10th International Congress on

ACOUSTICS

Volume 3
CONTRIBUTED PAPERS
continued
FOREWORD

The 10th International Congress on Acoustics, held under the auspices of the International Commission on Acoustics (ICA), is jointly sponsored by the International Union of Pure and Applied Physics of UNESCO and the Australian Acoustical Society. The Congress theme is "Acoustics in the 1980s" and the series of invited and contributed papers present an opportunity to examine current activities and new developments in all branches of acoustics.

Volumes II and III of the Congress publications include over 600 abstracts of the contributed papers published as submitted by the authors. The ten invited lectures suggested by the ICA are published in full version in Volume I.

The full version of the papers presented at the Satellite Symposia on "Engineering for Noise Control" in Adelaide and "Basic Causes of Noise Deafness" in Perth are also published in separate volumes.

In accordance with the directions of the ICA, one page only was allotted to each abstract. During the Congress registered participants will be able to buy copies of the full version of any papers provided by the authors.

J.A. ROSE
Chairman
Congress Executive Committee
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ACOUSTICS
THE ACOUSTICS OF THE NEW "SURROUND" CONCERT-HALL OF UTRECHT, THE NETHERLANDS

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Architect Hertzberger's surround concert-hall, opened in January 1979, is praised by both musicians and music critics for its acoustics. The octagonal hall (1500 seats, width 36 m) has its platform on one side, and steep stands on all sides, providing excellent sight. The obvious disadvantage of this shape is the large proportion of listeners behind and beside the orchestra (more than in the Berlin Philharmonie). The acoustic consultants finally agreed to the overall shape on the condition that all their recommendations would be accepted; they were.

The main attention was directed towards the creation of a desired pattern of reflections, namely the early lateral ones. To this end we suggested to interrupt the stands by vertical planes at different heights (c.f. Cremer's vineyard steps). They are clearly visible. We advised changes in steepness, better sightlines, and we claimed a number of platform reflectors. We chose the average height such that a volume of 17020 m$^3$ (600,000 cu.ft.) resulted. The ceiling is diffusing, providing oblique reflections. The hall's materials are concrete and thick plywood.

The necessary steps were taken to obtain NR 20 for air conditioning and outside noise (mark the 9.5x9.5 m$^2$ rooflight; insulated against aircraft and rain noise).

A 1:20 model was built (chipboard, egg-crates as occupied seats) a.o. in order to check on reflection patterns and on the effects of the (6+1) reflectors with 0.65x0.65 m$^2$ facets for oblique reflections; 6 are 2.6x2.6 m$^2$, the central one is 5.6x5.6 m$^2$; they hang at 10 m and 9 m respectively above the stage.

A measurement of RT in the model filled with nitrogen gave a value of 2.3 s.

THE OUTCOME

Three test concerts with full audience enabled us to obtain pulse responses with many early reflections and values for $T_{60}$ as in table below; EDT values, computed from recorded signals, are also shown. Jordan's inversion-index is: 1.14 (500), 1.11 (1000), 1.09 (2000).

The SPL's throughout the hall differ only slightly (< 1.5 dB).

MORALE

Contrary to a widespread opinion the Utrecht hall proves beyond doubt that the acoustics of a surround concert-hall can be very satisfactory.

Table:

<table>
<thead>
<tr>
<th>Hz</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1000</th>
<th>2000</th>
<th>4000</th>
</tr>
</thead>
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<tr>
<td>$T_{60}$ (s)</td>
<td>2.15</td>
<td>2.30</td>
<td>2.45</td>
<td>2.50</td>
<td>2.30</td>
<td>2.00 (empty, with chairs)</td>
</tr>
<tr>
<td>$T_{60}$ (s)</td>
<td>2.00</td>
<td>1.95</td>
<td>1.90</td>
<td>1.90</td>
<td>1.90</td>
<td>1.55 (fully occupied)</td>
</tr>
<tr>
<td>EDT (s)</td>
<td>--</td>
<td>1.7</td>
<td>1.7</td>
<td>1.8</td>
<td>1.8</td>
<td>1.6 (fully occupied)</td>
</tr>
</tbody>
</table>

*) The author worked with Ir.L.G.Booij of TNO-Delft.
ACOUSTICS OF THE ELISABETH HALL IN HIRISHIMA

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Elisabeth Hall is a new auditorium of the Elisabeth Music University in Hiroshima, built for the classical music. The dimension of this auditorium is 34.4m×23.6m×19m and the audience capacity is 820 persons. On the stage, about one hundred musicians can play orchestra with 200 chorists. The plan and sections are shown in the figures below.

At the first stage of the designing, there was an anxiety that the ceiling might bring problems in acoustics. So the scale model (1/50) experiment was done to investigate the behaviour of the reflected sound from the ceiling. Through this experiment, it was found that if the ceiling were made of diffusive and absorptive materials, the acoustics at audience area would be in good condition.
Variable Acoustics of the IRCAM concert hall in Paris

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In the heart of Paris at a site traditionally known as Plateau Beaubourg, a center for contemporary art, the Centre Georges Pompidou, has been established. One department of it: IRCAM (Institute for Research and Coordination in Acoustics and Music), is housed in an separate completely underground new building. Within an area of three thousand square meters the IRCAM building houses studios of different sizes, an anechoic room, laboratories, offices, control rooms etc. and a room for musical performances the "Espace de Projection".

This "Espace de Projection", had to possess the utmost regarding spatial and acoustical flexibility and variability.

Spatial variability, is closely linked to acoustical variability. By changing the reverberation time, and keeping the spatial dimensions constant, one also changes the level of the reverberant sound, the critical radius etc.

The gross volume of the "Espace de Projection" amounts to 6800 m³. Inside this volume is placed an inner casing, that not only improves the sound-insulation to the surroundings, but, more importantly, it provides the means to vary the absorption and diffusion in the hall.

The walls and ceiling of this inner structure are build up out of elements that each consists of three rotatable prisms. The three sides of these prisms are successively absorbing, specular reflecting and diffusely reflecting. The inner mobile ceiling of the hall consists of three part, each of which can be raised or lowered independently. Between the ceiling parts there is a slit through which a roller curtain can be lowered over the full width of the hall, thus giving the possibility of bisecting or trisecting the hall.

The prisms of a particular element can be rotated independently of those of the other elements. All changes and movements can be manipulated from a central desk in the control room, eventually by computer. The maximum nett volume is roughly 24 x 15,5 x 12 m³ and can if needed be reduced to less than 500 m³. The reverberation time can be varied in the case of a maximum volume between 5 sec. and 1 sec. without audience; fully occupied, the maximum reverberation time is 3 sec. The frequency dependency is also adjustable.

The "Espace de Projection" has now already been in use for more than a year. Although it was of course not yet possible to explore all possibilities the general impression is very favorable.

It is found to be not difficult to adjust the hall to optimal acoustics, the hall is then rated very high as a concert hall. Varying the acoustics did in fact already become a means of musical expression.

The hall proves to be well suited for recording of music, even of large opera. Remarkable is then the extremely limited number of microphones needed.

Acoustical research in this hall has already yielded interesting results i.a. the relation between reverberation time and the number of absorbing panels is found to be represented significantly better by the Sabine formula than that by the Eyring formula.

In conclusion, to get satisfactory variable acoustics one has to take rather radical measures. The result proves than to be objectively well predictable and subjectively very rewarding.
THE TWO AUDITORIA OF AUSTRALIA'S NATIONAL CAPITAL - CANBERRA

THE CANBERRA THEATRE

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Introduction:

Like Washington, Brasilia and Islamabad the Australian National Capital Canberra was established on a new site remote from Sydney and Melbourne which were already large cities. Since the Federal Parliament first met there in 1927 the resident population has steadily increased from a few hundred to the present 230,000. This increase together with the presence of the diplomatic corps created a great demand for improved cultural facilities. To meet this need the Federal Government provided the Canberra Theatre in 1965. It has a stage and fly tower making adequate provision for large scale operatic and theatrical productions and a median value of reverberation time making it suitable for speech, for opera, and for many other forms of music.

Design objectives:

(1) Good intelligibility of speech in all 1000 seats without sound reinforcement.
(2) A uniform reverberation time of 1.5 seconds over the 200-4000 Hertz range of frequencies increasing up to 2 seconds below 100 Hertz.
(3) Reverberation time independent of audience numbers.
(4) Excellent projection of sound from the orchestra pit.
(5) Good sight lines for direct propagation of sound and no seat to be more than 100 feet (30.5 metres) from stage.
(6) High degree of diffusion in the reverberant sound field.
(7) Strong first reflections delayed not more than 50 ms.
(8) Retention of sufficiently strong lateral reflections.
(9) Exclusion of external noise and quiet air conditioning system.

