind aid, we designed a monosyllabic voice typewriter with a relatively simplified signal processing system.

TECHNIQUE

To minimize the cognition method of an open monosyllable, we identified at first its vowel component, then consonant component was isolated and identified by a simple pattern matching method. For example, /ka/ might be recognized as follows: (1) /-a/ was identified by a pitch synchronized pattern matching, (2) all vowel pitches of /-a/ were ignored to isolate /k-/, (3) then the consonant /k-/ was recognized by the pattern matching technique, (4) 'ka' was reconstructed by activating typewriter keys of 'k' and 'a' or a single key corresponded to the Japanese phonetic sign of 'ka'.

The cognition of vowel component was realized by a spectrum analysis using our 16 channel critical band filters simulating the frequency characteristics of hair cells and basilar membrane of an ear. Its frequency range was 200-4,400 Hz. The sampling of vowel was began retrogradely after 150 msec from the onset of voice with the pitch synchronized technique. Our pitch detector was the combination of low-pass filter and C class amplifier, so that the fundamental wave was enhanced for detection. The isolation of consonant component was carried out simply erasing all pitches of vowel retrogradely from 150 msec. The remainder might include the consonant and the transient. Whole remainder, however, was identified by the same method as for the vowel component. Original patterns for matching were stored in a memory of a small computer by arithmetic averaging of the newest personal voice data. In each case of correct printing of typewriter, the data were refreshed by adding the newest one, and averaging a certain number of recent data. The learning procedure was effective to avoid a technique to overcome a person to person difference of pronunciation. For the category determination of pattern matching, a square distance method was applied. Simplified block diagram of our technique is shown in Fig. (1).

RESULTS AND CONCLUSIONS

The score of correct cognition of vowels was more than 99%; of 6 plosive consonants was about 94%. The correction of error was established after 2-3 repetitions of averaging. Our present model works almost sufficiently. However, a feedback method of printed character to assure its correctness is needed for practical use.

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Fig.1 Block diagram of voice typewriter.
AN LSI IMPLEMENTATION OF DISTRIBUTED ARITHMETIC

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INTRODUCTION
The implementation of digital filters can be conducted by computing a number \( y_2 \) by:

\[
y_2 = \sum_{i=0}^{n} a_i x_{n-i}
\]

where \( x_{n-1}, a_i \) are discrete time variant numbers. When faced with a physical implementation of a signal processor computing (1), one needs to structure the arithmetic operation in a way which minimizes the number of multipliers and adders.

THEORETICAL BASIS
If the components of \( x \) are coded in 2's complement code such as:

\[
\begin{align*}
(x_{n-1})_w & = \sum_{i=0}^{w-1} a_i 2^i \\
\end{align*}
\]

where \( a_i \) are equal to 0 or 1, and \( w \) the number of bits defining the number.

Equation (1) can be computed by:

\[
y_2 = \frac{1}{2} \left[ A(x_0 \ldots x_{n-1}) \cdot A(x_0 \ldots x_{n-1}) \cdot 2^{n-1} + \ldots + \ldots + \ldots \right] + \sum_{i=0}^{n} a_i
\]

where \( A(x_0 \ldots x_{n-1}) \) are partial sums of \( a_i \) given by the examination of some weight bits of \( x_0 \ldots x_{n-1} \).

In the case of \( i=2 \), table 1 gives the value of partial sums to be used:

<table>
<thead>
<tr>
<th>( x_0 )</th>
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<th>( A )</th>
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<td>1</td>
<td>4</td>
</tr>
</tbody>
</table>

IMPLEMENTATION
An LSI circuit using a MOS process and two phase dynamic circuits has been implemented. It handles 16 bits words for \( x_2 \). The number of words is limited to 3. The value of each of them is restricted to the range \(-1\). The \( a_i \) are also limited to 16 bits but their values are in the range of \(-3\).

The LSI circuit performs also some functions needed by the digital filtering as: scaling, overflow detection and correction, round off of the result at 16 bits words. All these functions are implemented in a 214x214 mm chip, with a clock rate of 2.5 MHz. The chip performs 1.60,000 distributed arithmetic operations equivalent to 480,000 multiplications and 320,000 additions per second.

CONCLUSION
The use of this circuit allows to implement in an economical way digital filters. For instance, with a sampling rate of 16 KHz, a bank of 48 second order digital filters can be built.

REFERENCE
A. Croisier, D.J. Esteban, M.E. Levillon and V. Riso,
In this paper it is described a transversal (feedforward) predictive coding scheme in which the residual signal is coded by means of a sub-bands type coding technique.

INTRODUCTION

It has been recently shown [1] that, although transversal predictive coding technique does not carry out any relative signal to noise improvement with comparison to a feedback predictive coding scheme, it allows to improve the perceptual pleasantness of the coded speech.

On the other hand, sub-bands coding technique with Quadrature Mirror Filters [2] has been shown to be attractive in the sense that it improves both the resulting signal to noise performance and the perceptual quality of the coded speech.

SYSTEM DESCRIPTION

The proposed system is based on association of short-term prediction [1], whose coefficients are determined by means of the PARCOR method [3], and Splitband Voice Coding Scheme (SVCS) [2] which is used to encode the residual signal resulting from inverse filtering of the original signal.

RESULTS AND CONCLUSION

Different simulations have been performed assuming 8 K samples per second and compression factors range of 5 to 8 (16 Kbps to 12 Kbps resulting information rate), it has not been observed any signal to noise performance improvement with respect to the figure provided by the SVCS but a real increase of the coded speech pleasantness.

Although attractive, it has nevertheless to be pointed out that the additional cost resulting from the hardware complexity of the short-term prediction is not justified when applied to voice telephony.

Different comparative taped results will be played at the conference.

REFERENCES


INTRODUCTION

Speech decomposition in sub-bands has been shown to be a really promising approach for speech coding /1/. A quasi perfect sub-band decomposition based on a tree arrangement of Quadrature Mirror Filters (QMF) has been applied to speech coding at low bit rate /2/. This paper deals with the equivalent parallel arrangement of the sub-band splitting.

FORMULATION

With reference to the QMF principle, it can be demonstrated that the tree arrangement has an equivalent parallel arrangement which is shown in the figure in case of four sub-bands splitting. The sub-band signals are first decimated by a factor 4. Then, the sampling rate is increased to the initial sampling rate by inserting 3 zero valued samples between each sample. The z transform $S(z)$ of the reconstructed signal $s(n)$ is given by:

$$S(z) = \frac{1}{4} \sum_{i=1}^{4} H_i(z)[H_i(z),X(z)+H_i(-z),X(z)+H_i(-z),X(-z)+H_i(-z),X(-z)]$$

If the filters are chosen so as to verify the following phase relations:

$$\phi_3(w) = \phi_4(w) = \phi_2(w) + \frac{\pi}{2} = \phi_1(w) + \frac{\pi}{2}$$

it can be checked that the aliasing effects corresponding to $X(jz)$, $X(-jz)$ and $X(z)$ are cancelled, resulting in a quasi perfect reconstruction of the signal.

INTEREST AND CONCLUSION

This type of decomposition can be used in the area of speech coding /1/, /2/. The contiguous filters can be designed with non equal bandwidths, so as to take in account the articulatory index /1/. In addition, the extension to more than four sub-bands can be made straightforwardly. Although being equivalent to the tree approach, this approach can be of interest to simplify the filters implementation.

REFERENCES


Die Arbeiten /1/,/2/ stellten Grundlagen für die technische Realisierung der Automaten zur Erkennung der polnischen Vokale und Frikative vor.


LITERATUR

/1/ Jassem W., Szybiata D., Dyczkowski A.: Rozpoznawanie spółgłosek polskich w typowych zdaniach. Prace IPPT 43/1975

SPEECH RECOGNITION OF FORTRAN PROGRAMS

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INTRODUCTION

We are trying to recognize connected speech concerned with Fortran programs and Japanese sentences (1). In the system for Fortran programs we can make the most of syntactic informations but existence of English letters makes the performance difficult.

RECOGNITION SYSTEM

In the acoustic processor we extract several characteristic parameters such as Parcor coefficients, formant frequencies, the number of zero crossings, and select a candidate phoneme every 10 msec, in which silent and buzz parts are dealt with as a kind of phoneme. Taking the stationality of speech wave into account, the candidate phoneme is adopted or excluded, and a phoneme string is formed. From this phoneme string we extract characteristic phonemes such as vowels and /s/ continuing over 50 msec and silence preceding voiceless stops. On the other hand, every same phoneme appearing successively in the original phoneme string is condensed into a single phoneme. This condensed phoneme string is used as the input for the following procedure.

Candidate words are, at first, selected referring to syntactic informations expressed by state transition networks (2,3), and restricted preliminarily by the first two phonemes in each item of the word lexicon and moreover by the characteristic phonemes. The final candidate words, which are restricted finally by pragmatic informations such as variable names, are compared with a part of the input phoneme string, whose range is decided referring to the number of phonemes in the candidate word. Eventually some word strings are obtained for the whole input phoneme string, and the most reliable word string is adopted.

A syntactic error correcting routine is set to work whenever recognition procedure does not satisfactorily come to the end of the statement. In this routine the key words (e.g. "READ(/,digit,)" for READ statement) and some supplemental characteristic words (e.g."+,-,*," for ARITHMETIC ASSIGNMENT statement) of each statement type (e.g. READ statement) are searched in the whole input phoneme string so as to find the correct statement type. Knowing the statement type, recognition procedure should be begun again from the beginning of the statement.

PERFORMANCE AND CONCLUSION

The vocabulary is 140 words, including 26 English letters. Though we cannot yet distinguish enough among some English letters and so some confusions among them may happen and some requirements such as utterance of /kaigio/(means "carriage return") at the end of each statement and exclusion of comments are imposed, the system can recognize simple Fortran programs (about level IV) spoken by a single speaker.

REFERENCES

TENTATIVE DE POINT ET PROSPECTIVE EN MATIÈRE DE TRAITEMENT NUMÉRIQUE DES GRANDEURS ACOUSTIQUES

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L'auteur passe en revue les différentes méthodes de traitement numérique des grandeurs acoustiques ; pression acoustique $P(t)$, niveau de bruit $L(t)$, l'analyse d'octave ou tiers d'octave, etc...

Il décrit ensuite plus en détail les méthodes et les systèmes mis en œuvre au L.A.M.I. et qui portent surtout sur l'acquisition, le stockage, la transmission et le traitement numérique du niveau de bruit $L(t)$.

Il expose quels résultats pratiques ont été obtenus, quelles possibilités ils offrent et surtout, quels enseignements ont pu et pourront être tirés des grandeurs intégrales ou statistiques diverses que ces traitements fournissent.

Il examine ensuite quelles sont les perspectives futures de tels systèmes tant en matière d'application à la surveillance de l'environnement qu'à la recherche fondamentale.

Il envisage les possibilités de normalisation de tels systèmes et les problèmes qu'elle soulève.

Enfin, pour conclure, il tente de recenser ce que les toutes récentes méthodes et techniques numériques peuvent encore apporter au traitement du signal ou de l'information acoustique.

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INTRODUCTION

This paper deals with the application of acoustic pattern recognition to the quality control of high-rate industrial production processes. Recognition techniques are used to identify faulty components, defects in operating assemblies, and excessive sound levels.

In most cases where acoustic signals are now used in quality control, workers attempt to identify production faults by listening. Quality control may be accomplished more uniformly, more efficiently, and more humanely by applying pattern recognition principles to do these tasks automatically or semi-automatically.

METHODS

We found that the costs for the implementation of automatic control systems—which are generally in competition with auditory testing—are given above all by the effort for the development and not for the hardware realization of the system. For complex machines, the signal analyses take the greatest effort. The extraction of features of the air sound and vibration signals is done by "adapted signal processing", which allows relatively fast analyses. The resulting test algorithm are realizable inexpensively in semiconductor devices.

RESULTS

The automatic acoustic quality control of several products was investigated (e.g. electromotors, gears, geartrains, bearings, tiles). Some systems are already used in industrial production. For very complex products (e.g. combustion engines) interactive test methods are in development.
INTRODUCTION

The fundamental principles of our impulse measuring method were published in (1), (2). Since this time the experimental realisation of this method was rather simplified, and some new theoretical results were derived, too.

THEORETICAL BASES

The theoretical base of our method is an approximative treatment of the Huygens Rayleigh integral published in (3). From eq. (30) of paper (3) can be expressed the reflection coefficient by means of the complex spectrum of both, primary and reflected impulses. If, however, the whole reflected impulse has to be recorded the measured sample must be large at least as the ellipse given by eq. (21) of the paper (3). If the sample is not large enough the reflected impulse is deformed by the diffraction. From eq. (28) can be derived the formulae by means of which it is possible to eliminate an influence of the diffraction for the elliptic sample the edge of which is given by eq. (21), but, $a_0$ must be substituted by $a_0 < a_0$. For both, the unbaffled and baffled circular samples it is possible to eliminate a diffraction effect by means of eq. (11), (17) of the paper (4).

EXPERIMENTAL TECHNIQUES

As the source of impulses there an electric discharge was used. This type of source is advantageous because it gives nearly spherical wave which has very good reproducibility. At first an oscillographic record of impulses was used (2). The oscilograms, however, enabled record of very short time intervals of impulse only, and the digital evaluation of oscilograms was very laborious and inaccurate one.

Nowadays, the Digital Event Recorder B&K type 7502 is used. Reflection coefficient, and other parameters of absorbers are computed from digital record of impulses by means of the computer TESLA 200.