The author was appointed by the National Capital Development Commission to advise on criteria, on shape, volume and all acoustic aspects of the design. The paper discusses the design philosophy paying particular attention to the desirability of making the reverberation time independent of audience numbers. Artists appearing in an unfamiliar theatre find it much easier to perform before a large audience if the acoustic response remains much the same as it had been during the rehearsal without an audience. As the theatre has been used for practically every type of performance comment ranging from very favourable to unfavourable were to be expected. The substance of the comments received over a period of 15 years should be of interest to designers faced with the technical difficulties of providing for programmes ranging from unaided speech to symphony orchestral and choral concerts.

Acknowledgements: architect A. Noc. Simpson of Yunken Freeman Pty. Ltd.; permission to publish this paper from the National Capital Development Commission.
Introduction:

The new home of the Canberra School of Music was provided by the Federal Government in 1974. In addition to a wide range of teaching and practice facilities it has an auditorium with a large platform and seating for 1500 people.

Unlike the Canberra Theatre it was intended for music only, the acoustic properties being particularly favourable for the symphony orchestra.

Design objectives:

1. A full audience reverberation time of 2 seconds over the 315-4000 Hertz range of frequencies increasing up to 5 seconds below 125 Hertz.
2. Reverberation time independent of audience numbers.
3. Excellent sightlines for direct propagation of sound and no seat to be more than 100 feet (30.5 metres) from the platform.
4. High degree of diffusion in the reverberant sound field.
5. Strong first reflections delayed not more than 50 ms.
6. Maintenance of sufficiently strong lateral reflections.

The author was appointed by the National Capital Development Commission to comment on the suitability of the site from the point of view of external noise and vibration, to advise on criteria, internal shape, volume and other acoustic aspects of design.

After outlining the basics of design: results of measurements both with and without audience are presented. The results should dispel the belief that sound absorbent floor coverings are not appropriate in auditoria intended for music. As relative humidity is often very low in Canberra humidity controlling equipment has been included in the airconditioning system. Observations indicate that the influence of R.H. on high frequency reverberation time can be very significant and should not be ignored when designing an auditorium to have relatively long reverberation time. Some consultants believe that taking return air through risers under raked seating will introduce base absorption preventing the desired rise in reverberation time at low frequencies. In this auditorium and in the Canberra Theatre there was no evidence to support that belief.

Acknowledgements: Architect Daryl Jackson or Daryl Jackson Architects Pty. Ltd.; Permission to publish this paper from the National Capital Development Commission.
SOUND DECAY IN ROOMS WITH SUSPENDED DIFFUSORS

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In reverberation chambers the necessary sound field diffusion is frequently achieved by suspended plates or other diffusing elements which scatter the sound energy more or less uniformly into all directions. The required number and size of the diffusers is determined empirically on the basis of experimental investigations [1,2]. However, quantitative understanding of the sound decay in rooms with non-uniform wall absorption and with suspended diffusers is still lacking.

In this paper the influence of scattering elements on the sound decay in rectangular rooms with totally absorbing floor is investigated theoretically. The critical parameter is $h/\lambda$ ($h$ height of the room over the absorbing floor, $\lambda = 1/nQ$ mean free path length of sound particles between successive scattering processes, $n$ density and $Q$ scattering cross section of diffusers). Two cases can be treated without difficulties (see figure):

a) $h \gg \lambda$. Then the influence of diffusers consists in removing continuously energy from the sound rays and in directing it into the absorbing area. The decay constant $2\delta$ increases lineary with $h/\lambda$ in this region.

b) $h <\lambda$. The sound propagation can be conceived as diffusion of sound particles. The decay constant $2\delta$ drops down monotonically with increasing $h/\lambda$.

According to the diagram, one can expect a maximum of $\delta$ reaching or even exceeding the Sabine value $\delta_{Sab}$ somewhere between both limiting curves, i.e. for $h \approx \lambda$. This statement can be confirmed by the results of Monte-Carlo simulations of sound decay in rectangular rooms with specularly reflecting walls and with scattering elements distributed according to Poisson's law.

THE USE OF REVERBERATION FACILITIES IN THE 1980s
Riverbank Acoustical Laboratories

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The normal uses of reverberation rooms are defined for absorption, transmission loss, sound power, and impact measurements. Current state-of-the-art techniques and procedures used in the United States are reviewed for these four procedures. The use of computer control and on-line data reduction can define experimental accuracy much more clearly than earlier methods.

Some new techniques for measurement of transient vehicle noise in reverberation rooms are described. Results are presented for heavy-duty truck measurements in a 1000 m² reverberation room at Riverbank Acoustical Laboratory.
Building work on the new laboratories in the School of Architecture, University of Auckland has recently been completed. The acoustics section of this laboratory includes a suite of measuring rooms which rank internationally. The University has designated the activity to be centred on this facility The Acoustics Institute. Supported by appointments from the Building Research Association of New Zealand and with the active support of the Auckland Industrial Development Division of the D.S.I.R., the Institute is conceived as an interdisciplinary centre for advanced studies in acoustics providing also a consultative service to industry, particularly the Building Industry, the professions, and the community. Consulting activities and contract research are seen as essential aspects of the Institute.

The facility consists of three reverberation chambers, an anechoic room and teaching, administrative and control spaces. The reverberation chambers are assymmetrical, shaped to minimise border or niche effects on transmission samples while maintaining volumes of about 200m³. The chambers are arranged ensuite to provide for power measurements, transmission and insertion loss measurement on building elements and ventilating system components. One of the reverberation chambers is designed as the "receiving room" for transmission measurements. It is isolated on a coiled steel spring system with a resonant frequency of 3.5Hz. A hydraulic ram provides the means of lowering a 10m² floor section to the chamber below for floor/ceiling measurements. The third chamber is designed for power measurements on large internal combustion engines with appropriate provision for cooling water and exhausts. A variable flow very-low-noise air supply is provided to feed this suite with controlled quiet air for the measurement of the sound power of ventilating system components.

Diffusion in these chambers is discussed in a companion paper.

The Anechoic room has a 5.2m cube working space inside a 1200mm deep wedge system with a cut-off frequency of 63Hz. It too is isolated on 5.5Hz coiled springs and is accessible by large machines. Development of a measuring system and the details of the wedge design are discussed elsewhere.
DIFFUSION OF SOUND FIELDS AND DIFFUSION REQUIREMENTS IN REVERBERATION CHAMBERS
SCHOOL OF ARCHITECTURE, UNIVERSITY OF AUCKLAND, NEW ZEALAND.

DODD

GEORGE

SCHOOL OF ARCHITECTURE, UNIVERSITY OF AUCKLAND, PRIVATE BAG, AUCKLAND, N.Z.

The Laboratories in the new School of Architecture building of Auckland University include a suite of chambers for acoustical measurements on materials and noise-producing machines and devices. The design of the reverberant chambers has included consideration of means for ensuring a "diffuse" sound field inside.

The implications of different definitions of diffusion and the necessity for diffusion in measuring chambers will be discussed.

The problem of finding a method for measuring the degree of diffusion will be explained and it will be shown that in the more recent standards recommendations the question is bypassed by concentrating on the required degree of measurement precision.

The methods that have been used traditionally for creating a diffuse field will be described and compared with the currently widely-favoured rotating vane diffuser which has been chosen for the Auckland Chambers.

Brief mention will be made of new developments, for example the recently published Quadratic Residue reflectors, and their possible application in reverberation chambers.
Depuis 20 ans, les moyens d'essais acoustiques ont pris un développement considérable, ceci en raison des exigences des industriels constructeurs en matériels mécaniques, électriques, électro-mécaniques et musicaux.

Demain, ces matériels seront fabriqués dans tous les pays du monde industriel, mais ils recevront obligatoirement un étiquetage international qui précisera le niveau de nuisance sonore produit par le matériel construit.

Les exigences des Pouvoirs Publics et des Instituts de Consommation nécessiteront des moyens d'essais nombreux et importants, et apporteront donc un bouleversement dans la conception de la chambre sourde, cathédrale de produits absorbants.

Les chambres sourdes se distinguent en trois familles par leurs qualités acoustiques, c'est-à-dire par la coupure de fréquence (cut off) et une absorption recherchée à 99 %.

Nous avons les chambres d'essais pour les mesures de 20 GHz et 40 Hz (antennes), celles de 50 à 16 000 Hz (mécanique, électromécanique), celles pour les mesures "marines", entre 6 000 et 100 000 Hz (détecteurs).

Après avoir participé à la réalisation de 50 chambres sourdes, j'ai constaté que les fréquences de coupure demandées étaient de plus en plus difficiles à atteindre avec les moyens absorbants actuels. Dans le cas d'un Centre d'Essais pour matériels de Travaux Publics, par exemple, il faudra, en 1980, tabler sur des critères de 50 à 30 Hz, soit des "dents" absorbantes de 2 à 3 mètres. Les budgets seront tels que les chambres sourdes seront des ouvrages d'art.

C'est cette évolution qui sera présentée dans ma communication, illustrée par 40 diapositives, qui montreront des exemples de choix de produits absorbants, l'emploi des coûts et leur technologie (formes et dimensions), et enfin, les techniques de pose pour améliorer l'acoustique dans des volumes parfaitement déterminés avant la construction.
SPARE YOUR NEIGHBOUR!
The beneficial effect of a 3 dB lower level

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van den Eijk           Jan
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At previous ICA-congresses I paid attention to the question which sound insulation will provide a reasonable protection against neighbour's noises. The purpose of the present paper is to show that often one may lessen one's neighbour's annoyance appreciably by lowering the sound level of one's radio etc. by an amount which is hardly detectable and which does not influence the quality of the reproduced sound adversely. This follows from /1/ but seems not to have been brought forward explicitly enough. The phenomenon mentioned is based on the fact that if the level in the transmitting room - and therefore also in the receiving room - is lowered by a few decibels only, this may have an appreciable effect on the total time during which the level in the receiving room surpasses a threshold value below which no annoyance develops. This threshold itself is not known exactly. Therefore the following can only show the principle and the order of magnitude of the effect.

From Fig. 1 in /1/ it follows that a level reduction of 3 dB (which will hardly be noticed) reduces the percentage of time when 20 phon is surpassed, from 30% to 20% or even from 10% to 5%. From /2/ we can derive that 3 dB more sound insulation, which is equivalent to a 3 dB lower level in the transmitting room, increases the percentage of those who are satisfied with the sound insulation from 15% to 27% or even from 27% to 50%.

Van Rooijen /5/ found that, with an airborne sound insulation up to the requirements, 70% of 400 respondents to an inquiry heard sound from their neighbours in their living room and 25% in their sleeping room. An increase in sound insulation of 4 dB - which leads to the same level in the receiving room as a reduction of 4 dB of the sound level in the transmitting room - reduced these percentages to 30% and 10% respectively.

Therefore: SPARE YOUR NEIGHBOUR, IT COSTS SO LITTLE!

I am aware, of course, that 3 dB less noise is not a solution for all noise problems. I only want to underline that in noise control problems it is not only the level of the intruding noise which is important, but also the percentage of time during which the intruding noise surpasses an "annoyance threshold". (In fact we know this also from research on the annoyance of aircraft noise.)

REFERENCE:

/5/ J.N.M. van Rooijen: Sound nuisance and sound insulation between one-family houses. (In Dutch.) Bouwcentrum Blauwdrukken, Rotterdam 1976.
AN EXAMINATION OF THE PROBLEM OF SCHOOL ACOUSTICS.

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INTRODUCTION:
A number of changes have occurred in the architectural solutions for school design problems in the light of recent trends in educational patterns and philosophy. This has resulted in school "classrooms" becoming more flexible and less likely to be divided by solid walls serving as acoustic barriers, but rather by "operable walls" or, in some cases, no barriers at all. Many acoustic problems have been caused or accentuated by these changes, and it is the purpose of this paper to note those problems and discuss some of the design and assessment methodologies that have been proposed for the open situation. The results given are based on a literature review and some initial observations and measurements (including a user survey) at a contemporary, and a traditional, school.

LITERATURE REVIEW:
The literature review, which included subjects such as the effects of noise on learning and peripheral tasks, and acoustical studies of both "open plan" offices and schools, indicated that while actual noise level, up to about 90 dBA, had no direct and immediate effect on learning; content, time duration, intermittency and even individual personality played a part in affecting pedagogic efficiency, albeit in mostly experimental situations. However, the available design and assessment methods tend to concentrate on background level and/or articulation index, which seems to limit their usefulness markedly.

OBSERVATIONS AND MEASUREMENTS:
The measured levels at the contemporary school were found to be above those considered satisfactory by any of the assessment methods (L 65 dBA), while those at the traditional school were reasonable. Subjectively, however, the contemporary school was "acceptable" (much more so than the traditional) - the basic problem in the newer school being intelligible speech intrusion from neighbouring classrooms, or intermittent noises. Neighbouring classroom noise did not seem to affect the teachers, and often did not affect communication (AI 0.8), but the students identified it as a major source of distraction.

CONCLUSION:
The problem is not just a simple matter of reducing the background noise level, as has been erroneously accepted in many of the design guidelines, nor is it likely that a design method based purely on communication and privacy is adequate, but rather a combination of these and other aspects, including source type (particularly speech), peaks, and actual tasks, needs to be considered. The role of each of these factors has not yet been quantified.
NOISE CONTROL MEASURES IN THE NEW SINGAPORE INTERNATIONAL AIRPORT

University of Singapore

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This paper describes some of the noise control measures incorporated in the passenger terminal building at the new Singapore International Airport at Changi. As the acoustics advisor to the Changi Airport Development Division of the Public Works Department, the overall consultant of this multi-billion project, the author reviews some of the special steps taken to incorporate some of the latest acoustics techniques in this airport.

This includes the establishment of norms for various factors such as permissible background noise levels to each of the rooms, offices, arrival and departure halls. Specifications called for include optimum sound absorption and reverberation times, airborne sound insulation for internal and external building partitions, structural borne sound insulation as well as a sophisticated sound reinforcement system.

Because of the stringent requirements following some of these specifications the factor of skill and workmanship of the contractors involved in the erection of the partitions had to be allowed for. As a result all suppliers and contractors were requested to send their building proposals for testing. This was done in the newly established Acoustics Laboratory of the University of Singapore at Kent Ridge. Some of the tests carried out based on International Standards ISO R140 and ASTM E90-75 on sound transmission loss and ISO R354 and ASTM C423-66 on sound absorption are evaluated.
Scale Model Study on Sound Propagation in Building Structure

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Propagation characteristics of solid-borne sound in building structure were studied by scale model experiment.

The 1:20-scale model made of p.v.c. has a simplified building structure as shown in Fig.1.

In the experiment, vibration acceleration levels (V.A.L.) in 1 octave band were measured at widely distributed points in the structure when a certain point was excited by a electric shaker, as shown in Fig.2.

From the results of the experiments, it was revealed that the relative V.A.L. at any points is determined by its distance (r) from the exciting points via the shortest route in the structure, regardless of the propagating direction. Then assuming that variation of V.A.L. (Lr) in the structure could be expressed as follows,

\[ Lr = P_o - N \log r - M r \]

\( P_o \) depends on the strength of excitation), the values of \( N \) and \( M \) were derived from the results of measurements by the least square method. The results are as follows.

1) The value of \( N \) is 20 regardless of frequency.

2) The value of \( M \) becomes larger as frequency becomes higher.

Fig.1 Elevation and plan of the scale model.

Fig.2 Plots of relative V.A.L. in each octave band against \( r \) with regression curve.
Vibration generated by Sydney's new Eastern Suburbs Railway (E.S.R.) was predicted back in 1972 to generate structure borne noise levels within the proposed Theatre Royal immediately over the E.S.R. of Noise Rating (NR) 66. The reduction of structure borne noise involved the isolation of vibration both under the new Theatre Royal building and at the railway tracks.

The rubber bridge-bearing pads on which are mounted the Theatre Royal columns are described. The results of their noise reduction are shown.

The design of the railway track bed incorporates a continuous rail support intended to reduce low frequency vibration generated by sleeper mountings. The track bed itself is mounted on ribbed neoprene pads to isolate vibration at high frequencies. Measurements of the noise reduction due to the track design are shown.

A discussion of the relative advantages of structural and source vibration isolation is included.

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ENVIRONMENTAL IMPACT OF BUILDING ENERGY CODES ON ARCHITECTURAL ACOUSTICS

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Two of the major national building energy codes/standards for new construction in the United States that have emerged during the last five years are:

1. ASHRAE (American Society of Heating, Refrigerating, and Air-Conditioning Engineers 90-75, and

2. BEPS (Building Energy Performance Standards) of the US Department of Energy.

Since the basic objective of these two standards is to reduce the overall energy consumption in new building designs through added insulation, efficient lighting, improved HVAC (Heating, Ventilating, and Air-Conditioning) systems and controls, and use of alternate/renewable energy sources, the acoustical characteristics, in turn, reflect improved interior noise environment emanating from lower window to wall area ratios, significantly reduced infiltration/exfiltration losses and use of properly sized fans, motors, pumps and other electrically-driven efficient machinery.

It can be fairly concluded that both the prescriptive and the performance energy standards for new buildings appear to help the architectural/environmental acoustics criteria during the design and development phases of construction.
METHOD FOR FIELD MEASUREMENT OF FLOOR IMPACT SOUND LEVEL USING HEAVY IMPACT SOURCE

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As a standard impact source, ISO tapping machine is widely used internationally, and it is suitable for evaluating the shock absorbing effect of floor finishes against light-hard impacts as is caused by high heeled shoes' walking or falling of knife etc.

However, even if the same evaluation value against tapping machine, a light and hard impact source, is obtained, floors having different thickness and mass show perfectly different, from each other, floor impact insulation performances against heavy and soft impacts as is caused by running or jumping children in multifamily dwellings.

It is impossible at all to evaluate appropriately, on the basis of measurement result only obtained with a single standard impact source, the floor system comprising various nonlinear elements from surface finishing materials to inner construction.