RESULTS AND CONCLUSIONS

Very good agreement between results of our impulse measurements, and classical stationary ones are achieved when homogeneous isotropic absorbers are measured. Results for nonhomogeneous anisotropic absorbers are dependent on the position of a measuring point. There is no comparison for this type of materials because for them the statistical absorption coefficient can be defined only, but, impulse measurements yield an absorption coefficient for discrete angles of incidence.

REFERENCES

(1) Z. Kyncl, 7th ICA (1971), 333
(2) Z. Kyncl, Accoust. Journal USSR (1975), 1, 49
(3) Z. Kyncl, 6th Conf. on Acoust., Budapest (1976), 4.10/1
(4) Z. Kyncl, ACOUSTICS'75, Varna (1975), 265
INTRODUCTION

Our research is to obtain descriptors for the automatic recognition of vehicles from their acoustic emission. The purpose of this paper is to show that pulse position modulation of the gas pulses causes the noise from a vehicle's exhaust system to possess characteristics that depend on the design of the exhaust manifold.

THEORETICAL BASES

The assumption is made that the noise from an exhaust pipe is the convolution of the noise output, \( n(t) \), from the engine with the impulse response, \( x(t) \), of the exhaust system.

\[
p(t) = n(t) * x(t)
\]

A number of different techniques for computer analysis of this noise are compared including the optimum comb.

\[
oc = 1/ \sum_{i=0}^{k-1} (P_{n+i} - P_{n+i-m})^2
\]

This was applied in (1) to the pitch period analysis of speech.

EXPERIMENTAL TECHNIQUES

Recordings are taken from moving vehicles under normal operating conditions and therefore contain not only exhaust noise, but also directly and indirectly radiated engine noise together with sounds radiated from the many other noise sources. The optimum comb, illustrated in Fig. 1, can be used to distinguish some of the engine characteristics in the presence of these other noise sources.

RESULTS AND CONCLUSIONS

As reported in (2), a short-term power spectrum of a vehicle's acoustic emission may contain harmonic series from both the interval between successive cylinder firings and also from the interval between successive firings of the same cylinder. The relative magnitude of the harmonic series depends on the regularity of the interval between the gas pulses reaching the inlet to the exhaust system. These intervals depend on the design of the exhaust manifold and the effect gives rise to the different amplitude peaks in the optimum comb.

REFERENCES

INTRODUCTION
In the speech processing research, quasiperiodic signals are found. In this paper will be developed some functions to analyze this type of signals without any reference to its origin. The properties of the normed linear spaces are systematically used. In an accompanying paper (3), the use of these functions to the pitch extraction and voicing problems will be shown.

THEORY
1. Spaces. Let us consider the linear space $E_p$ of real functions $f$ of a real variable, which admit the Lebesgue integral of its power to a real exponent $p > 1$, over any finite interval. That coincides with the Riemann integral if it exists and became a sum if $f$ is a succession (3).

2. Windows. Now we consider the set of positive, bounded supported functions $w$ of unit area and the equivalence relationship $w_0 \equiv w_1$ if $w_0(t) = (1/g) w_1(t/\alpha)$, being $\alpha$ a positive real. Each class of equivalent functions will be called a window (see fig. 1).

3. Norms and distances. We define a family of norms in $E_p$ as:

$$\|f\|_{\alpha,p} = \left( \int_{\alpha} w_\alpha |f|^p \, dm \right)^{1/p}$$

Therefore $\|f - g\|$ is a distance (1, 2) between the functions $f$ and $g$.

4. Autodissimilarity. This function is the normalized distance between $f_1 = f(z + t + \tau/2)$ and $f_2 = f(z + t + \tau/2)$; $z$ is a dummy variable for integration. We define the autodissimilarity as:

$$\text{ads}_{f_1, f_2}(\tau) = \frac{\|f_1 - f_2\| - \beta \|f_1 - f_2\|}{\|f_1\| + \|f_2\|}$$

Where $0 < \beta < 1$, and $\alpha$, the support length of $w$, is made proportional to the absolute value of the shift $\tau$, the real variable. It is verified, for any $p$, $v$, $\tau$, $\beta$, and any real number $\varepsilon$:

$$0 \leq \text{ads}(\tau) \leq 1 \quad \text{ads}_{f_1, f_2}(\tau) = \text{ads}_{f_2, f_1}(\tau)$$

By varying $\tau$, we define a new function of $\tau$, the aperiodicity $A(\tau)$. In the intervals of $\tau$ where $A(\tau)$ is smaller than an arbitrary threshold $a$, $f(\tau)$ will be considered quasiperiodic. Its quasiperiod $T$, also function of $\tau$, will be defined as the shift $\tau$ where $\text{ads}(\tau)$ is a minimum. The intensity $I(\tau)$ will be $I(\tau) = \|f(z + t)\|_{\alpha \tau}$, that is, the norm of $f$ in $\tau$, being $\alpha$ an integral multiple of $T$.

In the intervals where $A(\tau)$ is greater than $a$, $f$ is called aperiodic, there is not quasiperiod and the intensity is calculated with $\alpha$ being a constant. Some other continuity requirements on $I(\tau)$ and $A(\tau)$ are used in the quasiperiodic-aperiodic decision.

REFERENCES
INTRODUCTION

Many digital speech transmission systems, e.g. ADPCM and ADM, cannot be adequately evaluated by the use of sinusoids or bandlimited white noise.

In this paper we propose a generative model of an artificial test signal with the average temporal and spectral characteristics of speech, which is more suitable for the performance evaluation of the above mentioned systems.

DESCRIPTION OF THE MODEL

The proposed model is represented by an excitation function (sound source) and a spectral shaping function (vocal tract) with the structure of fig. 1:

![Block diagram of the artificial signal generation model](image)

where:

- G indicates a generator of periodic waveforms allowing also some kind of pseudorandom modulation of pitch and amplitude,
- \( u_n \) is the mathematical excitation signal simulating a typical glottal waveform (1),
- \( F_1 \) is a digital linear phase filter performing spectral flattening of \( u_n \),
- \( W_n \) is a white spectrum signal,
- \( F_2 \) is a digital all-pole filter carrying out the predictive vocal tract model (2),
- \( s_n \) is the generated artificial signal.

It has been inspected that the temporal and spectral characteristics of the artificial signal agree with the average speech characteristics.

COMPUTER SIMULATION RESULTS AND CONCLUSION

Computer simulation tests of digital speech transmission systems, a 32 kbit/s adaptive differential PCM (ADPCM) (3) and a 32 kbit/s adaptive delta modulation (ADM) (4), have been carried out with test signals constituted by ten telephonic voices, the artificial signal, a 800 Hz sinusoid and a spectrally shaped Gaussian noise: the performance evaluation in terms of overall and granularity SNRs and of percent probability of slope overload with the signal \( s_n \) is within the range corresponding to telephonic speech and shows on the whole an agreement to the average speech performance better than with the sinusoid or the Gaussian noise.

REFERENCES

(4) P. Castellino, CSELT Internal Report No. 76.02.021.
The exact mathematical model (1) of the electromechanical system enables to analyse the most important system features influence on the nonlinear distortions of the input signal. For stationary gaussian input model, the analysis of its power spectrum is sufficient. The power spectra $W_x(f)$ of the set of input models were formed with wide, narrow and narrow-portioned band, all shifted along $f$-axis within the acoustic frequency range to reach the approximation of the real acoustic signals features (2-3). The new method, based on the multidimensional transfer function (mtf) of the derivation of output power spectrum was worked out. The method enables to analysis the each order mtf influence on output power spectrum, and consequently, to analyze the nonlinear distortion of each order separately (2).

As an example, the features of the 2-nd order system transfer function influence on output power spectrum is described here. The formula is the starting point (2): \[ W_y(f) = \int \left[ \int \left( \frac{df}{dt} \right)^2 W_x(f) \right] dt \] where $W_x(f)$ - the 2-nd order distortion power spectrum effected by the system with the 2-nd order mtf $H_2(t_1, t_2)$, $f_1$ - lower limit frequency of $x(t)$ with $W_x(f)$, $f_2$ - upper limit frequency of the system. The value of $W_y(f)$ at the fixed frequency $f$ depends on the subset of $H_2(t_1, t_2)$ values derived from $H_2(t_1, t_2)$ by taking $f_2=f-f_1$. When $W_x(f)$ is fixed, from (1) follows that the given input will be distorted in different manners by various systems, while various inputs may be distorted in completely different ways by the given system. To make it clearer, the model system /Fig.1/ was worked out (2). The model reflects schematically the type of the nonlinearity with memory of the 2-nd order real acoustical systems transfer function.

Fig.1. Modulus of the $H_2(t_1, t_2)$ model.  Fig.2. The 2-nd order system output power function.  

REFERENCES

(1) A. Gabor, J. Zarzycki, Proc. of 9-th ICA, Madrid, 1977
(2) J. Zarzycki, Ph.D. Thesis, ITA Technical University of Wroclaw, 1975
ELECTROACOUSTIC SYSTEM NONLINEARITY AND MEMORY ANALYSIS USING VOLterra SERIES

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The traditionally used methods of the description and measuring of systems nonlinear distortions yield to unprecise and often false results. In order to find the reason of this, the system mathematical model, fully describing its nonlinearity as well as memory, should be found. The model was worked out using the Volterra series to describe the system performance:

\[ y(t) = \sum_{n=0}^{\infty} h_n(t_1, t_2, \ldots, t_n) x(t_1) x(t_2) \cdots x(t_n) \]

where \( x(t) \) - finite power input, \( y(t) \) - output, \( h_n(t_1, t_2, \ldots, t_n) \) - \( n \)-th order Volterra kernel / \( n \)-dimensional system impulse response / \( h_n(t_1, t_2, \ldots, t_n) \) describes only linear part, while \( h_n(t_1, t_2, \ldots, t_n) \) completely describes the \( n \)-th order nonlinearity with memory. In frequency domain, the output is completely characterized by the set of multidimensional transfer functions \( H_n(f_1, f_2, \ldots, f_n) \) is \( n \)-dimensional Fourier transform of \( h_n(t_1, t_2, \ldots, t_n) \), as well as by the set \( H_n(t_1, t_2, \ldots, t_n) \) in time domain. The sets \( \{ h_n(t_1, t_2, \ldots, t_n) \} \) and \( \{ H_n(f_1, f_2, \ldots, f_n) \} \) in certain situations set \( \{ H_n(t_1, t_2, \ldots, t_n) \} \) completely and unmistakably describes the system nonlinear distortions, distinctly from traditionally used quantities. The nonlinear distortions of any system input can be derived when the mentioned sets are known (4).

The methods of measuring \( h_n(t_1, t_2, \ldots, t_n) \) and \( H_n(f_1, f_2, \ldots, f_n) \) measuring are known. Starting from the Schetzen's measuring method of \( h_n(t_1, t_2, \ldots, t_n) \) (3), the sinusoidal method of \( H_n(f_1, f_2, \ldots, f_n) \) measuring was worked out and then realized (1). The Fig. 1 shows the exemplary measuring result.

Now, when the informations brought by the traditionally used methods of the system nonlinear distortions analysis can be evaluated when the exact mathematical system model /1/ and the measuring results /Fig. 1/ have been obtained. The 2-nd order nonlinear distortions were characterized by the 2-nd harmonic or intermodulation and difference frequency distortions when using tone /or tones/ as inputs. But these characteristic are only fragments of \( H_2(f_1, f_2) \) /the 2-nd harmonic is its diagonal line /\( f_1 = f_2 \)/ and the remaining characteristics are its different sections. However, the nonlinear distortion of the 2-nd order are fully characterized only by the whole \( H_2(f_1, f_2) \) function and for its measuring there are needed and are sufficient two input tones with frequencies shifted along \( f_1 \) and \( f_2 \) axis. The explanation of the higher order nonlinear distortions is similar (1).

The considerations show that the crosscorrelation function between the input /ergodic, gaussian noise/ and system output, as well as its response for input being Dirac impulse bring full informations about system linear distortions and mixed up together partial informations about its nonlinearities. However, the correlation method as well as impulse method may bring all informations about nonlinear distortions when the set \( \{ H_n(t_1, t_2, \ldots, t_n) \} \) will be measured (1-2).

It is seen that the exact system model, based on Volterra series, has cleared the reasons of failures of traditional attempts of nonlinear distortions description. The model has also enabled to propose the exact method of its description and measure. Every simplifications may be done only on the basis of the sets \( \{ h_n(t_1, t_2, \ldots, t_n) \} \) and \( \{ H_n(t_1, t_2, \ldots, t_n) \} \) analysis.

REFERENCES
1) A. Gabor, Ph. D. Thesis, ITA Technical University of Wroclaw, 1975
DIE BERECHNUNG DER FREQUENZKENNLINLIE£EN BEI DER SIGNALAUFSPEICHERUNG UND WIEDERGABE MIT HILFE DER FOURIER-TRANSFORMATION

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Bei der Signalspeicherung, wo der Aufzeichnungsprozeß durch Lichtstrahlen, Elektronenstrahlen, Holographie und auf ähnliche Weise realisiert wird, ist die Intensität in so genanntem Spalt konzentriert. Ebenso wird bei der Wiedergabe durch eine optische oder elektronische Methode ein Spalt verwendet um eine elementare Aufnahmefläche zur Wiedergabe des Signals zu erreichen. Im idealen Fall soll in der ganzen Fläche des Spaltes die Intensität gleichmäßig verteilt sein. In den praktischen Fällen ist jedoch die Intensität ungleichmäßig verteilt und in der Richtung gegen die Ränder des Spaltes nimmt ihr Wert gleichmäßig ab. Diese Verteilung der Intensität im Spalt bestimmt die Frequenzkennlinie des Aufzeichnungsprozesses und bei der Wiedergabe die des wiedergegebenen Signals. Den Verlauf der Frequenzkennlinie kann man durch folgendes Verfahren bestimmen.