It was decided to adopt the falling impact of automobile tire as a new impact source of a sufficiently soft stiffness and a large mass, and prescriptions of JIS standard in conformity with this have been described by two methods, that is, specifications concerning matter and performance.

Except during blowing, heavy impact source shall be kept apart from the floor surface and blow only once the floor surface at one fall. When it is regarded as a single mass system, its equivalent mass shall be 7.3±0.4kg. The dynamic equivalent stiffness, determined from the total duration of impulsive force, seen from its surface in contact with the floor shall be (1.6±0.1)x10^5 N/m. The coefficient of restitution shall be 0.8±0.1. The speed at the moment of its collision against the floor shall be equivalent to that realized when the source's impact surface comes into contact with the floor after its free fall from a height of 0.9±0.1m above the floor.

The heavy impact source can be such that it does not always comply with the requirement above, concerning its mass, stiffness and height of falling, as far as the impulsive force-time characteristics obtained from its fall on a smooth and rigid surface with sufficient effective mass comes within the range shown in Fig. 1. A 5.20-10^-4kg tire approximately satisfies this specification at its pressure (1.52±0.1)x10^3 Pa, and corresponds to the actual impact time characteristics of jumping children.
Development of a digitalized measuring instrument for reverberation time

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This paper describes the developed results of a small-sized digitalized measuring instrument by which the reverberation time of auditoria and studios can be simply measured in the field.

The calculating process of the instrument is based on Schroeder’s "integrated impulse method"\(^1\), and the fundamental principle is the replacement of visual linear approximation of logarithmic decay curves by the ordinary interrupted band-noise method with the least squares method. A block diagram of the instrument is shown below. In the instrument, the reading out procedure of wave forms of tone burst signals with 1/3 octave band spectra in all frequency ranges from ROM is adopted. The sound pressure response to a tone burst in an enclosure under study is converted into digital form by an AD converter by first sampling time rate. The frequency is 80 kHz. The converted signals are squared and integrated in real time. Outputs of the integrator are detected by second sampling time rate (for example, five milliseconds), and are written in order into the RAM of a signal processing unit. After completion of writing, logarithmic calculation is done immediately, and reverberation time is obtained from the slope of the discrete decay values calculated by the least squares method. Measured results are displayed on a videotape or the like. By complete digitization and automatic measuring operations like this, accuracy of calculation is improved and measuring time is considerably shortened. In case of the second sampling time rate of five milliseconds, the computation processing time of the data is within two seconds at most.

This measuring instrument is able to measure not only reverberation time but also each room acoustic parameter such as the ratio of early to reverberant sound energy and the "center time" of pulse response.

1) M. R. Schroeder: JASA, 37, 409 (1965)

![Block diagram of the measuring instrument](image)
REVERBERATION TIME EVALUATION BY RUNNING DIGITAL INTEGRATION

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The decay rates of reverberation of interrupted noise bands can be determined using a digital integration method. For two adjacent time frames of the decay curve of equal length $\Delta t$ the squared time signal is digitally integrated. The result of the integration $A_t$ is proportional to the energy of the signal in the time frame. The reverberation time $T$ can be calculated by the values $A_t$ and $A_{t+1}$ assuming an exponential decay of sound energy.

$$T = 6 \cdot \ln 10 \cdot \Delta t / (\ln A_t - \ln A_{t+1})$$

By varying the length $\Delta t$ of the time frames and shifting their position a number of values of the reverberation time for one decay curve are achieved.

To determine the exact value of the reverberation time it is necessary to select a suitable integration time $\Delta t$. For a number of rooms with different qualities of absorption we determined the reverberation time with different frame length. Experiments have proved, that it is necessary to adapt the frame length to each decay curve. The variance of reverberation time being a function of the frame length is used to select a suitable frame length.

Normally the exponential decay of the sound energy is superimposed by a constant value caused by background noise. It can be shown, that this leads to deviations if one uses conventional methods with logarithmic converters. The deviations of the measured reverberation time from the exact one depend on the signal to noise ratio. The method described above allows the correction of the measured values by taking the background noise level into account. The complete measuring system can be implemented on a microcomputer.

DIRECTIONAL MEASUREMENT OF REVERBERATION

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Echograms have been used to study the amplitude-time characteristics of individual reflections, but no simple way of studying their directional properties has been available. Such a system would be useful because it would give more information than a single diffusion figure. Meyer (1954) applied a technique of measuring echograms with a parabolic microphone and constructing three dimensional models of the arriving sounds. These "hedgehog" displays were formed by sticking pins of different lengths into a ball. Jaffe (1977) devised a system which also provided directional displays via a parabolic microphone. It involved making echograms at 30° intervals and arranging them radially on a page.

A system has been devised by the author which measures the directional characteristics of reverberation in real time. It performs a measurement of sound power vs. direction for the arriving reverberation signal from an impulse. An orthogonal trio of velocity microphones and a pressure microphone are positioned in a coincident fashion at the desired measurement location. Such arrays are used in the British NRDC Ambisonic surround sound recording system (Cooper-Shiga 1972), and were the inspiration for the system discussed here. The room is excited by an impulse generator, in the author's experiments this is a spark source (Cabot et al 1977). The velocity microphone outputs are each multiplied by the pressure signal to obtain the sound power in each of the three directions. Two of the signals may be connected to the X and Y inputs of an oscilloscope for viewing the resulting power response in the plane containing the two velocity microphones. Mathematically we have

\[ X = f^2(t) \sin \theta \]
\[ Y = f^2(t) \cos \theta \]

The resulting display is an array of radial lines, where the length of the line represents the power of a reflection and the angle represents its arrival angle. By simple electronic processing, the angle may be extracted and plotted vs. time. With slightly more processing, the dependence of line length on level may be removed and the length instead made proportional to reflection arrival time. Using techniques from ultrasonic imaging systems: It is possible to tilt and rotate the plane of view or to provide a three dimensional perspective view of the arriving reflections. The author is currently investigating the application of circular statistics (Cabot 1977) to the data from this system. It is also hoped to incorporate the "starburst" display format into a computer modelling program for room impulse responses.

This paper describes some physical acoustic properties of the recently built Oduduwa hall of the University of Ife, Nigeria. The author has made some measurements of the natural acoustic properties of the hall. Subjective evaluations of the hall by some listeners were also made. From these studies it was found that the reverberation time for the empty hall at midfrequencies i.e. 500Hz to 1000Hz was 1.2 secs. which is rather too low for a multipurpose hall of volume 9000 m$^3$. Sound diffusion is fair. The upholstered seats which have no perforations at the bottom can cause "bubble" resonance thus making the hall unsuitable for recordings. Subjective opinions agree on the good intelligibility and audibility of sound in the hall. However, about 7% of those questioned believe that there are echoes in the auditorium. This will be investigated objectively later.
THE ACOUSTICS OF THE PARLIAMENTARY HALL OF GREECE

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INTRODUCTION

The parliamentary hall of Greece is the main hall of the neoclassical building of the Parliament, situated in the very centre of Athens.

DESCRIPTION

Built around 1840 by F. von Gartner, the building was used as the King's palace until 1928, when it was redesigned by A. Kriesis to house the Parliament.

Remaining untouched since that date the building has two main halls, the hall of the Parliament and the hall of the Congress.

Surrounded by auxiliary rooms the main hall has a volume of 5,468 m³ and 407 desks. The hall is almost cubic with a concave shaped back wall and is surrounded by four rectangular recesses situated in a height, housing auxiliary services, the press as well as the public.

The walls of the hall are covered mainly with rare marble and decorated or painted plaster. Its roof is made of opaque glass, providing a natural overhead lighting.

MEASUREMENTS

Reverberation time measurements made with the aid of a shotgun and an instrumentation recorder gave the results shown in the figure, curve 1 for the hall empty. Curve 2 represents an estimation of the reverberation time of the hall occupied under normal session conditions. Curve 3 gives the projected reverberation time curve for the hall occupied, after the introduction of corrective measures discussed below.

DISCUSSION

Although oversized for its capacity and without any absorptive treatment, besides the internal surfaces of the recesses, the hall's diffusion together with the action of the wooden constructions on the floor and the glass panelling on the roof, mainly in the lower frequency end, provide an exceptionally even reverberation time throughout the whole spectrum.

The severe limitations imposed by the managing authorities in order to preserve the originality of the hall led to the adoption of the following measures: a) introduction of 125 m² of mineral wool covered with white cloth to imitate plaster in remote parts of the hall, b) introduction of 175 m² of resonators in front of the wooden desks, tuned at 125 Hz and c) introduction of 7 plexiglass reflectors at a height of 7 m above the speaker's rostrum. The expected results are shown in curve 3 of the diagram.
In major developments, the term "multi-purpose" auditorium may simply mean that the room will be used for drama as well as concerts of classical music. This poses some problems, since a theatre designed for drama (and good speech intelligibility) may not provide an appropriate acoustic environment for playing or listening to classical music.

In smaller developments, typical of small towns or urban local-government districts, the "multi-purpose" auditorium has to cater for a very wide range of activities. In addition to drama and music, the room may also have to be used for public dancing, receptions, cabaret, flower shows, polling booth, etc., etc. Thus, not only is a variable acoustic environment desirable, but flexible seating layout and floor geometry is required.

The major conflict in such auditiona is the requirement for a flat main floor area for all uses other than drama and music. This then severely restricts the possibility of providing good sight/direct-sound lines for much of the audience, when the room is used for drama and music - unless a fairly elaborate system of changing the main floor levels, seating and access is installed. Such an installation may be much too costly compared to the financial resources available.

An alternative approach is to accept the fixed flat main floor but to minimise its usage for audience seating when drama and music are presented. This allows a generous balcony to be constructed, with excellent sight/direct-sound lines, the projection over the main floor area being much greater than would normally be recommended for good sound distribution underneath. The prime seating area is thus located in the balcony and seats under the rear of the balcony should be removed, or at least be made available at low cost.

An auditorium constructed according to this principle will be described. From the balcony, the existence of the main floor seating area is barely discerned, and from the stage, the front rows of the balcony seating are close enough to establish good rapport. The importance of the auditorium's management personnel understanding the concept of the design must be emphasised, although measurements in the completed auditorium showed that speech intelligibility was quite good under the rear of the balcony, but the sight/direct sound lines, as expected, were poor.
SOUND INSULATION TARGETS FOR BROADCASTING STUDIOS

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Architectural designs for Broadcasting Centres, which may include both TV and radio studios, control suites, various technical facilities and a wide range of ancillary accommodation, are invariably complicated by the problems of ensuring the optimum airborne sound insulation between any noise-critical zone and all other adjacent or remote sound sources throughout the building complex. All studios and their control rooms demand stringent control of background noise levels. The next decade will see the advent of new recording and broadcasting techniques which will set still higher standards for the control of the noise background in order that the enhanced dynamic range of these systems can be fully utilised. On the other hand, studios and monitoring rooms frequently contain powerful sound sources, particularly when amplified 'pop' music is performed. To ensure that the architectural design of a studio centre is cost-effective, the inter-room partition selections must avoid both the excessive capital costs of 'over-design' and the costly operational penalties arising from inadequate control of residual noise and external sound break-in.

To assist the planning of Studio Centres and similar buildings, a rationalised system of design targets for sound insulation is proposed. The inter-room airborne sound insulation is obtained from a Table listing the required sound level reduction from any potential 'source' room, (studio, control room, plant enclosure, building exterior, etc.) to any other sound-sensitive zone. The Table is based on the requirement to maintain all noise break-in to NR 15 (or less) in the 'receiving' room. Standard corrections are applied for areas requiring a different background noise target, or for 'source' rooms where amplified music will be performed. Using these corrected 'noise reduction targets', the appropriate specifications for sound-insulating walls and partitions can be made, and the final design performance checked after completion of the building work.

The Insulation targets are proposed as a series of 'stacking' insulation contours having average values ranging from 30 dB to 90 dB in 5 dB steps. The gradients of the contours have been selected to match both the known requirements for inter-studio insulation and the measured performances of standard sound-insulating structures. Examples of this selection technique will be illustrated by reference to typical case histories.
INTRODUCTION

The importance of the need to carefully select the internal boundary materials of a building fabric for acoustical as well, as for structural, economic and aesthetic reasons is well recognised by architects and engineers today. If the criteria to control the acoustical character of an enclosure is to:

a) Modify the Reverberation Time,
b) Reduce existing Sound Pressure Levels,
c) Dampen Standing Waves etc.,

then appropriate sound absorbent materials of specific areas at selected locations will most likely be employed. As an example, the use of absorbent curtains, most likely made of wool, is a very suitable, though technically not well-documented material. (1 to 3).

EXPERIMENT

Recommended methods of measuring the absorption coefficient of materials and building elements are to be found in International and National Standards. This paper describes equipment used in a simplified method of establishing from \( \alpha \) the diffuse absorption coefficient and NRC of thin absorbent materials, such as various types of curtains, placed some distance away from rigid surfaces (slides 1 to 6). Results thus obtained are then compared to the measured diffuse sound absorption coefficients of the same materials (slide 7).

Finally, the field performance of such thin absorbents are studied by measuring Reverberation Time and Sound Pressure Level modifications in enclosures by the introduction of these materials and comparing the results with theoretical predictions (slide 8).

CONCLUSIONS

It is suggested that the proposed simplified method of obtaining the NRC of thin absorbent materials e.g. absorbent curtains, will promote the wider use of these materials by architects, engineers and designers to control the acoustical character of architectural spaces.

REFERENCES

The following topics will be discussed
1. Subjective results of concert hall studies.
2. Implication for concert hall design.
3. New sound diffusing structures based on quadratic residues and primitive roots.
6. Reverberation time calculation using proper integral equations.
INTRODUCTION
From the literature [1] it is known that "spaciousness" or "spatial impression" (in German: "Räumlichkeit") is a significant subjective attribute of the sound field in a (concert) hall. In general, spaciousness is meant to express the degree in which the perceived sound space extends beyond the space occupied by the sound source, or in other words: it is a measure for the "broadening of the sound source". It is supposed that spaciousness is related to the influence of early lateral reflections. Further, evidence is available that a large degree of spaciousness is allied to low interaural correlation. Also other subjective notions are encountered in literature [1], that partially seem to have the same meaning, e.g. "spatial responsiveness". A high degree of spatial responsiveness implies the feeling of the listener to be "enveloped in the sound".

The aim of the present investigations is to explore the nature of the different notions, their interrelationships, and the psychophysical relations between the percepts and the physical parameters involved. In this initial stage of the investigations, one essential simplification has been introduced, viz. binaural presentation of the sound stimuli by headphones.

EXPERIMENTS
I. Subjects were presented with dichotic white noise stimuli of which the interaural correlation degree could be varied between 0 and 1 by a computer. Further, the sound pressure level and the bandwidth of the stimuli, equal at both ears, could be changed. Using a psychophysical scaling technique (magnitude estimation), subjects were instructed to estimate the width (broadening) of the sound image. The results can roughly be summarized as follows:
   a. An approximately linear relationship exists between the perceived width of the sound image and the inverse of the interaural correlation factor.
   b. The subjective width increases with increasing sound pressure level, and
   c. Low-frequency stimuli possess larger width than high-frequency stimuli.

II. Subjects were presented with a dichotic stimulus composed of the same white noise or anechoic (dry) orchestral music to both ears plus one repetition (reflection) after $\tau_1$ msec delay to one ear and a second repetition after $\tau_2$ msec delay to the contralateral ear. The addition of the two delayed repetitions is able to evoke a dramatic increase of spatial impression, in the sense that the sensation of having the sound compressed and localized in the middle of the head, is replaced by a sensation of being enveloped in the sound field. This sensation is accompanied by an out of head localization.

Implications of both experiments will be discussed in more detail, in the context of our knowledge of the functioning of the auditory system. One outcome of the present experiments is the need to relate spaciousness to the shape of the interaural cross-correlation function, rather than to its long-term average.

LITERATURE
To obtain appropriate lateral-to-frontal reflected energy ratios in concert halls we have proposed the use of large lateral suspended surfaces independent of the boundaries of the reverberant volume (1). The necessarily large size of these surfaces makes the diffusion of reflections from them desirable to avoid the effects of specularity, false localisation, colouration and a too sharply defined reflection boundary in the seating areas. The Quadratic Residue surfaces proposed by Schroeder (2) offer determinate scattering and we have explored the possibility of using these to diffuse the reflections from the principal lateral reflectors in the Wellington Town Hall. In the resulting design, provision of lighting and ventilation, construction details, values for surface absorption and aesthetics are reconciled with the theoretical requirements of the Q.R. surface. In this paper we discuss the choice of the diffuser bandwidth, and factors bearing on the selection of $\pi$ as the generating prime number. The stepwidth, and the optimum number of periods in a diffuser section, the unit step depth and the significance of the "Sin$\pi$" function $x$ in the effective scattering produced in our design have all been taken into account.

Preliminary experiments which compare specular reflection of music with diffuse reflections of various bandwidths have been carried out in support of this proposal and the results will be discussed.

REFERENCES

(1) Marshall, A.H. and Hyde, J.R., J. Sound Vib 197963 (2) 201, 211

(2) Schroeder, M.R., J.A.S.A. 1979 65 (4) 958, 963
One of the characteristics associated with good concert hall acoustics is a diffuse reverberant sound field. This is not however a condition we expect in simple designs as virtually all the absorption is concentrated on only one of the six principle surfaces, namely in the form of the audience on the floor. With a point source the reverberation will be dominated by a two-dimensional horizontal field above the absorbent surface. Indeed for diffuse conditions a vertical line source is to be preferred suggesting that members of each orchestral group of instruments would be better placed one above the other! To create a diffuse field in such unpromising conditions, diffusing elements are added to ceilings and walls, however few guidelines are available suggesting the necessary degree of such treatment. It is pertinent however to discuss the reasons for a diffuse field being desirable. An example has been reported (1) of a non-diffuse field being encountered in a hall, which proved subjectively acceptable. With the present state of the art such a situation is not to be recommended universally. A diffuse field is desirable for reverberation time calculations to predict accurately. It also provides a reasonable guarantee of satisfactory spatial properties of the reverberant field, as well as probably being a characteristic of the world’s greatest concert halls.