Zuerst muß man den Verlauf der Verteilung der Intensität im Spalt in der Richtung der Speicherbewegung bestimmen. Im Bild la ist diese Funktion der Intensität I = f / f / gegeben, welche man in einen Zeitimpuls verwandeln muß, also I = f / f / nach Bild lb.

\[ S = \frac{b}{v} \]

\[ v \] - Geschwindigkeit der Speicherbewegung

\[ f = 2\pi \omega \]

Bild la, b Die Intensitätsverteilung im Spalt


\[ F / \omega = \int_{-\infty}^{+\infty} \frac{f}{t} e^{-j2\pi ft} dt \]

INTRODUCTION

Here presented is an efficient approximation method to obtain the logarithmic-frequency-scale representation of discrete frequency spectrum of a band-pass signal using the double sampling technique (1) and an approximate interpolation formula with the sampling function.

DISCRETE FOURIER TRANSFORM OF A BAND-PASS SIGNAL

Let \( g(t) \) be band-limited in the frequency region \( \mathcal{B} = \{ -f_L, f_L \} \cup \{ f_1, f_2 \} \) and we want to obtain the logarithmic-frequency-scale representation of the discrete frequency spectrum of \( g(t) \) in the frequency region \( \mathcal{D} \). Set \( f_1 \) and \( f_2 \) so that they satisfy the following equations:

\[
\xi = \frac{f_1 + f_2}{f_1 - f_0} \quad \text{integer}, \quad 0 \leq \xi, \xi' \leq f_1
\]

and the sampling interval \( T \) and the shifting factor \( \phi \) are determined as

\[
T = \frac{1}{f_1 - f_0}, \quad \phi = \frac{2m-1}{2E}, \quad m \in \{1, 2, \ldots, E\}
\]

Then, the signal \( g(t) \) is double-sampled at the discrete points \( t = nT \) and \( (n+\phi)T \) \( \{ m \in \mathbb{N} \} \). Let \( G_0(k\Omega) \) and \( G_2(k\Omega) \) denote the DFT of the sampled sequences \( \{ g_0 \} = \{ g(nT) \} \) and \( \{ g_2 \} = \{ g((n+\phi)T) \} \) respectively, where \( \Omega = 2\pi/NT \) and \( k \) is an integer. The discrete frequency spectrum of \( g(t) \) can be calculated as follows:

(i) for odd \( \xi \)

\[
G(\frac{\xi-1}{2T} + k\Omega) = \frac{e^{j\phi\xi}}{2T \sin \frac{\xi \phi}{2}} \left\{ -e^{-j(\xi+1)\phi \pi} G_0(k\Omega) + e^{-j\xi\phi \pi} G_2(k\Omega) \right\}
\]

(ii) for even \( \xi \)

\[
G(\frac{\xi}{2T} + k\Omega) = \frac{1}{2T \sin \frac{\xi \phi}{2}} \left\{ -e^{-j\xi\phi \pi} G_0(k\Omega) + e^{-j\xi\phi \pi} G_2(k\Omega) \right\}
\]

DISCRETE SPECTRUM ON A LOGARITHMIC FREQUENCY AXIS

Let \( \omega_{m} \{ m \in \mathbb{N} = \{-N/2, 0, \ldots, N/2\} \} \) denote the equi-interval sampling points between \( f_L \) and \( f_H \) Hz on the logarithmically scaled frequency axis, and \( \omega_m \) can be expressed as

\[
\omega_m = e^{\frac{j\pi}{N}} \omega_{m'} \quad \text{where} \quad \omega_{m'} = 2\pi \sqrt{\frac{m}{N}}, \quad \text{and} \quad m' = \frac{f_L}{f_H} m
\]

Then, the approximate value \( G_1(\omega_m) \) for discrete spectrum \( G(\omega_m) \) at \( \omega = \omega_m \) is given by the following approximate interpolation formula:

\[
G_1(\omega_m) = \frac{1}{2} \left( G(\omega_m) + G(\omega_m') \right)
\]

where

\[
G(\omega) = \frac{1}{2} \sum_{k=-\infty}^{\infty} G(k\Omega) \sin \frac{N\pi}{2} (\omega - k\Omega)
\]

REFERENCES

(1) M. Yanagida et al., IEEE Int. Conf. on ASSP (1976), 141
MEASUREMENTS BY RETROEXCITED SIGNALS

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INTRODUCTION

The measurements by retroexcited signals have several advantages in comparison with those from direct signals, as it is possible to obtain additional information when measuring transmitting media constants. So, re-cycling signals through them and then processing the resultant retroexcited and recorded signals, has been a convenient method of extending the general measuring capability.

THEORETICAL BASES

The work previously reported (1) and the generalized theory (Novoa, to be published), including: the retroreverberation gain mathematical model, the gain relations, the retroexcited transfer function, the retroreverberated transfer function, the transmitting medium transfer function isolation, the resolving power of the method, the growing peaks phenomena and the accumulation of noise in the retroexciting process, form the theoretical bases of this work.

EXPERIMENTAL TECHNICS

In order to facilitate the obtaining of the retroexcited signals, a device has been constructed, several suitable signals have been prepared for the special measurements and specialized technology has been developed. The desired measurement data, being obtainable by processing the steady state and/or the transients of the retroexcited signals.

RESULTS AND CONCLUSIONS

Special measurements have been possible by the practice of retroexciting signals, like damped natural frequencies discrimination and the recognition of some network structures by interpretation of the retroreverberated spectral sequences. Also some inexpensive more traditional measurements have been obtained, such as: pole diagrams, signal averages and transmitting medium gains.

The method has proved practical when applied to acoustics, vibrations and electrical circuits. Further experimental work is being planned expecting to improve the measuring accuracy.

REFERENCES

EFFECTS OF ACOUSTIC MEDIUM AND SYSTEM PERTURBATIONS ON THE PERFORMANCE OF ADAPTIVE ARRAYS

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INTRODUCTION

The derivation of the optimal characteristics of adaptive arrays require certain basic assumptions on the acoustic field to be processed (1). First generation adaptive processors are designed under the assumptions of plane-wave signals, uncorrelated signal and noise fields, ideal propagation medium and distortionless receivers. This paper investigates the effects of perturbations to these assumptions on the performance of adaptive array processors.

THEORETICAL BASES

In actual operating conditions, the ideal assumptions do not hold and it becomes extremely important to determine the sensitivity of the adaptive processors to the medium and system perturbations. When system requirements permit, the use of processors matched to a given perturbation may be desirable. In other cases, it becomes necessary to devise additional constraints to minimize the sensitivity.

EVALUATION TECHNIQUE

In the major part of the performance analysis a ten element equally spaced line array associated with element space frequency domain adaptive weighting is considered. The weight vector, derived from the maximum likelihood criterion, is given by

$$z = \frac{\mathbf{R}^{-1} \mathbf{E}_0}{\mathbf{E}_0^\top \mathbf{R}^{-1} \mathbf{E}_0}$$

where the covariance matrix, $\mathbf{R}$, and the steering vector, $\mathbf{E}_0$, are subjected to perturbations.

Figure 1 shows the performance results for curved wavefront signal as an example. The processor output signal-to-background ratio, $\text{SBR}_\text{q}$, is plotted versus the amount of curvature in wavelength, $\lambda/\lambda$, for various input signal-to-noise ratios, $\text{SNR}_\text{i}$, for the adaptive, ABF, and conventional, CBF, processors. The results for adaptive processors designed in the time domain and/or beam space are also obtained for comparison purposes. The studies were extended to the design of processors that are matched to a given perturbation.

RESULTS AND CONCLUSIONS

The steady-state analytical results obtained in this study indicate that multipath and wavefront curvature effect the performance most. The amount of degradation is, however, strongly dependent on the signal-to-noise ratio and it is tolerable for small signals of interest. For large signals, the degradation can be minimized by imposing additional constraints or different types of spatial filtering.

REFERENCES 1. A.M. Vural, IEEE EASCON'75 Record, pp. 34A-M.
BARCELONA SYMPOSIUM

Sound Recording and Reproduction
THE OBJECTIVE MEASUREMENT OF ACCURACY IN SOUND REPRODUCTION

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INTRODUCTION

Objective tests of the quality of reproducing equipment have been criticized from two points of view. One is that objective measurements cannot provide universally valid results because of hearing differences among listeners, differences in room environment, and differences in taste. The second is that objective measurements sometimes fail to predict subjective evaluations on which experts agree.

THE FUNCTION OF A SOUND-REPRODUCING SYSTEM

If a "high-fidelity" system participates in creating musical sound, it would be as useless to evaluate the system through objective measurements as to judge a Stradivarius violin by acoustic measurements. But if the purpose of a reproducing system is to recreate with maximum accuracy sound whose character has already been determined, the objections listed above do not hold up.

Differences in individual hearing do not affect the comparison of a copy to its original, because the same hearing aberrations come into play for both the live and the reproduced sound. Room environment profoundly affects the final acoustic output of a reproducing system, which simply means that the room must be taken into account as one link in the chain of reproduction. Taste may determine whether a listener prefers a Stradivarius to a Guarnerius, but it does not affect the objective determination of the accuracy of reproduction of either. Taste can only be involved where a choice must be made between different kinds of inaccuracy.

VALIDATION OF OBJECTIVE TESTS BY SUBJECTIVE "LIVE VS RECORDED" COMPARISONS

When a reproducing device is reported as measuring well but sounding bad (or vice versa), someone has measured the wrong things, however accurately. If measurements are to be valid indices of reproducing accuracy, they must first be validated by subjective evaluations in which the listeners compare the reproduced sound to the original sound.

Examples of unvalidated (or, as in Fig. 1, demonstrably invalid) tests of reproducing equipment are given. "Live vs recorded" test-validation techniques are described, using as the reference source either live musicians, or an electronic stand-in for live musicians, or random noise.

Fig. 1 The smooth on-axis pressure response of this high-frequency speaker fails to predict a nasal quality produced by the peaked power response. More complete data do predict it.
Holophony is the acoustic counterpart of Holography. Its theory can be developed using a new analysis of Huygens' Principle applied to sound waves (analysis based on the "cutting-off" functions theory) (1). Its aim is the perfect reconstruction of an acoustic event in the 4D-time space. The time-reconstitution has been resolved by Edison and his successors, so it is not a problem now. But the space-reconstruction question is still open, and we can say that for the present Holophony is the only scientific response to it.

Beforehand, there was a preliminary question to resolve: is human hearing tridimensional? Literature gave us several elements (especially about the horizontal and median planes) and we made several tests which are reported (2) which prove that the response is positive. We tried then to apply Holophony theory. This theory implies a continuous density of acoustic sources covering all the walls of the auditorium (i.e. of the home-listening room); these sources should be at the same place occupied by the microphones during the recording event; and each source must have a cardioidal or conchoidal radiation pattern. But, practically, we do not have to respect precisely all these conditions. Obviously, we must work with a finite number of sources i.e. with a discrete density of sources. Let us say here that we can also consider each point of the membrane of a loud-speaker as an elementary source so that a loud-speaker surface may be seen as a continuous density of sources. We might also think of a large membrane (covering all the walls of a room) moved by many motors regularly disposed whose movements could be the solution of the secondary sources equation. Whatevsoever, the disposition of microphones during the recording event depends of the number of sources, due to psychoacoustic factors which will be discussed (3).

Finally, if the room is not too much reverberative, the shape of the radiation pattern of the sources does not seem an essential factor as it is when we apply Holophony to Active Acoustic Barriers.

We can reproduce an acoustic event with one single source: but monophony cannot give any spatial knowledge of the musical event (except perhaps for a solo played at the exact place where the monophonic loud-speaker is located). Using two reproduction sources, we get what is usually called stereophony. This means gives information in the horizontal plane in front of the listener but nearly not in the third dimension. It is the same with "tetraphony" or "quadrophonie" which uses four reproduction sources in the horizontal plane. Therefore we began a complete study of 3D-Stereophony. Then the four loud-speakers completely surround the listener; any two of them may be considered as a stereophonic couple, which is known to give the best spatial effect (3). For this purpose, we worked on different non-plane configurations of four sources (tetrahedrophony (4) and others). The results of our actual tests, not yet achieved, seem to give the preference to this "3D-Stereophony".

References:
(3) R. CONDAMINES, Contribution à l'étude psychophysique de la régulation du niveau sonore dans la prise de son, Thèse Docteur ès Sciences, Paris VI, 1972.
One of the basic problems in room acoustic measurements has always been the direction of a certain reflection and especially its frequency content. For instance, what is the effect of a certain type of ceiling structure and surface in a concert hall or a studio.

Gating techniques are a simple solution to these problems.

A toneburst signal is transmitted from a loudspeaker and measured at the listening position as indicated in Fig. 1.

The received signal consists of a number of bursts arriving from different directions at different times. Any of these can be measured separately using the measuring gate as indicated in the figure.

Since the toneburst frequency can be swept, the frequency response of any of these reflections can be measured.

The technique is also applicable to measurement of absorption - and reflection coefficients as a function of frequency and direction.
MAGNETIC TAPE RECORDING - WHERE HAVE WE COME FROM AND WHERE ARE WE GOING?

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INTRODUCTION

On the 29th August in the year 1885 Edison's assistant, a Mr. S. Tainter, filed a patent for a magnetic recording system which used a fountain pen loaded with magnetic ink to write on a rotating cylinder. In the recording mode the pen was attached to a diaphragm which vibrated the pen. In the replay mode the pen was replaced by a magnet which was intended to vibrate the diaphragm in sympathy with the magnetic ink recording.