Many sophisticated methods have been proposed for measuring diffusion; though none appears to be completely satisfactory or widely accepted. In our experience a lack of diffusion affects the reverberation time before affecting the linearity of measured decays. In acoustic models it is easy to compare the measured reverberation time (R.T.) with predicted values. It is reasonable to assume that in a model without seating (i.e. with only the incidental absorption of the walls etc) that the situation is diffuse. The absorption of seating can be measured in a model reverberation chamber and the R.T. in the model with seating readily predicted. A difference in R.T. between predicted and measured values can be taken to indicate inadequate diffusion. Diffusing elements can be added to the model until equality exists. We have tested 3 acoustic models at scales of 1:8 or 1:50 in which inadequate diffusion was encountered. These experiences will be reported. A particularly fundamental example indicated that in a rectangular hall with a coffered ceiling adequate diffusion is not a trivial matter.

The conclusion of these experiences is that inadequate diffusion is probably a characteristic of many concert halls, and that acoustic models are the only means at present of guaranteeing suitably diffuse conditions. Acoustic models at scales of 1:50 (2) or larger are suitable.

REFERENCES
The extent and significance of frequency fluctuations in the initial part of decaying sound in an enclosure excited by a wide band noise or of decaying vibrations in a structure subjected to impulsive excitations have not been fully explored. It is well known that the linearity of the reverberant decay curve has long been the criterion for the quality of sound perception in theatres and auditoria. The shape of the decay curves has always intrigued acousticians. There appears to be nothing that can be called a 'normal' decay curve. Most decay curves do have an initial slope different from the slope over the entire decay. Some reverberant decay curves also possess a concave shape in the initial part for which there appears to be no satisfactory explanation in the literature. These factors depend not only on the geometry and sound absorption characteristics of the bounding surfaces but also on the nature of the sound source for e.g. a piano, a pistol shot etc.

The properties that can be accounted for from the frequency response fluctuations in the decaying state are (a) the diffusion of sound in the decaying state (b) the early decay time and its deviation from the reverberation time over the entire fall or decay and (c) the predominance of low or high frequencies in the decay of sound or vibration and its relationship to the shape of the initial part of the decay. Frequency irregularity in the first 50 or so milliseconds of the decaying sound has been used as an index for diffusion and also to correlate the subjective qualities with physical measurements such as early decay time, reverberation time etc.

The work was performed while the author was with the Physics and Engineering Laboratory (P.E.L.) of the New Zealand Department of Scientific and Industrial Research and is published with the permission of the Director, P.E.L. and the Director-General, Health Department.
Effects of arriving direction on perceptible threshold and disturbance of speech echo
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In this paper, the effects of arriving echo direction on the threshold of echo perception and on the echo disturbance are investigated with hearing tests, in order to contribute to more accurate estimation of the echo disturbance for practice. Hearing tests were performed by the method of minimal changes in an anechoic chamber. The direction of the primary sound is the front of the subject in the horizontal plane and the level is 75dB(A). Directions of the echo are every thirty degree in the right hand of the horizontal plane. Delay times of the echo are 10, 20, 40(these are only for the threshold), 80 and 160ms. The stimulus is a male speech(10sec). The minimal changes of the echo level are 0.5dB for the threshold and 1.0dB for the disturbance, respectively.

Fig. 1 shows the results at the delay time 80ms, as an example. The perceptible threshold depends on the arriving echo direction and there is a significant difference between the thresholds. On the other hand, the echo disturbance dose not show the same dependence as the perceptible threshold. Only when the echo arrives from the same direction(front) as the primary sound, the echo disturbance is more difficult to be caused than other directions, and there is no significant difference in the echo disturbance between arriving directions, except from the front. Also, the results at the other delay times show a similar tendency to Fig. 1.

Fig. 1. Echo perceptible threshold and echo disturbance of speech as a function of arriving direction(delay time:80ms).
Evaluation de la qualité des auditoriums par le "rhyme test" modifié.

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Une méthode subjective pour l'évaluation de la qualité de l'écoute dans les salles, récemment proposée par les AA., a été l'objet d'une communication au 9ème Congrès ICA (Madrid, 1977). Cette méthode se base sur la détermination de l'index d'intelligibilité à l'aide du "rhyme test", modifié pour être sensible à l'effet de la réverberation. A ce but, on a employé des mots quadrisyllabiques, précédés, ou non, d'un mot introductif. Cette procédure a l'avantage d'être d'un emploi facile, sans nécessité de faire appel à des auditeurs spécialisés. Il demande seulement d'avoir à disposition le matériel phonétique enregistré sur ruban magnétique et un système d'amplification pour la diffusion en salle, si celle-ci n'est déjà dotée d'un tel système. Le matériel phonétique employé par les AA. comprend une liste de base de 100 mots quadrisyllabiques (20 groupes de 5 mots) et 4 listes additionnelles, constituées par les mêmes mots précédés d'un mot introductif monosyllabique, ou respectivement de 2, 3 ou 4 syllabes.

Dans cette communication on réfère sur les résultats obtenus dans quelques salles de grandes dimensions par la méthode sommairement décrite. Un intérêt particulier présente, entre les salles considérées, la grande Salle d'audience de la Cité du Vatican (salle "Nervi"), qui est jugée acoustiquement très satisfaisante par tous les auditeurs bien qu'elle présente des caractéristiques objectives non conventionnelles: forme particulière et très grandes dimensions (6900 places assis, ou 14000 places débout). Les résultats des essais effectués dans cette salle sont résumés dans la Fig.1.

Bibliographie


In the present work, sound reduction results of different samples of lightweight partition walls, as measured in laboratory and in field conditions, are compared.

Laboratory measurements were carried out in an horizontal transmission chamber of 118 m$^3$, with structurally coupled rooms, that basically fullfills the requirements of the German standard DIN 52 210 for laboratories with flanking transmission. In order to evaluate the longitudinal reduction index ($R_{ff}$) of the chamber, a flexible, multilayer partition has been installed between both rooms. Thus, sound reduction was measured and from the average velocity levels of the emitting surfaces, the different transmission paths have been evaluated.

During field measurements, most emphasis was placed again in the determination of energy propagation through indirect paths. In both cases, mounting conditions were carefully controlled, in order to make results comparable.

A good correlation between both measurements and the calculated values has been found, except at low frequencies, probably due to the presence of a small number of modes in the test rooms and a poor coupling between these modes and those of the sample partition.

As it was expected, in most cases, sound reduction results as obtained in laboratory are a good estimate of the future acoustical performance of partitions when installed in buildings (in similar mounting conditions).

Anyway, further investigation is required in order to study the discrepancies at low frequencies (below 250 Hz), between measured and theoretical values, in comparison with results obtained in another laboratories with similar characteristics.
The Design of Facilities for an Acoustic Research Establishment

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The National Acoustic Laboratories (NAL) of the Australian Department of Health has, as one of its major functions, the investigation of the effects of noise on people. This means that research is required into physiological and psychological effects of noise, communication in noise, hearing protection devices and noise measurement and analysis. The Laboratories are to be relocated in a new building which will contain a complex of facilities designed for the conduct of such acoustic research. The facilities include anechoic chambers, reverberation rooms, plane wave tubes, high intensity noise rooms and quiet rooms.

The design of these facilities has required acoustical and architectural investigations to establish:

(i) the specific purposes for which the facilities will be used on completion, and in the future;

(ii) the optimum dimensions required to achieve these purposes;

(iii) the maximum permissible levels of background noise and vibration within the facilities;

(iv) the existing and potential ambient levels of noise and vibration at the site;

(v) the attenuation requirements to achieve the desired interior noise levels, and the methods of construction to attain the desired attenuation;

(vi) methods of vibration isolation;

(vii) materials and arrangements required to achieve the degree of sound absorption required;

(viii) methods of providing access for people, equipment and services into the facilities without compromising the acoustic performance;

(ix) optimum layouts to provide all the specified requirements within a relatively compact space.

In the paper, the factors which were considered during the investigations are summarised, and an overview of the final design is presented.

Verglichen werden: a) bewertetes Bauschalldämm-Maß Rʾw
b) A-bewertete Pegeldifferenz Außen-Innen (L_Aa - L_Ai) und die
c) A-bewertete Mittelungspegeldifferenz Außen-Innen (L_eqa - L_eqi)

a) Rʾw ist gleich dem Luftschallschutzmaß LSM (nach DIN 4109 Bl. 2) zuzüglich eines festen Wertes von 52 Rʾw = LSM + 52. Dieses Rʾw erfährt zwar durch die Sollkurve (nach DIN 4109) eine der A-Kurve (DIN 45 633) ähnliche Frequenzbewertung, die lediglich im tieferen Frequenzbereich (100 - 250 Hz) eine um 3 dB stärkere Bewertung bewirkt. Jedoch sind beim Rʾw die Einflüsse der Raumabsorption und der Prüffläche eliminiert worden und auch der Einfluß des Außengeräuschspektrums bleibt hierbei unberücksichtigt.