This was probably the birth of magnetic recording, but perhaps the real credit for the invention of practical machines should go to Valdemar Poulsen, a Dane. Poulsen patented a wire recorder on the 1st December 1898 and subsequently developed several practical wire recorders for use with telegraph/telephone systems.

Magnetic tape in the form of a steel tape appeared on the scene about 1900 and remained in the form of a steel tape until the second world war, the Blattnerphone developed by Ludwig Blattner being used by the British Broadcasting Corporation in 1931. This machine ran 6mm steel tape at 1.5 m/s providing 20 minutes playing time; editing was by soldered joints in the tape.

During the years 1939 to 1945 there was intense development work on recording systems in Germany, with the result that by 1945 there were plastic based tapes with an iron oxide coating replacing the steel tape. The practical magnetic recorder had arrived. At a tape speed of 77 cm/s the frequency response was 4 dB from 60 Hz to 10 kHz, with a signal-to-noise ratio of 50 dB.

THE MODERN RECORDER

Today the performance of a cheap cassette recorder running at 4.76 cm/s is equivalent to that of the 77 cm/s 1945 vintage machine, but it is interesting that in the professional field the tape speed has changed little and that the improvement in signal-to-noise ratio is only 1 dB per year.

The specification of a modern recorder always quotes signal-to-noise ratio, frequency response and many other parameters. These often meaningless specifications, including distortion, are blindly accepted as a figure of merit for machines and for tapes. Certainly the specification has some relation to practical performance, but there are many other factors which contribute to the quality of the reproduced sound.

For example, if we consider ‘distortion’, the tape and the machine manufacturer specify third harmonic distortion at a single fixed mid-frequency and at a fixed and high recording level. Why? Because it is conventional.

In practice we are recording a wide range of levels and frequencies and we are consequently not only interested in the third harmonic distortion at a fixed point, but also in all other forms of distortion. These include the more objectionable non-harmonically related distortions such as intermodulation distortion and also sideband distortion caused by minor tape speed variations.

All recorders suffer seriously from this defect which does not show itself in the wow and flutter measured by conventional means. Flutter sidebands with a good quality professional machine may be only 40 dB below signal level at all recording levels. Is 1% distortion (this is -40 dB) acceptable at all levels?

This is but one of the shortcomings of the modern machine which will be discussed in relation to the design of tapes and of machines, but what is the next generation of magnetic sound recorders. Digital recording has many attractions — it also has new shortcomings.
INTRODUCTION

In recent years, numerous authors have applied the Preisach model of magnetization process in Rayleigh area to signal recording. Even when quite many partial problems have been elucidated, the application itself consisted in qualitative description only. Until now, no Preisach model has been presented that would provide quantitative calculations of tape recording characteristics, and whose limits of validity have been stated.

QUANTIFICATION OF THE PREISACH MODEL

The quantification of the Preisach model consists in a proper approximation of the Preisach surface $\mathcal{E}(H_s, H_m)$. It turned out that under the assumption of normal distribution of the coercivities, $H_s$, and interactive fields, $H_m$, of the particles in the active tape layer, and at statistical independence of both random variables, $H_s$ and $H_m$, the most suitable approximation proves the Gauss function. Under these assumptions model calculations of tape recording characteristics and remanent characteristics have been realized. The computed characteristics showed a good agreement with experimentally stated values of real tapes.

THEORETICAL LIMITATIONS TO THE PREISACH MODEL

The approximation holds in one half of the first quadrant of the Preisach model limited by the $H_s$-axis and the line $H_s = H_m$ (the remanent part of Preisach surface), where the whole recording process takes place. We extended the basis of the model over the whole first quadrant, which newly enabled us to study the magnetization processes in recording media also outside the area of the remanent phenomena. We named the new Preisach surface, as obtained in this way, the total Preisach surface. The coefficient of rectangularity of the hysteresis loop of the tested medium equals to the ratio of the volume of the remanent part of the surface to the volume of the total Preisach surface. We found the experimentally stated shape of the total Preisach surface in close neighborhood of the $H_s$-axis to be direct proportional to the course of reverse susceptibility, which is not any single-valued quantity. Therefore also the total Preisach surface proves not single-valued in this region (and therefore is statistically unstable). For high values of random coordinates ($H_s, H_m$) the total Preisach surface reaches negative values. Since negative values of probability density have no physical sense, the zero contour of the total Preisach surface represents the limits of validity of the Preisach model.

RESULTS AND CONCLUSIONS

From the presented paper follows that the remanent part of the Preisach surface can be used for computations of the tape recording characteristics. It further brought the limitations to the validity of the model and the region of statistical stability of the total Preisach surface.
SUBJECTIVE AND OBJECTIVE METHODS OF VOICE IDENTIFICATION

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The problem of identifying a person through his/her recorded voice did become an important issue since the practical development of the tape recorder, which allows for the easy obtainment of voice samples of the unknown telephone caller and the known persons who are suspected to be the unknown caller. Methods for voice identification are classified within a continuum spectrum of techniques, but two groups can be distinctively separated: subjective and objective methods. Subjective methods consist essentially of a) aural comparisons of perceptual features of the voice samples, and b) visual examination of spectrographic similarities between these samples. Objective methods are based on extraction of efficient parameters from the voice samples and producing a decision (identification of elimination) through a computer algorithm.

Methods of voice identification of course are not exclusive and can and actually are used conjunctively.

Two general aspects of the voice identification or elimination problem can be distinguished: laboratory experiments and practical applications, mainly in the forensic and in the intelligence fields.

Laboratory Experiments in Voice Identification

a) Subjective Methods using aural, short term memory process. Bricker & Prozansky (1) obtained an average percentage of correct identifications of 75%. Trials were closed, using 2 unknown speakers, sentences 2.4 sec duration.

Stevens et al. (2) used also aural, short term memory procedures, obtaining percentages of correct identifications from 18% to 6% for closed texts.

Hall (3) using mimicked voices by a professional mimic obtained a 75% of correct identifications.

Up to the present the largest experiment using speech spectrograms for voice identification or elimination purposes was performed by Tosi et al. (4) He obtained 6.3% of false identifications and 12% false elimination open tests, using up to 9 clue words, new contemporary spectrograms.

b) Objective Methods. These methods in general are not very well developed as yet. Experiments by Hair, Rekieta (Texas Instruments) and Becker et al. (5) were not conclusive. Su and Fu. (6) were obtained correct identifications from 100% to 10% according to the variables involved, using nasal consonants for purpose of identification.

A new approach for automatic identification was introduced by Tosi and associates (7) using choral spectra of speech. Errors of false identification were 5.5% comparing speakers using three different languages. With this choral spectra method problems of temporal alignment and lost of transients that plague other methods are eliminated here.

Field Application of Voice Identification Techniques

So far this field application of voice identification had been based on a combination of aural and visual examinations of speech spectrograms commonly known as "voiceprints". Up to Dec. 1976 approximately in 70 criminal trials in the USA and three in Canada this evidence has been introduced successfully, except in one case that it was not accepted by the court. Several criminal laboratories in the USA have a voice identification unit, and the Institute of Voice Identification of Michigan State University trains persons in this art. Some scientists deny the reliability of this method, arguing that it is too subjective to be accurate.
SEVILLA SYMPOSIUM

Hearing and Industrial Noise Environments
INTRODUKTION

An increased "quality of life" does also require noise reduction of machinery and equipment utilized at working places as well as at home. A condition for this is the existence of noise limits and labelling so that the noise performance of the equipment can be compared with the limits. However, limits can only be established when measuring methods are agreed upon - if possible on an international basis.

In view of the number of machines to be measured, simple methods must be adapted and the outcome of the measurements must be sufficiently reliable and unequivocal. The concept of A-weighted sound power level lends itself to the purpose. It is a one-figure quantity that can be measured in different ways, and the outcome correlates reasonably well with subjective noise-ranking.

MEASUREMENT METHODS

Sound power level measurements can be taken either in a free-field (over a reflecting plane) or in a reverberation room. In both cases broad-band and pure tone noise signals can be measured with an accuracy depending on the efforts. Such methods are standardized and described in International Standards ISO 3741, 3742, 3743, 3744, 3745 and 3746 recently issued.

ISO 3743 describes a special method in which a test room with a reverberation time of about 1 s is used. By letting the reverberation time increase slightly towards low frequencies the sound power level (A) can be calculated from the sound level in dB(A) by simply adding a constant.

The room can be established at low costs, the measurements are easily taken (rules for the number of microphone positions are specified) and the work can be done by non-acousticians. It is assumed that the method will have many applications on small machines, e.g. household appliances, for which test codes are in preparation by IEC.

REFERENCES

S. Damgaard Kristensen: A special reverberation test room for sound power determination.
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INTRODUCTION

Textile machines are generally very noisy. In weaving mills where shuttle looms are used, noise levels can reach values as high as 100 or 105 dB(A). Permanent hearing loss induced by exposure to the noise of these machines is well known. However, in many mills, thanks to new types of looms, better machines laying out..., noise levels are lower.

The survey, the results of which are reported here, aims to:
- determine the noise level caused by different types of looms,
- study, on an experimental basis, the influence exerted by machines spacing and room characteristics on the final noise level in the factory.

SURVEY AND RESULTS

Twenty-two mills have been surveyed and five types of looms studied. The looms differ from one another mostly in the technical means used for passing the horizontal threads between the warp threads. Grips, bullets, jets of air or jets of water can be used instead of the conventional shuttle. The noise levels measured 1 m away from the external frame of the machines are given hereunder:
- shuttle looms: from 87.5 to 92.5 dB(A)
- air jet looms: from 86 to 97 dB(A)
- bullet loom: 87.5 dB(A)
- grip looms: from 62 to 66 dB(A)
- water jet loom: 77.5 dB(A)

These levels are those which would be found for one loom and in a hemispheric free field. In a mill, noise levels measured at the same distance are higher and depend mostly on machines spacing and room characteristics. The difference between the level measured "in situ" and the noise level caused by a single machine in a hemispheric free field is called "amplification". This difference ranges from 4 dB(A) to 10 dB(A).

CONCLUSIONS

The survey shows that noise levels in weaving mills can be reduced in using quieter looms, in improving room acoustical characteristics and in allotting reasonable space to the machines. The noise levels which have been found range from 87 dB(A) to 102 dB(A); even between mills equipped with the same type of looms a difference of 6 or 7 dB(A) could be found.
INTRODUCTION

The noise exposure for men standing the normal one-in-three schedule in propulsion machinery spaces on some large naval ships is 94±3 dBA 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) accruing from 12 days continuous steaming under these conditions.

THEORETICAL BASES

Relations between Permanent Threshold Shifts (PTS) and TTS have been used as bases for protective and/or preventative legislation establishing 90 dBA exposures for 8-hour day, 5-day weeks for 20 years to be the maximum safe limits. Few field data exist for validating exposure periods of 4-hours on and 8-hours off for 7-day weeks.

EXPERIMENTAL TECHNIQUES

Three groups of men were selected from the Engineering Department as follows: 11 who stood watches in "quiet" places (≤69±5 dBA); 28 who stood watches in continuous noises of 94±3 dBA, 11 of whom were some sort of hearing protection (≤8±7 dBA) 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 such that each 4-hour period of the 24-hour day was covered at least once during the 12 days at sea.

RESULTS AND CONCLUSIONS

It turned out to be impossible to get the men to take their audiograms immediately after the termination of their watch. Therefore, a comparison was made between their baseline (pre-sailing) audiogram and their audiograms both before and after watches after at least 10 days at sea. The unprotected noise-exposed group showed consistent TTS at 4, 6, and 8 kHz (which reached the 12 level for 4 kHz in their right ears which were always tested after their left ears). When the baseline and both the fore-and post 4-hour watch period audiograms were averaged for both ears at 3, 4, 6, and 8 kHz the unprotected noise group showed a 5 dB higher hearing level (HL). Although these group means were not statistically significant, they are at least indicative that previous exposure to these noise levels may be hazardous to their hearing. Similarly when looked at in a different manner, 8 men of 39 who had usable audiograms showed some consistent TTS. Of these, 7 were in the unprotected noise-exposed group. The remaining one was in the quiet group but he consistently listened to rock and roll music at very low levels via ear phones in his off-watch hours.
The aim of this study is to define a new target for noise control in factories, workshops and other places where people stay for professional purposes: the acoustical comfort, which would allow everybody to use his ear as an instrument of communication and vigilance, permitting him to survey properly his environment in order to prevent, at the best, many sorts of accidents. Legitimacy, conditions and opportunities of such comfort will also be discussed.

First will be quoted some statistics showing that professional ear diseases or injuries represent a very small fraction of the total number of occupational accidents. But these results must be interpreted with some caution: everybody knows that many work places are much noisier than they might be after proper acoustical treatment...

Nevertheless practical eradication of occupational deafness seems easily attainable within the foreseeable future, of course such a result will be greeted with favour, but it should, by no means, be interpreted as the end of active research in direction of improved control of industrial noise. Another step is now in view: in many professional environments, the human ear could be restored in its most primitive function, that is, as an instrument of vigilance and survey of surroundings.

In fact there exist many similarities between hearing and sight. As well as the eye is able to perceive a very very small amount of light quanta (i.e. few photons), so is the ear also sensitive enough to approach, in one way or another, the quantal levels of mechanical vibration (1). But on another hand the angular field of audition is always $4\pi$, while the field of peripheric vision is only $2\pi$ and that of foveal vision only a fraction of steradian. And the tridimensionality of auditory perception, if even incomparable with the visual one, appears certain (2).

Restoring the vigilance function of hearing leads one to expect a correlatively reduction of accidents, as many experts of safeness believe, in spite of the fact that we cannot compute or measure now the exact value of this correlation. In my opinion, a reduction from 80 to 50 dB of the noise would reduce the number of accidents by about 30% or more.

The means of such a reduction are well known and in continual improvement: 1°) control at the sources, by a better design of machines and an extended use of insulating devices; 2°) control during propagation, by passive devices like absorbing or deflecting walls or by active sound absorbers (3).

Continuing on this way, the new objectives of Noise Control could be the establishment of a true ACOUSTICAL COMFORT LEVEL anywhere people have to work or stay for extended periods of time. This level can be tentatively located at 20 or 25 dB below the thresholds of ear injury, more accurate determinations could be made in special cases, using masking, signalling and other technical data.

More substantial progress will need much well concerted and sustained effort: cooperation between acousticians and machine designers should be early, regular and highly encouraged by ministries and other public authorities.


Abstract

Hitherto, it has been common practice in most countries for the sound attenuation of hearing protectors to be measured by means of a threshold shift method, using pure tones as sound stimuli and an anechoic room as the acoustic environment. About 120 different models have been tested in the Physikalisch-Technische Bundesanstalt by this method over the last 15 years. A compilation of the results is given, showing typical performance characteristics of various kinds of protectors and the physiological limitations of protection.

In addition, several alternative methods which were proposed in the past will be discussed, including the newly drafted international standard method, which uses third octave bands of noise instead of pure tones and a quasi-diffuse sound field instead of a free sound field, as well as different high sound pressure level techniques and an artificial head method.

If in any real noise situation, the noise spectrum is not wholly known, the sound attenuation data, usually presented graphically versus frequency, cannot easily be taken to evaluate noise reduction and select the best hearing protector model. Consequently, several simplified methods for calculation noise reduction and rating purposes have been proposed. Their use and limitations will be discussed.
INTRODUCTION

Le traitement du bruit comme risque hygiénique et la valoration de leur action patologique sur l'homme, dans la législation espagnole, se trouve très lié au commencement de l'usage de la machine dans les procès industriels.

TEXTE

Le chemin parcouru par la problématique du bruit industriel, dans le contexte legal espagnol, débute dans le XIXème siècle avec l'approbation du "Catalogue des Mécanismes Preventifs" par Royal Ordre du 2 Août 1900. Après un période où la législation présente cette problématique d'une façon très générique, dans la décennie des années 70, par l'approbation de l'Ordonnance Générale d'Hygiène et Sécurité du Travail, l'élaboration d'un Plan National d'Hygiène et Sécurité du Travail et la rédaction d'une Norme Technique Réglementaire sur l'homologation des protecteurs auditifs, on a commencé la définition d'une façon plus concrète de l'action de l'administration dans ce domaine.

Malgré l'existence de cette gamme de normes et réglementations sur le contrôle, surveillance, etc, dont leur élaboration a été chargée à plusieurs Ministères, nous pouvons constater une manque de cohérence dans leur mise au jour et planification, motivée, peut-être, par la méconnaissance globale de cette problématique.

Le souhait de présenter l'évolution suivie après le commencement de l'étude de la problématique du bruit industriel, jusqu'au présent, aussi bien que l'intérêt dans l'exposition d'un tour d'horizon sur les projets déjà existantes pour un très proche futur, sera le sujet fondamental de cette communication.

Le rôle des Services Médicaux des Entreprises, l'action de l'Inspection du Travail, la ligne suivie par l'Institut National de Médecine et Sécurité du Travail et la présentation du Service Social d'Hygiène et Sécurité du Travail, dans leur agissement direct sur l'évaluation, contrôle et normalisation du risque hygiénique du bruit, constitueront les points basiques du "Traitement du Bruit industriel dans la Législation Espagnole".
ICA - SCOPE WORKING GROUPS (SPECIAL SESSIONS)

WG-1. - Hearing Thresholds of Isolated Human Populations.
WG-2. - Sound Propagation Outdoors.
WG-4. - Effects of Noise on Wildlife Communication.
WG-5. - Effects of Noise on Sleep.
HEARING THRESHOLD LEVELS OF "ISOLATED" HUMAN POPULATIONS: A BROAD SEARCH FOR NEW INFORMATION

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INTRODUCTION

Industrial workers who are exposed to substantial amounts of noise over periods of many years incur the risk of significant permanent noise-induced hearing loss. To reduce this risk many countries are now setting mandatory limits of noise exposure. These limits (typically 85 dB A-weighted for eight hours daily) are based mainly on hearing threshold data for human populations exposed to various levels and durations of occupational noise. Other data show that most people living in technologically advanced countries but not exposed to occupational noise develop appreciable loss of hearing in old age. Estimates of the magnitude of this loss as a function of age vary substantially which may indicate that presbyacusis, as usually estimated, includes appreciable hearing loss due to the prevailing noises of modern life (Glorig et al., 1962). The concept of a "safe" limit of noise exposure therefore remains elusive.

NEW PERSPECTIVES

In 1962, Rosen et al reported that relatively little presbyacusis occurred among the Mebans, a preliterate people living in an area of the Sudan which happened to be noise-free. This finding was confirmed by high frequency audiometry two years later. Not only had the Mebans superior hearing in old age but they were free of several ailments which are prevalent in industrialized countries such as elevated blood pressure. Several other studies of hearing in "far-away places" have served to focus attention on the "geographical" factors which may affect measurements of presbyacusis. For example, studies in Ghana, Jamaica, Nigeria and elsewhere suggest that statistical presbyacusis may include hearing loss related to diet (lipid level, presence of toxins, etc.), general medical disorders and genetic factors (e.g. Rosen et al., 1970; Hinchcliffe, 1973).

METHODOLOGY FOR FUTURE STUDIES

From the variety of information now available an epidemiology of hearing loss is beginning to emerge. A well-designed program of hearing studies covering specific populations in various parts of the world could shed valuable light on the factors, including noise pollution, which determine hearing level as a function of age. To be of maximum value such studies should employ standardized audiological instrumentation of superior design and should include serological, neurological and ophthalmological examinations, determinations of general physiological condition and the prevalence of disease, genetic classification, and the collection of relevant environmental and nutritional information. A standard methodology is clearly needed for the collection, analysis and presentation of data including the determination of biological and chronological age. The populations selected for study should be clearly identifiable and distinctive by reason of geographical isolation or social structure, and preferably untouched by modern technology since noise exposure, where appreciable, is usually difficult to quantify. These matters are of continuing interest to the members of ICA-WG1 (Special Conference, 1976.)

REFERENCES

TEMPORARY THRESHOLD SHIFT AND VEGETATIVE REACTIONS CAUSED BY NOISE DURING LIFETIME.

Jansen, G. and Griefahn, B., University of Mainz

Noise does not only cause an increase of hearing threshold but also a lot of vegetative reactions. A most simple indicator of these reactions is the decrease of fingerpulse amplitudes during noise caused by a reduction of peripheral blood flow.

Hearing threshold and temporary threshold shift. In aged people with higher hearing threshold the TTS is smaller than in younger subjects. These differences due to age only could be found in male subjects not in the females though the hearing threshold was the same, in the aged men and women. The differences can be explained by atherosclerosis which begins later in women.

Hearing level and fingerpulse amplitudes. With increasing age the noise-induced reduction of peripheral blood flow becomes smaller. But in civilized people age is accompanied with an increase in hearing threshold and atherosclerosis. From different experimental trials we have some indications that the smaller extent of reaction is caused mainly by atherosclerosis.

Temporary threshold shift and fingerpulse amplitudes. In people with an excellent hearing a negative correlation between TTS, and noise induced-reduction in fingerpulse amplitudes had been proofed, whereas people with bad hearing and aged people did not show any correlation between both parameters.

Investigations of isolated living people. Further investigations should be done in order to find out the factors causing presbyacusis. We know that TTS and the decrease of fingerpulse amplitudes are both caused by a vasoconstriction of the small blood vessels but the extent of the vasoconstriction as well as the correlation between both parameters is an indicator of atherosclerosis. They should be recorded both in further investigations.
SOME FEATURES OF HEARING THRESHOLDS IN GREENLAND

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UMANAK

In infancy and childhood many Greenlanders achieve permanent hearing losses due to middle ear infection.

In a hunters' district on the West Coast (Umanak with a total population of 2,163), 372 out of 374 school children were examined audiometrically. 50 of 150 children (33%) living in the town of Umanak itself had a permanent hearing loss, and 32 out of 222 children living in small villages or settlements spread over the district had hearing losses (14.4%). In Denmark proper the comparable figure is 4%. The hearing losses in the villages and settlements were smaller than those found in Umanak itself, possibly due to a healthier diet, natural Greenlandic food, better social conditions.

25% of the hearing impaired children had hearing losses at 4000 and 8000 Hz only -- (shooting), 50% from 25 to 40 dB losses (average of 250-2000 Hz), and 25% more than 40-dB in one, or both ears, about half of the cases being found in either group.

ANGMAGSSALIK

The majority of the population of the Angmagssalik district (with a total population of 2,479) on the East Coast of Greenland were examined audiometrically and otologically. All school children, all persons at 50 years of age and upwards, and half of the population in the age groups in between were tested, whereas it was not possible to test the small children. There were found 733 persons with hearing losses which was more than half of the persons examined. 60 school children had moderate to severe hearing losses.

ALL GREENLAND

Most of the cases of middle ear infection in children become inactive when they grow up, but with residual defects of the tympanic membrane and hearing losses. Otosclerosis is infrequent in Greenland and has been found in 2 cases in Eskimos yet. Hereditary sensorineural disorders are found at some of the isolated places where everyone is related to each other, severe deafness after meningitis also. Noise traumas are frequent in certain districts also amongst women and almost all children above 10-12 years. This is because they accompany the hunters and take part in the shooting. Noise in the fish canneries is responsible for other cases. And, finally the population grows older, therefore we see more and more cases of presbyacusis, proportionally, however, still less than in Denmark proper.

The Greenlanders' motivation for treatment, including hearing aid use, is different from that of the Danish population due to the great number of people with hearing losses of 30-40 dB at least. They simply do not realize that this is abnormal at all.
INTRODUCTION

This paper describes several phenomena that can interact during sound propagation outdoors. Understanding of the mechanisms is important so that, for example, measurements of road vehicles under standard test conditions can be used to calculate community noise at greater distances; also, airport noise should be predicted accurately so that noise-reducing measures are applied where the noise problems are worst.

EFFECT OF GROUND SURFACE (see also S. Lindblad and S. Thomasson, following paper)

Sound is almost perfectly reflected at the surface with no change of phase when the ground between source and receiver is hard (concrete, asphalt, packed earth). For an elevated source and receiver, there is interference between direct and reflected waves because of the path-length difference.

When \( h = h_1 = 1.2 \text{ m}, d = 15 \text{ m} \) the frequency of the first interference minimum is \( \approx 900 \text{ Hz} \); when \( d = 7.5 \text{ m} \) this frequency drops to \( \approx 450 \text{ Hz} \).

When the surface is acoustically soft (grass covered) there is a phase change on reflection, which leads to a region of excess sound attenuation near the surface that can be as large as 40 dB as \( h = h_1 + 0 \), or \( d + \infty \).

This excess attenuation is observed only at high frequencies because a ground wave penetrates into the shadow region at lower frequencies.

EFFECTS OF METEOROLOGY

Sound paths are concave downwards in downwind propagation or in a temperature inversion. Sound can then penetrate the shadow region more easily. Excess attenuations near zero dB at most frequencies are a commonly observed condition, and the sound level far from the source is then given by the effects of simple geometrical spreading and molecular absorption; sound levels higher than this value are rare.

Downwind or under conditions of temperature lapse, refraction of sound paths leads to the formation of a shadow region near the ground. This coexists, over soft ground, with the shadow region caused by small values of surface impedance. The excess attenuation due to the two types of shadow region is not in general the sum of the two individual effects. Instead, the maximum excess attenuation commonly observed is in the range 15 to 25 dB — a similar maximum value to that often observed for barrier attenuation (see U. Kurze, following paper). This leads to speculation about "flanking paths" common to each of these mechanisms of shadow production: one possibility is scattering by turbulence located in the volume above the shadow (see E. Brown, following paper).

EFFECT OF TOPOGRAPHY

Schemes for predicting sound levels near highways, airports, etc. normally assume the presence of flat ground. Highway prediction schemes usually allow for the excess attenuation of barriers (or equivalent such as rows of houses). Intervening hills can also act as a barrier. Sometimes this excess attenuation is already achieved at no cost because of finite surface impedance. When the receiver is elevated above a generally flat ground, as when houses are located on a hillside facing the source, the effect is to reduce the excess attenuation otherwise provided by the finite impedance of the ground surface.
This paper confines the discussion to the problem of propagation above a plane ground and treats some general results of Reference 1 below, where further references are given. For the sake of simplicity all effects of the atmosphere (such as attenuation, inversion etc. that may be important) are omitted in the following.