Bedingt durch die relativ dichte Mobilisierung der heutigen Wohnungen ist die L_eqi. Absorptionsfläche A in Wohn- und Schlafräumen bei tiefen Frequenzen (100 Hz) mit A = 19 m² recht hoch (doppelt so groß wie früher) und steigt nur relativ schwach mit der Frequenz an.

Unter Berücksichtigung der Fensterfläche S und der L_eqi. Absorptionsfläche A ergibt sich ein recht hoher Raumeneinfluß K = 10 log S/A. K beträgt gemittelt über 16 Terz- und alle 137 Räume K = -7,6 dB.


c) Die gemessene A-bewertete Mittelungspegeldifferenz L_eqa - L_eqi (gemessen an 10 Fenstern) ist um rd. 2 dB kleiner als die A-bewertete Pegeldifferenz L_Aa - L_Ai und entsprechend ca. 3,7 dB größer als Rʾw. Außerdem ist die Abhängigkeit von der Verkehrsdichte gering.
When buildings are exposed to excessive noise from ground transportation, it is frequently necessary to increase the sound attenuation of the building envelope - particularly that of the exposed facades. In some cases the additional costs are borne by a compensatory authority and it is obviously important that the most cost-effective methods are used.

However, it is well known that it is difficult to predict the sound attenuation that will be achieved in practice using the results of laboratory measurements of sound transmission loss. In many situations, particularly in existing buildings, flanking paths contribute significantly to overall sound transmission, but this is difficult to quantify at the design stage.

A frequently used solution is to double glaze and seal all windows, and to seal all other openings as far as possible, such as under-floor ventilators, eaves vents, etc. However, this necessitates the provision of alternative ventilation, usually mechanical ventilation or an air conditioning system; vent sealing may also lead to accelerated deterioration of building elements. It is therefore important to determine the contribution of these various sound transmission paths.

It has been possible to carry out some before-and-after studies in dwellings which have been provided with double glazing, but it has proved difficult to make a systematic study whilst intruding in private dwellings and it has not been possible to compare alternative systems.

A temporary experimental building has been designed specifically to determine the effect on sound transmission of changes to the facade. Two test rooms, representing a bedroom and a living room will have one facade exposed to a busy street; flanking transmission between the rooms and the control room will be minimised. Initially the exposed facade is of timber stud construction, and various doors and windows will be fitted. The next stage will represent brick veneer construction and again, different door and window elements will be fitted. The effect of under floor and eaves sealing will also be determined. Finally, the timber stud wall and the suspended timber floor will be removed and a second brick skin constructed; this, with a concrete floor will then represent one level of a multi-storey apartment building.

This project is supported by the Australian Research Grants Commission and the N.S.W. State Pollution Control Commission.
Sound Transmission Through Facades - Microphone Locations

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BURGESS

Marion Anne

University of N.S.W., P.O. Box 1, Kensington N.S.W. Australia, 2033.

While there are standard recommended methods for measuring the sound transmission through facades the methods used by researchers do not generally follow these procedures. The aim of this ongoing project is to study and evaluate existing methods for determining sound transmission through facades and to select an accurate, practical and reliable method for use with available external noise sources, such as road traffic, aircraft and general community noise.

The first part of the study has been the influence of microphone location both outside and inside. The location close to the facade would seem most suitable for a wide range of situations including measurements at higher stories. The nature of the surface of the facade makes the fixing of the microphone less than 20mm, as specified in Standards, somewhat difficult. Even this distance of 20mm is significantly large for the higher frequencies, and comparison of measurements close to and well away from the surface showed inconsistent differences above 3150 Hz. Below this frequency the differences were approximately 3 dB which is in keeping with ISO 140/partV and contrary to ASTM E336 which includes an allowance of 6 dB for pressure doubling at the surface.

The location of the microphone close to the facade on the receiving room side was also studied. Room effects could then be allowed for once the furnishings had been determined. Comparisons between measurements at this location and a space average within the receiving room are being studied.

A special laboratory constructed adjacent to a major road and in an industrial area will allow further comparative measurements to be made on a range of facade constructions.
In the last time there are realized in Czechoslovakia dwelling and civic buildings from the whole-wooden constructions on the base of wood. As the sound isolating properties are dependant on the surface density of the construction acoustically simple walls are replaced by the multiply constructions. It's also for that reason, that the constructional element (wooden) has surface density 2 time lower than that one made from steel and 10 times lower than the iron-concrete element. Then the acoustical simple walls on the base of wood would be in disadvantage to in comparison the silicate constructions.

The whole-wooden constructional systems and the dividing constructions on the base of wood come under drying processes (they loose 15 % humidity in the course of 1/2 year). This results in the creation of breaches and they negatively influence on the sound transmission loss of R (dB) walls. Experimental research shows that the difference between the labor. values and the practical results after one year's exploitation is the difference $\Delta I_L = 9.6$ dB and it's 4.8 times according to ČSN. The statistical results also show the measured values of different objects on the base of wood.

This knowledge of the sound isolating properties of the dividing constructions on the base of wood is necessary for the suggestion of the constructional connections and joints of walls. It is important to apply correctly the packing connections especially these in whole-wooden constructional systems, which can improve sound isolating properties up to $\Delta I_L = 10$. 
Calculation of the Resonance Frequencies of an Irregularly Shaped Enclosure with Rigid Walls.

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Using a perturbation approach developed by H. Feshback [Phys. Rev. 65 (1944), 304] it is possible to calculate the resonance frequencies and pressure distributions within enclosures that have non-uniformities in shape and/or absorption at the walls.

A rectangular perspex model was constructed with one end wall that could be sloped at different angles. This enabled the resonance frequencies and pressure distributions to be measured for a number of different enclosure shapes. The resonance frequencies for the perturbed enclosure, $\nu_N$, can be expressed in terms of the resonance frequencies for the rectangular enclosure $\nu_N$.

\[
\nu_N^2 = \nu_N^2 + \frac{c^2 A_{NN}}{4\pi^2} + \frac{c^4}{16\pi^4} \sum_M \frac{A_M^2}{N^2 - M^2} \left( \frac{\partial^2 \psi_N(r)}{\partial n_S} \right) \int_S \psi_M(r) \frac{\partial \psi_N(r)}{\partial n_S} \, ds
\]

(1)

where

\[
A_{NN} = \frac{1}{V_v A_M^2} \int_S \psi_M(r) \frac{\partial \psi_N(r)}{\partial n_S} \, ds
\]

the integration being over the perturbed surface $S$ and $\partial/\partial n_S$ is the derivative in a direction normal to $S$. The perturbation is such that the volume is within the unperturbed volume $V_v$.

The results from the computation of the resonance frequencies using eqn. (1) and the measured resonance frequencies are shown in Table 1.

<table>
<thead>
<tr>
<th>Enclosure #0 (rectangular)</th>
<th>Mode</th>
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<th>0,1,0</th>
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<th>0,1,2</th>
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<tbody>
<tr>
<td>Computed</td>
<td>724</td>
<td>911</td>
<td>1149</td>
<td>1164</td>
<td>1358</td>
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<td>1635</td>
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<td>1823</td>
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<tr>
<td>Measured</td>
<td>723</td>
<td>915</td>
<td>1153</td>
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<td>1363</td>
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<td>1635</td>
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<table>
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<th>0,1,0</th>
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<td>911</td>
<td>1148</td>
<td>1163</td>
<td>1341</td>
<td>1464</td>
<td>1468</td>
<td>1622</td>
<td>1724</td>
<td>1822</td>
<td>1846</td>
<td></td>
</tr>
<tr>
<td>Measured</td>
<td>723</td>
<td>916</td>
<td>1157</td>
<td>1164</td>
<td>1354</td>
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<td>1630</td>
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<table>
<thead>
<tr>
<th>Enclosure #2 (angle 14.9°)</th>
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<tr>
<td>Measured</td>
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Agreement between the exact theory and measurement for enclosure #0, and between the measured frequencies and those calculated using the perturbation method for enclosure #1 is good with high correlation coefficients. For a larger perturbation, as in enclosure #2, the correlation between the measured and calculated frequencies is still high (0.997). There was no evidence of a resonance at 1024 Hz for the (101) mode.