**THEORETICAL MODEL**

As to the boundary condition of the ground there are essentially two different theoretical approaches; that the ground is locally reacting (normal impedance) or that the ground could be described by some kind of layer or layers. Under certain conditions it is possible to show that these two approaches may be unified; if the refraction index of the layer \(|n| > 1\), i.e. if the sound velocity of the layer is much smaller than that of the air. There are several interesting results of such an analysis. First we get an explicit expression for the point admittance, viz.

\[
v = (\frac{\rho_0}{\rho_1}) n \tan(nk_0z_1)
\]

in terms of the quantities of the layer. Secondly, by assuming the layer to be porous, we get a certain knowledge of the frequency dependence of the density of the layer \(\rho_1\) and of \(n\), and thus the point admittance. Thirdly we may develop a new method of estimating the ground impedance. Finally the model makes it possible to obtain an approximate solution of high accuracy to the problem of the

**IMPEDANCE BOUNDARY**

The solution of the sound field emitted by a point source is well-known, and as the solution in unique, different papers on this subject differ only in their analysis and possible approximations. In this context there has been some trouble whether the solution is to contain "surface waves" or not. As to the mathematical side of it — if the solution should contain a certain term, here called "surface wave" term — we may choose this term to be seen or not, according to our wishes, as it may be hidden behind integral expressions or infinite series. As to the physical side of it — if this term could dominate the field in any practical case, we may then speak of "surface waves", — it has been verified that this may certainly be the case above a porous layer at low frequencies. How the essential trouble left with the solution to this problem is how to estimate the

**GROUND IMPEDANCE**

As mentioned above, the assumption that the ground could be described by a porous layer leads to a known frequency dependence of the admittance \(v\). The unknown constants of eq. (1) may be reduced to four real ones, which may be estimated by measuring the frequency response when the source and the receiver are located close to the ground by fitting a theoretical frequency response to it. There seems to be a quantitative agreement with other methods, but an advantage with the one given here is that we obtain a simple continuous description of the frequency dependence of \(v\), and that it gives a direct check of the desired quantity — the sound field. So far the method has shown to be applicable to normal outdoor surfaces.

**REFERENCE**

(1) S-I Thomasson, J. Acoust. Soc. Am., "Sound propagation above a layer with a large refraction index". (To be published in 1977.)
SOME PROBLEMS TO BE RECOGNIZED IN EVALUATING THE BEHAVIOUR OF BUILDINGS EXPOSED TO NOISE AND VIBRATION

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A theoretical or an experimental approach to the problem may be conveniently based on a close study of the chain: source - path - receiving end. At the first step all types of sources should be classified accordingly, because, e.g. traffic noise may affect the building in a different way than the noise of household appliances; also in-door and out-of-doors position of the source should be distinguished. Briefly, all existing noise and vibration sources inside and near the building should be classified (some attempts have already been made /1/). The organization of the classification at this moment is of no prime interest to us; however, it should be based upon effects of the source characteristics on the noise generation inside the building.

Noise and vibration paths are strongly related to the source type and source position. Apparently, the main point of the research is the behaviour of sound and vibration direct and side paths. At first we may neglect the out-of-door sources, thus reducing the number of paths to those which are comprised in the elements of the building structure: walls, ceilings, floors etc. which then become objects of our study. These partitions should be also classified with respect to their characteristics (i.e. light and heavy partitions, double partitions, etc.), but it seems not to be quite satisfactory; the bindings should be considered too. Furthermore, the very building behaves as an entity or as a system of a multiple degree of freedom; it should be emphasized that an experimental (or theoretical) research should therefore cover the effect of the type of the building structure on the noise generated by a source located at a different heights of the building. On the other hand - smaller elements than a partition should be studied too. In modern buildings partitions are of non-uniform cross sections, owing to the piping installed, which - as measurements in the field show - have a definite influence on noise transmission.

As a receiving end we may take a room of a normalized absorption. In evaluating the permissible noise levels, some kind of room classification is also desirable, which has been already worked out in many national standards. There is another problem in the receiving room, and that is measuring and calculating the noise produced by vibrating boundary surfaces, which deserves a due attention.

A particular question on a building behaviour is the ageing of the building. It seems that it has a second order effect on the acoustical characteristics of a structure, and therefore this problem may be put aside for the present.

Better knowledge of the noise and vibration transmission through building elements will also aid to choosing appropriate materials for vibration isolation.

Finally, the problem of noise propagation in buildings may be approached both theoretically and experimentally; the experimental data will serve us to form theories and practical formulas to calculate the desired quantities.

To conclude: a suitable classification of noise and vibrational sources, transmission paths and receiving rooms will enable us to recognize already solved and to be solved problems in acoustical characterisation of buildings.

There are several ways in which vibration and noise transmit in buildings. The initial excitation may be by sound acting over a surface or by localized mechanical excitation of the structure. Transmission may be through airborne or structureborne paths, the structural vibrations resulting in noise radiation from surfaces. The conversion of vibrational to acoustical energy may be by a direct path through common wall or floor, or indirectly following flanking transmission. The direct path is normally the most important, but flanking transmission becomes significant if the direct path is highly attenuating (greater than about 45 dB) or if the reception point is at some distance from the source in a multistorey building. The transmission of vibrations in the structure occurs through longitudinal, transverse and bending waves. The bending waves are the most important for re-radiation of noise because of the relatively large movements which they generate normal to the surfaces. The study of bending wave transmission is complicated by the variation of both propagation velocity and damping which occurs with frequency.

The basic approach to a study of flanking transmission is to consider transmission across plate junctions and this has been the main area of theoretical study. It is possible to derive the ratio of mean square velocities of the components of a four plate junction, giving good agreement with experiment. However, there is lack of information on vibration transmission in real structures and a recent research programme has been undertaken to provide this information.

An electronics system has been developed to give rapid measurement of the mean square velocity at a point on a surface following impulsive excitation of the building structure. Impulses were repeated at intervals of two seconds and the resulting accelerometer response at the pick-up point was integrated to give velocity, squared and then integrated over the duration of the response. The result, which is the average mean square velocity at the reception point, is displayed digitally. Repetition at a number of points over a surface gives the average mean square velocity of the surface and this is related to sound radiation from the surface. The time delay between the impulsive excitation and the arrival of the resultant vibration at the reception point is also given by the apparatus.

Measurements have been carried out in a number of buildings to investigate both vertical and horizontal propagation and the importance of lightweight to heavyweight junctions in energy transmission. If the input is to a floor (i.e. a heavy plate) there will be relatively good transmission of bending waves into an attached wall (i.e. a light plate). Conversely, input to the wall gives lower transmission to the floor.

The power of the sound radiated into a room from vibrating walls and floor can be obtained from the mean velocity and radiation coefficient of the surface. The value of the radiation coefficient is frequency dependent but approximates to unity above the critical frequency, which is different for light and heavy constructions. In general, the critical frequency was about 125 Hz for the floors which were investigated and about 400 Hz for the walls. It is possible for the sound radiated from the walls of a room to be similar in level to that from the floor, so that insulation of the floor will not be effective in noise reduction.
In several series of experiments, physiological and behavioral parameters of beef and dairy cattle, sheep, ponies, swine, and mink were studied to determine the effects of different sources and levels of noise on these animals. Data covering a period of 12 months showed no evidence of an effect on milk production of dairy cattle resulting from flyovers by jet aircraft or proximity to an air base (6). Feedlot beef cattle, horses, sheep and lactating dairy cattle were exposed to from 4 to 8 sonic booms per day and observed behavior reactions were minimal. When eating behavior and activity of ponies, weanling calves, finishing steers and cows with calves were monitored before, during and after the animals were exposed to two simulated sonic booms (about 200 newton/sq. m), all animals showed a clear startle response but quickly returned to pre-boom activity. Of importance was the total absence of general panic in any of the animals. The behavior of female pigs was studied by using an electrical sparking device which caused a bluish white spark and made a distinct hissing and cracking sound. Many of the pigs were bothered by the spark at first, but almost all of the pigs showed very rapid adjustment to it. There was no association between the average time required by the pigs to return to feeding and their average daily gain (5). To determine possible harmful effects of noise, pigs, boars, and sows were exposed to reproduced aircraft and other loud sounds at various stages of the life cycle. Feed intake, efficiency of feed utilization and rate of gain of treated and untreated pigs were similar. The reproductive cycle was not affected by loud sounds (1). Reaction to sound was initial alarm followed by a quick return to normal activity and the absence of continual arousal (9). The cochlea of the ear from experimental pigs in the above study were examined and no evidence of injury or changes suggesting impaired auditory function were found (7). An experiment was conducted on two commercial mink farms to determine the effects of simulated sonic booms (8 booms/day with overpressures of 22 to 90 newton/sq. m) upon reproduction (pregnancy, parturation, and kit production) of farm-raised mink. Reproduction of the mink that were boomed was normal (8). An interdisciplinary experiment was conducted to determine the effects of 3 real or simulated sonic booms (about 290 newton/sq. m) on farm-raised mink. In this study, the exposure of mink to intense sonic booms during whelping season had no adverse effect on their reproduction or behavior. On the basis of these experiments, it seems reasonable to conclude that loud noises have little or no effect on the behavior and productive capacity of farm-raised animals, at least at the intensities and frequency of occurrence to which they were exposed.

REFERENCES

(1) Bond, James et al., USDA, ARS Tech. Bull. No. 1280 (1963)
(2) Bond, James et al., Livestock Environment (1974) 170
(3) Bond, James et al., USDA Report No. FAA-EQ-72-2 (1972)
(6) Parker, J. B. and N. D. Bayley. USDA, ARS 44-50 (1960)
(8) Travis, H. F. et al., USDA, ARS 44-200 (1968)
A review of the literature on the effects of noise on wildlife reveals that the acoustic levels and dose to which test wildlife have been subjected are not well quantified. In some cases, the noises of interest are described only by their source. In others, simply an A-weighted or C-weighted level is given. Very few papers discuss the temporal patterns of noise and their effects on wildlife, since sound level measurements at a “wild” animal’s ear can not be obtained in a conventional manner. Nonetheless, a fairly precise picture of the spectral and temporal pattern that exists throughout a habitat area can be predicted, using the method described in this paper. This information is required to accurately assess the environmental impact of noisy activities on wildlife.

The method, adopted from one developed to predict aural detectability of military vehicles, utilizes a program that can be run on a desk-top computer. The program uses source spectral and temporal characteristics as input data, as well as readily measured atmospheric parameters, and terrain and vegetative parameters that can be fairly easily estimated. Also needed are the distance-time distribution of the noise source from, and the hearing characteristics of, the subject animal and noise statistics of the background sound in the habitat area. The program has been used with some success under actual field conditions to predict the acoustic impact of off-road vehicles on human forest users. This paper describes the computer program as it currently exists, and presents a proposed pilot project to verify its applicability to large mammals.
INTRODUCTION

The effects of transmission lines on wildlife have been little studied. Some known areas of conflict include: (a) wildlife mortality due to collisions (Willard and Willard 1973, unpublished), (b) wildlife avoidance of powerline corridors (1), and (c) bird electrocutions (2). Some other questions have received little or no attention. Is the corona-noise associated with EHV lines damaging to the hearing ability of birds nesting and perching on the towers? What will be the effects of electro-chemical oxidants on these birds? If birds and aquatic wildlife are able to navigate and communicate by weak electrical fields, as suggested by recent studies (3), what will be the results of introducing EHV transmission lines into the environment of these creatures?

This study is an attempt to identify the degree of use of EHV transmission line corridors by large mammals and the degree of use of transmission line towers for nest support structures by birds. From survey data, potential conflict situations were identified.

METHODS

To determine the degree of use of EHV corridors by large mammals, the authors conducted surveys along a 500kV transmission line in Idaho and Montana. Direct and indirect observations of large ungulates and other mammals were correlated with noise and force field observations below the lines.

To determine the degree of use of power towers as perches and nest sites for birds, hundreds of miles of lines were flown and driven from 1974-1976. Nests were identified according to tower locations, species of bird, and location of nest on tower.

RESULTS AND CONCLUSIONS

Along a 500kV powerline corridor in a heavily forested area of Idaho, large ungulates were found to use the cleared transmission line corridor at rates comparable to use-rates for other forest clearings. Corona noise did not empirically alter the behavior of these animals. Other problems to terrestrial wildlife associated with transmission corridors are discussed and solutions proposed.

Transmission towers, including EHV support structures were found to be very important as nesting sites for corvids and raptors. One area of extreme use was identified in northern Sonora, Mexico. Along 84.5 km of powerlines in open desert shrub, 39 nests were found. Two young birds, reared on a 500kV tower in Oregon, showed no signs of hearing impairment due to corona noise. EHV conductors produce noise at levels that have been found to produce damage in laboratory animals (4). Other possible problem areas are identified and studies proposed.

REFERENCES

(1) D.R. Klein, Science 1971, 173(3995), 393-398.
(4) U.S. Environmental Protection Agency, NTID, 1971, 300.5, (Call No. EPI.2:N69/8).
INTRODUCTION
L'ouverture de l'ère de l'aviation commerciale supersonique avec l'avion franco-britannique "Concorde" et l'avion soviétique "Tupolev T44" nous oblige à considérer le bang comme une nouvelle caractéristique de l'environnement.

Le bang sonique est-il nuisible ? Peut-il provoquer directement ou indirectement des troubles pathologiques chez les animaux ? Si ces troubles pathologiques existent, ont-ils des conséquences économiques sur les productions animales ? Quelles sont la fréquence et l'intensité maximale du bang que peuvent supporter les animaux domestiqués et sauvages ?

TECHNIQUES MISES EN ŒUVRE :
On utilise des générateurs ou des simulateurs de bangs dérivés du principe des tubes à choc :

Le générateur employé a été celui de l'Institut de recherche franco-allemand de Saint-Louis. Il peut réaliser tous les bangs des avions militaires actuels et du "Concorde" et produire des super bangs.

Le simulateur mobile de bang permet d'obtenir le bang caractéristique du Concorde (300 ma, 1 mb).

RESULTATS ET CONCLUSIONS :
Des bangs simulés d'avions supersoniques ont été produits sur des oiseaux, des chevaux de course, des poissons, des vies sauvages, des poules et des poulets en élevage industriel, des œufs de poules et de faisanes en incubation, des taureaux au cours du colt, des porcs en élevage industriel ...

Ce bangs d'intensité de 1 mb jusqu'à 6 mb ne provoquent que des réactions de survet chez les animaux. Il est important de noter que la répétition quotidienne des bangs entraîne rapidement une accoutumance des animaux qui ne réagissent plus au stimulus. Le comportement et les productions animales ne sont pas altérées par les effets répétés des bangs des avions supersoniques.

REFERENCES :
Ph. COTTEREAU : L'Aéronautique et l'Astronautique 1974, 5, 45, 28.
INTRODUCTION
Physiological responses of animals to various aspects of the physical environment are well known, however, responses to the audio environment are limited with most data reported for humans and small laboratory animals. For large domestic animals (cattle, sheep, swine) most reports are concerned with specific sounds such as sonic booms. It was the goal of this series of studies to characterize physiological responses for animals exposed to various sound environments.

THEORETICAL BASIS
Homeostatic responses to environmental variables (light, temperature, pressure) are well documented. Logically, sound, which is neurally sensed and interpreted by animals, may also alter physiological function.

EXPERIMENTAL TECHNIQUES
In trial I, sixty lambs were exposed to three sound types: (1) United States of America Standard Institute white noise (USASI); (2) Instrumental Music (IM); and (3) Intermittent Miscellaneous Sound (IMS). Each sound type was studied at 75 and 100 dB. Heart rate and respiratory rate were determined using telemetry techniques and growth responses were recorded. The treatment regime allowed for assessment of acclimation to sound exposures. In a second trial, lambs were exposed to 75 and 90 dB of USASI sound and measurements of thyroid function (Free T4 Index) were made. In a third trial, 79 ewes were exposed during the proestrus stage of the estrous cycle to two sound types (USASI vs. 4000 Hz pure tone) with two program techniques (continuous vs. intermittent play). All treatments were at 100 dB. All ewes were laparotomized and ovarian structures characterized.

RESULTS AND CONCLUSIONS
Reports on animal response to ambient sound suggest various physiological and psychological effects (1, 4) with altered growth rate reported for rats (3). Data here indicate that heart and respiratory rate respond in concurrence with other general stressors. Immediate heart rate response to music varied less than it did to white noise, suggesting that animals respond differently to different sound types. Acclimation to both sound type and intensity was apparent. Growth rate differences were found due to both sound type and intensity which concurs with previous work (3). In trial II, a significant decrease in T4 index was noted in lambs exposed to 90 dB sound which supports findings of audio oriented changes in endocrine function (2). Trial III indicated that 4000 Hz pure tones played intermittently significantly altered ovarian observations by increasing the number of corpora lutea per ewe. More recent work indicates a similarity between sound exposure and exogenous administration of follicle stimulating hormone which again suggests that the sound environment can alter endocrine function.

REFERENCES
INTRODUCTION

With the advent of EHV transmission lines, corona discharge audible noise began to be an environmental concern. When line noise is approximately 53 dB(A) or higher, some complaints can be expected from persons living near the line, and numerous complaints when levels are above 59 dB(A) (1). Concerns have also been raised about the possible effects of transmission line audible noise on wildlife (2-3).

This paper reports on efforts initiated by the Bonneville Power Administration (BPA) to study the possible effects that transmission line audible noise, and other operational parameters, may have on wildlife.

THEORETICAL BASES

Transmission line audible noise results when the localized surface electric field gradient on a conductor exceeds the corona onset level (1). The audible noise consists of a crackling, hissing sound, with a 120 Hz "hum" occasionally present.

Transmission line audible noise levels appear too low to cause direct physiological harm to wildlife, however, the possibility exists that the noise could interfere with wildlife communication or could affect wildlife movement near the right-of-way.

EXPERIMENTAL TECHNIQUES

Field studies were initiated in 1974 to begin to document the effects of BPA lines on wildlife. The experimental approach was to compare wildlife populations and behavior on transmission rights-of-way with nearby control areas. Study methods include direct observations, trapping, track counts, and time lapse photography.

In addition to audible noise, transmission line parameters considered in the studies included, electric and magnetic fields, access roads, and right-of-way vegetation. Study areas include a 500-kV ac transmission line in northern Idaho (Goodwin and Lee, to be published), a 1100-kV ac prototype line in western Oregon and a ±400-kV dc transmission line in central Oregon.

RESULTS AND CONCLUSIONS

The highest audible noise level measured so far in the above studies is 68 dB(A) which was recorded on the right-of-way of the 500-kV ac line beneath the single conductor per phase design. Although such high levels were annoying to humans, no adverse effects on wildlife were observed which could definitely be attributed to operational parameters including audible noise. If such effects occur, we believe that under field conditions, most are probably too subtle to be distinguished from effects due to other environmental variables, e.g., weather.

Transmission line designs which produce audible noise levels acceptable to people should also produce no significant adverse effects on wildlife.

REFERENCES

DISTURBANCE OF SLEEP BY TRUCK NOISE AND CONSEQUENT ADAPTATION

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Ottawa, Ontario, Canada

INTRODUCTION

Since we find that traffic noise is the most prevalent cause of complaints among residents the noise from a passing truck was used as the stimulus to study disturbance of sleep. Short tone bursts were used only briefly to verify the suspicion that the duration of the stimulus was the important variable in determining the response.

EXPERIMENTAL PROCEDURE

Subjects slept in the laboratory in a room fitted to resemble, reasonably well, a bedroom (rugs, drapes, etc.). Sleep was monitored by a Beckman electroencephalogram modified to operate unattended for up to 10 hours. Only frontal electrodes were used. The standard paper record was supplemented by recording the same signal on an AM tape recorder which had been modified to reduce the tape speed to 6 mm per second. The tape was analysed by playing back on a 38 cm/sec machine so that the signal was audible and could be analysed by standard audio equipment to produce amplitude-time records at any desired frequency in about 8 min. The lengths of these records were usually about 60 cm providing a good birds-eye view of the progress of sleep during the night.

Besides shifts in sleep level, behavioral awakening responses were obtained unambiguously by requiring the subject to push a button when he awoke for any reason whatsoever.

In determining how the disturbance varies with amplitude subjects were not tested on successive nights nor more often than three nights per week, to minimize the effect of any possible adaptation. They were tested in three groups, young (below 30 years of age), middle aged (mainly 40 to 50 years) and old (mainly above 65 years).

RESULTS

Figure 1 shows the average probability of a shift in sleep level as a function of peak level of truck noise for the three groups of which the middle aged group was the most sensitive. In addition the probability of awakening is also shown.

To test the degree to which adaptation may occur ten more subjects were tested (ages 16 to 35) sleeping 24 nights in succession with the truck noise at 65 dBA. For total number of shifts in sleep level no adaptation was evident. Fig. 1 Circles show the average probability of shifts in sleep level vs peak truck noise levels. The crosses represent the behavioral awakening response only.

Tone bursts with a duration of 0.3 second must have A-weighted levels about 40 dBA higher than the noises of a passing truck to elicit the same average response.
In studies of the effects of noise on sleep, the types of noise sources, noise descriptors, age and sex of subjects, response classification schemes used, and test methods are apparently as numerous as the number of laboratories engaged in the problem. There is a need for review of our accomplishments in order to point out similarities in study techniques and apparent trends in the data. Most importantly such a review would enable us to identify inappropriate techniques and additional requirements for research.

Based on a continuing review of literature published in the U.S., Japan, Canada, and Europe, it has become clear that most laboratory and field studies have used the sleep electroencephalogram (EEG) as the major dependent variable in determining noise effect. Because these studies used a common technique for evaluating the EEG, there was an underlying similarity in certain classes of response: no-change, arousal, K-complexes, etc. In addition, most investigators used units of dBA to describe maximum noise level, but frequently did not provide an adequate description of its time-course.

The result of this review indicated that in order to predict the probability that noise will alter an ongoing sleep stage, the noise descriptor must include terms accounting for noise duration and its maximum level. Effective level in dBA (EdBA) is one such descriptor.

How does the total nighttime noise environment affect subjective sleep quality? Arithmetic techniques for combining noise exposures are well known, but techniques for assessing total sleep quality are not. A French study (1) included a factor analysis of several sleep-quality questionnaires. This study suggested the 3 areas of questioning necessary to indicate total sleep quality: (1) time required to fall asleep, (2) the soundness of sleep, and (3) feelings of well being on awakening. Data from studies that included all or some of the three question areas were used to develop a "Composite Sleep Quality" measure for each study.

Although sleep quality data are sparse, there appear to be systematic degradations in sleep quality with increasing exposures to nighttime noise.

Reference

N.O-M Schneider, Evaluation Subjective Du Sommeil Normal Ou Perturbe Par Le Bruit Relations Avec Certains Indicateurs Physiologiques Et Traits De Personnalite
On a effectué des enregistrements polygraphiques sur des riverains d'autoroute, avant et après la mise en service de la voie, de manière à mettre en évidence les effets du bruit de circulation nocturne sur le sommeil.

L'échantillon est de 12 hommes, âgés de 27 à 50 ans (moyenne 34), de professions diverses. Les enregistrements ont été réalisés au domicile de personnes :
- avant l'ouverture : 3 nuits d'habitation au port des électrodes (non enregistrées) et 3 nuits enregistrées (situation témoin).
- 1 an après l'ouverture : 1 nuit d'habitation, 3 nuits enregistrées.

Chaque nuit on enregistre : 2 EEG, 2EOG, 1 EKG, par un système télémétrique. Une préexpérimentation avait montré la possibilité d'enregistrement in situ du sommeil, d'une qualité convenable pour les besoins de cette recherche.

On a réalisé un traitement analytique de l'effet sur EEG des événements acoustiques isolés; les bruits de pic qui provoquent des éveils ou des changements de stades sont très variables selon les individus les stades et les périodes de la nuit. Un niveau de bruit de pic de 55 dBA dans la chambre du dormeur a une probabilité moyenne de provoquer un effet.

La seconde partie du traitement est beaucoup plus intéressante : elle porte sur l'effet du bruit sur le sommeil global. On a constaté une nette diminution du sommeil lent (III + IV) en condition bruyante (45 dBA en Leq : niveau énergétique à l'intérieur de la chambre et 55 - 50 dBA en niveau de pic). La durée passe de 74' (38% du T.S.T.) en situation calme à 47' en situation bruyante (11% du T.S.T.). Le sommeil paradoxal n'est pas touché en durée pour les niveaux acoustiques observés.

Il reste à interpréter cette diminution du sommeil lent. Si on connaît les effets de privations massives sur certaines sécrétions hormonales et sur l'apparition de symptômes dépressifs ou hypochondriques, on ne semble pas posséder de données pour des privations partielles et répétées.

Le bruit agit donc sur le sommeil comme les dépressions sévères et les hyperthyroïdies. L'effet du bruit anéme le sommeil lent à un niveau comparable à celui d'une population beaucoup plus âgée (entre 50 et 60 ans). La diminution de ce stade nuit à la qualité de la restauration métabolique nocturne de l'organisme. À cet effet global du bruit sur le sommeil on fera sans doute correspondre un indice acoustique prenant en compte le bruit global de la nuit. La recherche du niveau de pic provoquant un effet ne paraît pas devoir être poursuivie.

Les résultats de cette expérience devront néanmoins être confirmés avant tout choix d'un indice acoustique et avant la fixation d'un seuil.

T.S.T. : Temps de sommeil total = somme de I + II + III + IV + SP

Financement : Ministère de la Qualité de la Vie

Collaboration : Laboratoire de Médecine Expérimentale de LYON (Pr JouvET - Dr MOURET)
Our very small knowledge about sleep disturbances by noise leads to the requirement of summarizing the results of all investigations published till now. Unfortunately, only a few papers are really comparable in method and evaluation. From these results the curves demonstrated in this paper had been calculated. As a rule, the results of the other investigations show the same tendencies, so the tendencies of the results in this paper seem to be realistic, whereas the numerical results have to be regarded as tentative.

The particular results are:

1. **Intensity.** With increasing intensity the extent of the reaction becomes greater. At a maximum level of 68 dB(A) the probability of being awaked is about 10%, the probability of all reactions of at least a diminuation of sleep depth is 35%.

2. **Type of noise.** Up to a certain degree the extent of a reaction is determined by the bandwidth. The greater the bandwidth the greater the reaction.

3. **Duration of stimuli.** Though stimuli of less than 1 second lead to a smaller rate of awakenings, the extent of reactions does not change up to a duration of at least 40 seconds.

4. **Number of stimuli per night.** The probability of O-reactions and awakenings caused by each single stimulus does not change from 12 to 30 stimuli per night, whereas the probability of an awakening reaction increases with less stimuli and decreases with more stimuli.

5. **Content of information.** The content of information seems to be most important. Until now we have no possibility for measuring the content of information, so that we are not able to calculate the exact magnitude of the content of information and of the other exogenic and endogenic influences.

6. **Other factors.** From these investigations which had been done only occasionally it seems that a higher intensity of light as well as a higher temperature leads to a longer lasting reaction.

7. **Age.** The extent of reaction increases with age, the probability of a O-reaction becomes less.

8. **Time.** There is some indication that the probability of an awakening reaction becomes greater during the second half of night.

9. **Sex.** Differences in the reaction due to sex had been found in a few experiments. But the results are controversial.

10. **Stage of sleep.** The influence of the depth of sleep on the extent of reaction had been well investigated. During Delta sleep the rate of O-reactions is greater than the rate of awakening reactions but is less than in stages 1 and 2. Within stage 1 both types of reactions are greater than in the other stages.

11. **Fatigue.** Investigations within the normal range of fatigue had not yet been done. Only after sleep deprivation of two consecutive nights a smaller reaction had been recorded.
Over the past several years we have been attempting to understand how sleep may be disturbed by auditory noise. Our particular approach has been to select specific parameters of auditory stimuli which are known to differentially influence the response of awake individuals and then to assess these parameters in terms of their contribution to sleep disruption. The strategy thus generally involves determining the correspondence between how an awake individual responds and how an asleep individual responds to various physical parameters of auditory energy. We believe that this particular approach has considerable to offer since not only does it address the nature of sleep itself, but also, and more importantly, questions whether similar psychophysical relations govern a human's response to auditory stimulation during sleep and wakefulness. If these psychophysical relations are similar, then predicting the sleep disturbing properties of a particular auditory noise will benefit from the mass of data already collected on awake individuals. On the other hand, if these psychophysical relations are not similar, then predicting the sleep disturbing properties of a particular auditory noise will be considerably more difficult. Obviously, in this latter case, a precise prediction of sleep disruption must await definition of the auditory psychophysical relations that function during sleep or, at least, the development of appropriate corrections which may be applied to the awake data.

Our particular methodology approach to this problem is conceptually quite simple and involves monitoring an individual's scalp electroencephalographic activity to define various subcategories of sleep and then, when appropriate, presenting a short auditory noise to the sleeping subject. The individual's response to this noise is quantified by recording the change in the frequency pattern of the ongoing electroencephalographic activity. Invariably this change represents a shift in the frequency pattern toward higher frequencies, that is cortical desynchronization or what is termed arousal. Our procedures exclusively utilize on-line realtime computer technology for both the analysis of the frequency pattern of the electroencephalogram during sleep, the scheduling of the auditory noise presentation, and the assessment of an individual's response.

In a related series of experiments under this empirical strategy, we have demonstrated that sleep is, in fact, quite easily disturbed by auditory stimulation even though the individual may not be overtly aroused or even consciously aware of the occurrence of the auditory noise. Moreover, and most importantly, the amount of sleep disruption is not predictable from the individual's response to similar auditory stimuli occurring when the individual is awake. In other words, the classic psychophysical functions which describe how the human responds to extrinsic auditory stimulation during wakefulness do not predict how an individual will be aroused during sleep. Finally, even though it is presently impossible to exactly predict the amount of arousal precipitated by particular auditory stimuli, our data suggest that even minimal sleep disruption can carry over and influence waking performance if the requirements of this performance place certain mnemonic and/or cognitive demands upon the subject.

The critical message of these researches, so far as understanding the sleep disturbing properties of auditory noise is concerned, is that how an individual responds when asleep is not predictable from how an individual responds when awake. This in turn suggests that sleep is not simply quantitatively different from wakefulness but may, in fact, be qualitatively distinct and as such governed by a distinct set of psychophysical relations. To appreciate the disruption of sleep by auditory noise, one must then appreciate the uniqueness of an individual's response during sleep. And, it would appear that it may be necessary to do so not only to allow a "good night's sleep" but also to insure a "good day's work".
AUTHOR INDEX
9 INTERNATIONAL CONGRESS ON ACOUSTICS
MADRID 4-9 VII. 1977

M21 D'ARRIGO, GIOVANNI
S52 TAMOMO, A.
K2 DATTA, S.K.
L18 DAVIES, I.E.
K3 DE BILLY, M.
L14 DE JONG, B.
K32 DEFAYRE, ANDRE
L11 DELFINO, MICHAEL
K16 DICKENS, F.I.
L12 DICKIE, J.M.
G7 DIESTEL, HORST B.
K72 DIEZ, L.
N23 DINAPOLI, FREDERICK
O35 DJURIC, S.
M8 DODZIUKI, MIECZSLAW
K7 DOUCET, J.
L11 DOUGONETTE, L.R.
L21 DORTH, G.
R17 DUSEK, KAREL
N30 DYBA, ROMAN

K61 EDWIGE, M.
K2 EHARA, SHIRO
M8 EL-SAREM, M.
N49 ELLEMAN, D.D.
M0-4 ELLIS, DAVID H.
M0-2 EMBLETON, TONY F.W.
K9 ENERY, J.
K39 ENDO, K.
K37 ENDON, M.
N13 ENGEL, Z.
K22 ENNIN, R.I.
L37 EPISTEIN, DAVID
N29 ERMILIN, K.K.
K6 ESTEARN, D.
R6 ESTEARN, D.

G16 FAHY, FRANCIS
K50 FARRAHAN, HAROLD
K14 FANG, GAUTIER
K30 FARE, ALAIN
K70 FILIPCZYNSKI, LESZEK
O9 FINK, M.
K24 FISCHER, MYRIAM
L11 FLEAM, L.
P12 FLETCHER, HARVEY
N47 FLOCKTON, STUART
SB4 FORD, H.D.
03 FORREY
K77 FRIED, C.W.
N15 FRIEDRICH, HANS
K77 FRIZZELL, LEON

G17 FUKUCHI, Y.
R16 GABOR, ANDRZEJ
K15 GABOR, ANDRZEJ
K67 GAETTE, CARLOTTON, L.
K53 GAETTE, CARLOTTON, L.
R5 GALAND, C.
R6 GALAND, C.
K67 GALLEGUET JUAREZ, J.A.
K53 GALLEGUET JUAREZ, J.A.
L29 GANNON, E.C.
K9 GASSIS, S.
L8 GASTARD, J.C.
L7 GAZANNES, CLAUDE
K24 GESLIND, J.V.
K74 GIANETTI, C.
K56 GOBY, F.
M1 GOLENISCHCHEK-KUTUZOV, VADIM
K76 GOLINNA, IRINA
N36 GONCHAROV, V.P.
H9-4 GOODWIN, J.D.
H1 DOTH, TOSHIYUKI
N25 GROCKE, G.S.
K32 GROVES, R.D.
K45 GREEN, ROBERT E.
K44 GREQUSS, P.H.
H9-1 GRIEFHAN, BARBARA
H9-5 GRIEFHAN, BARBARA
H9-2 GROESSE, H.J.
H9-4 GRIFFITH, D.B.
K10 GUDAT, H.
K56 GUEGAND, V.
G18 GUENOUN, P.
G21 GUILLOT, J.P.
K36 GUINOT, J.C.
M16 GUREWICH, V.L.
M15 GUREWICH, V.L.

016 HACKLINGER, MARK
07 HAN, H.I.
P15 HARRAT, HELENA
K3 HARRISON, ROBIN
M14 HARTMANN, BRUCE
N25 HASSAAR, J.C.
H2 HALGAR, H.J.
K18 HAUER, G.
L36 HAUSMANN, G.
K32 HAW, G.
012 HEDEGARD, PETER
N3 HEISS, ALFRED
W6-1 HINCHLiffe, R.C.
S52 HO, M.T.
N16 HOCHLER, GEHARD

Institut für Technische Akustik
der Rheinisch-Westfälischen
Techn. Hochschule
Aachen
<p>| N11 | PFIZENMATTER, EBERHARD |
| N23 | PFETZSCHEGER, J. |
| K29 | PHOTHIPIHITCHITR, M. |
| N29 | PHOTHIPIHITCHITR, M. |
| K22 | PIERS, J. E. |
| N62 | PIGNON, JEAN-PAUL |
| N5 | PISANI, RAFALE |
| K62 | PISARENKO, G. G. |
| K46 | PLOQUE, F. |
| N20 | POLIEG, RYSZARD |
| K30 | POMARE, BERNARD |
| O25 | PONS, J. N. |
| N38 | POPPLEWELL, N. |
| L25 | PORTER, ROBERT |
| K32 | POULIOUEN, J. |
| L39 | PURSHOUSE, M. |
| L28 | QUAI, R. H. |
| K7 | QUENTIN, GERARD |
| K3 | QUENTIN, GERARD |
| K20 | RAJRAGOFALAN, S. |
| O18 | RAJKAR, GHOR |
| K71 | RANZ GUERRA, CARLOS |
| L7 | RANZ GUERRA, CARLOS |
| B19 | RASMUSEN, KNUD |
| K71 | RECORDER, LOPEZ, MANUEL |
| O4 | REDEL, A. |
| K61 | REYMOND, M. C. |
| N15 | RICHTER, K. |
| O17 | RICOV, YASUHIRO |
| K53 | RIEPA, E. |
| R4 | RISO, Y. |
| K75 | ROBERTS, W. C. |
| K53 | RODRIGUEZ CORRAL, G. |
| K67 | RODRIGUEZ CORRAL, G. |
| N8-1 | ROUKSE, R. |
| N28 | ROMILLY, NICHOLAS |
| N15 | ROMMENBERGER, DIRK |
| P24 | ROSE, O. |
| N27 | ROSENHOUSE, G. |
| L25 | ROSS, DONALD |
| P24 | ROSSING, T. D. |
| O14 | ROTH, OLE |
| K18 | ROUSEAU, M. |
| M3 | RUDENKO, O. V. |
| N50 | RUDNICK, I. |
| N26 | SACHSE, WOLFGANG |
| P14 | SAKAI, AKIRA |
| N49 | SAFFREN, M. M. |
| K79 | SAINGA, R. |
| O4 | SAKAMOTO, K. |
| O26 | SALAVAR, T. M. |
| G1 | SALAVAR, T. M. |
| R12 | SANCHEZ GONZALEZ, JAVIER |
| K17 | SANDHU, J. S. |
| R14 | SANDRI, STEFANO |
| K34 | SANTOBONI, SILVIO |
| O5 | SAPIR, J. |
| N24 | SAWA, Y. |
| K65 | SATO, H. |
| K39 | SATO, TAKUSO |
| K25 | SATO, TAKUSO |
| K33 | SATO, TAKUSO |
| P21 | SAU-BAYER, R. |
| R14 | SCHLIDT, D. O. |
| K29 | SCHLIDER, H. H. |
| N17 | SCHMIDT, W. M. |
| L13 | SCHOENBECK, MICHAEL |
| P19 | SCHMIDT, ROBERT |
| K13 | SCUDIERI, F. |
| K16 | SCUDIERI, F. |
| O6 | SCURO, Y. |
| K11 | SERRANOS, CORINA |
| K10 | SERRANOS, CORINA |
| O39 | Sessler, Gerhard M. |
| K12 | SETTE, D. |
| K16 | SETTE, D. |
| K52 | SHIMA, K. |
| P13 | SHAMS, MAHMOOD |
| M1 | SHAMUKOWY, NA |
| K59 | SHAMUKOWY, NA |
| R8 | SHIGEOKA, MINORU |
| O1 | SHIMBO, M. |
| P14 | SHIMIZU, R. |
| L2 | SHISHKOV, EMELINA |
| K39 | SHUKOYLOV, VA. |
| N1 | SIMONET, M. |
| O27 | SKVOR, ZDENEK |
| S65 | SLADKAY, J. |
| N4 | SLADKAY, PETR |
| K15 | SIMOVIK, ANTONI |
| L38 | SMITH, P. M. |
| K24 | SOCINO, GIOVANNI |
| O6 | SORIN, C. |
| K69 | SPEAKE, J. H. |
| L25 | SPINDDEL, ROBERT |
| N42 | STANTON, T. K. |
| K17 | STEPHENS, RAYMOND |
| K28 | STEPHENS, RAYMOND |
| K29 | STEPHENS, RAYMOND |
| M22 | STETI, CARMEN |
| S53 | STRUKA, JIRI |
| O3 | STUDENBAUER, GERARD |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>K16</td>
<td>STURLA, E.</td>
<td>L30</td>
</tr>
<tr>
<td>K33</td>
<td>SUNADA, T.</td>
<td>R20</td>
</tr>
<tr>
<td>K41</td>
<td>SZILARD, JOHN</td>
<td></td>
</tr>
<tr>
<td>M23</td>
<td>SZYMA, STANISLAW</td>
<td></td>
</tr>
<tr>
<td>N7</td>
<td>TACHIBANA, HIDEKI</td>
<td></td>
</tr>
<tr>
<td>M19</td>
<td>TAKAGI, ENSHIRO</td>
<td></td>
</tr>
<tr>
<td>F8</td>
<td>TAKASUGI, TOSHIO</td>
<td></td>
</tr>
<tr>
<td>K47</td>
<td>TAKEUCHI, P.</td>
<td></td>
</tr>
<tr>
<td>N19</td>
<td>TAKEUCHI, R.</td>
<td></td>
</tr>
<tr>
<td>P14</td>
<td>TAMURA, J.</td>
<td></td>
</tr>
<tr>
<td>O14</td>
<td>TANAKA, YASUAKI</td>
<td></td>
</tr>
<tr>
<td>P2</td>
<td>TANNER, ROBERT</td>
<td></td>
</tr>
<tr>
<td>K64</td>
<td>TARADA, OLRICH</td>
<td></td>
</tr>
<tr>
<td>K57</td>
<td>TARABA, OLRICH</td>
<td></td>
</tr>
<tr>
<td>M21</td>
<td>TARAGLIA, PIERO</td>
<td></td>
</tr>
<tr>
<td>L12</td>
<td>TAYLOR, D. W.</td>
<td></td>
</tr>
<tr>
<td>L16</td>
<td>TEMPLE, PAUL</td>
<td></td>
</tr>
<tr>
<td>N32</td>
<td>THEOBALD, M. A.</td>
<td></td>
</tr>
<tr>
<td>W5-5</td>
<td>THIESEN, G. J.</td>
<td></td>
</tr>
<tr>
<td>R12</td>
<td>THOMAS, DAVID</td>
<td></td>
</tr>
<tr>
<td>W5-2</td>
<td>THOMASON, SYEN-I.</td>
<td></td>
</tr>
<tr>
<td>R18</td>
<td>THOMSON, CARSTEN</td>
<td></td>
</tr>
<tr>
<td>S52</td>
<td>THORN, P. O.</td>
<td></td>
</tr>
<tr>
<td>K69</td>
<td>TORKI, YASUO</td>
<td></td>
</tr>
<tr>
<td>SB6</td>
<td>TOSI, OSCAR</td>
<td></td>
</tr>
<tr>
<td>P8</td>
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