By a simple change in the shape of an enclosure one does not destroy the low-frequency resonances and hence make the sound field more diffuse. The sound field at low frequencies still has very distinct pressure maxima and minima. The perturbation method has been extended to an enclosure having more than one sloping wall.
REVERBERATION IN SEMI-DIFFUSE SOUND FIELDS

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In most cases of practical interest the room shapes are neither regular nor irregular enough. In general, wave acoustics will be used for small, regular shaped rooms, and geometrical acoustics for large, irregular shaped rooms. There are many rooms in the intermediate region such that the walls of a room are perpendicular to the floor, or both to the floor and ceiling, and none are parallel, and so on. The following formula will serve for such semi-diffuse sound fields.

\[
(p^2)_{\lambda} = \frac{4gcW_0}{B} \left( \frac{1}{A_p} \int \frac{dn_p}{d\omega} \right) \exp(-A_p c t/4 V) + \sum_{i} \frac{\pi}{4 A_{ui}} \left( \frac{dn_{ui}}{d\omega} \right) \exp(-A_{ui} c t/\pi V) + \sum_{j} \frac{1}{2 A_{sj}} \left( \frac{dn_{sj}}{d\omega} \right) \exp(-A_{sj} c t/2 V),
\]

where B denotes the normalization factor such that

\[
B = \left( \frac{dn_p}{d\omega} \right) + \sum_{i} \left( \frac{dn_{ui}}{d\omega} \right) + \sum_{j} \left( \frac{dn_{sj}}{d\omega} \right).
\]

A's are absorptions of walls and \((dn/d\omega)\)'s are average densities of normal modes which can be expressed by

\[
(dn_p/d\omega) \simeq \bar{V} \omega^2 / 2 \pi c^3, \quad (dn_{ui}/d\omega) \simeq S_i \omega / 2 \pi c, \quad (dn_{sj}/d\omega) \simeq L_j / \pi c,
\]

to a first approximation. For a resonable estimate of their magnitude: (a) use the actual volume of the room for \(V\), (b) use the area of a plane surrounded by walls perpendicular to it for \(S_i\), (c) use the distance between parallel walls for \(L_j\). When no pair of walls is everywhere parallel, the terms in the summation over \(j\) reduce to zero. Further, when the room shape is irregular enough, the terms in both summations reduce to zero, and the result coincides with Sabine's formula.

In a listening room, for example, where the floor (and/or the ceiling) is dead as compared with walls which are perpendicular to the floor, the reverberation time for the frequencies such that \(f \gg (S_z/LH^2) \times 10^3 (S_z: \text{the area of the floor, } L: \text{the peripheral length of } S_z, H: \text{the room height})\) is given by

\[
T = \frac{0.128 S_z}{\alpha_{rez} L_{rez} + \pi \beta S_z}, \quad \alpha_{rez} = -\ln(1 - \tilde{\alpha}_{rez}),
\]

where \(\beta\) is the air attenuation coefficient and \(\tilde{\alpha}_{rez}\) is the mean absorption coefficient of the walls.

Reference
Let us assume that we have a reverberant room with a point sound source at \( x_0 \), with volume velocity \( Q(x_0) \) and frequency \( f \), and a point receiver at \( x \). The response function between \( x_0 \) and \( x \) is defined as

\[
R(f, x, x_0) = \left| \frac{p(x)}{Q(x_0)} \right|^2
\]

where \( p(x) \) is the sound pressure at \( x \). Let us make \( NL \) measurements of \( R \) between \( N \) source positions \( x_i \) at least half a wavelength apart and \( L \) receiver positions \( x_j \) at least half a wavelength apart and at least half a wavelength and at least the reverberation distance from any of the source positions. Let us average these values to obtain

\[
T(f) = \frac{1}{NL} \sum_{i=1}^{L} \sum_{j=1}^{N} R(f, x_i, x_j).
\]

The relative frequency covariance of \( T(f) \) is defined to be

\[
\frac{\langle T(f) T(f + \Delta f) \rangle}{\langle T(f) \rangle^2} - 1
\]

where the brackets \( \langle \rangle \) denote a local average over frequency. It can be shown that this relative frequency covariance is equal to

\[
\frac{1}{NL} \left( 1 + \frac{10}{8N} \left( 1 + \frac{19}{8L} \right) \right) \rho(\Delta f)
\]

where \( M \) is the statistical modal overlap and \( \rho(\Delta f) \) is Schroeder's "frequency autocorrelation function". We have

\[
M = \frac{3 \ln 10}{T} \left( \frac{4\pi f^2 V}{c^3} + \frac{3\pi f V^{2/3}}{c^2} + \frac{3 V^{1/3}}{2c} \right)
\]

and

\[
\rho(\Delta f) = \frac{1}{1 + \left( \frac{2\pi T \Delta f}{6 \ln 10} \right)^2}
\]

where \( V \) is the volume of the room, \( T \) is the reverberation time of the room, and \( c \) is the speed of sound.

For the case \( N = L = 1 \), we find that the relative frequency covariance of the response function \( R \) is

\[
\left[ 1 + \frac{1}{M} \left( \frac{3}{2} \right)^6 \right] \rho(\Delta f).
\]

For large values of \( M \) this reduces to Schroeder's result

\[
\rho(\Delta f).
\]
SOUND REFLECTIONS OF A PLANE AND CURVED RIGID PANEL
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First reflections of the rigid boundaries in an auditorium have very important roles in the early reflections.

The impulse response of a plane rigid panel is given by,
\[ f(t) = \frac{1}{2\pi} \int_{S} \frac{e^{-j\omega \cdot \tau}}{\sqrt{R^2 + S^2}} \delta(t - \frac{S}{c}) \, dS \]

where \( \tau \) is the distance between a receiving point and the image of a point source, if the panel has specular reflection point on it. \( R \) and \( S \) are position vectors of a point on the boundary from the image and a receiving point respectively. \( dS \) is an element vector of the boundary.

On the other hand, that of a curved rigid panel is given by the superposition of the reflection of each rectangle portion on the panel, treating only the first reflection by the Kirchhoff's boundary condition (see Fig.1),
\[ f(t) = \frac{1}{2\pi} \int_{S} \frac{e^{-j\omega \cdot \tau}}{\sqrt{R^2 + S^2}} \delta(t - \frac{S}{c}) \, dS \]

where \( U(t) \) and \( R(t) \) are a slip and ramp functions.

\[ A_1 = \text{const.}/(\cos \theta + \cos \phi) \]
\[ A_2 = \cos \theta \cos \phi \] (1)

Calculated and measured are compared in Figs.2,3 and 4.

The impulse response here is defined as the inverse transform of the transfer function filtered by the curve shown in Fig.2(b).

ON FLUTTER ECHOES IN SPHERICAL DOMES

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Flutter echoes are known since olden times, but only some particular cases and models have been considered. Among these, undulatory models for two parallel walls (1) and "optical" models for domes (2) are most outstanding.

The discrete time character of flutter echoes and the numerical support offered by computers suggest the convenience of using ray models. On the other hand, a considerable number of enclosures possessing highly marked flutter echoes have been built recently, then indicating the interest in knowing flutter genesis more deeply and the need for a better diffusion of Acoustics among architects.

With this aims in mind we have developed a ray method in the line of Krook's method (3), specially adapted to detection and analysis of flutter echoes. It permits an easy and rapid test of the conditions for sound to flutter, namely; existence of closed paths run cyclically; predominance of the acoustical energy of such paths over residual energy and discrete time structure with audible cadences. The trial signal consists mainly in a ray ring corresponding to an elementary solid angle whose direction scans the whole space. Successive deformations of the ray ring during propagation offer a satisfactory pictorial view of the flutter echoes.

This model has been applied to enclosures with spherical domes analyzing the flutter formation for different dome proportions. The time structure has been confirmed experimentally in a real enclosure, recently built, and in its scale model (1:8).

The knowledge of the genesis and nature of flutter echoes suggests immediately a way for suppressing that unwanted acoustical phenomena. The efficiency of such procedure has been confirmed experimentally in the above mentioned model.

SOME APPROACH TO REVERBERANT SOUND FIELD USING THE CONCEPT OF ENTROPY

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INTRODUCTION

Some acoustic fields are very simple. The others are very complex. Are there any appropriate indices to express the complexity or randomness of the fields? Concept of entropy popular in thermodynamics and communication engineering is introduced to approach the problem. As an application of special interest the complexity of reverberant field is considered.

ENTROPY

In general a physical system has many degrees of freedom (i.e. components). The complexity of the situation of the system depends on the contribution of each component. Entropy is a well known measure of randomness (or complexity) of the situation of the system and is defined as follows,

$$H = - \sum_i p_i \log p_i$$  (1)

where $p_i$ is the probability of the i-th component and

$$\sum_i p_i = 1 \quad (p_i \geq 0).$$

As for an acoustic field we may consider $p_i$ is the ratio of acoustic energy of the i-th component to that of the total field.

APPLICATION TO REVERBERANT FIELD

Imagine the sound field in a room. The field is formulated in several ways. For simplicity we suppose the field is stationary and is composed of a direct sound and its multi-reflected ones. In this case the total acoustic energy of the field is expressed as,

$$E = E_0 + \gamma E_0 + \gamma^2 E_0 + \cdots + \gamma^i E_0 + \cdots = \frac{E_0}{1-\gamma}$$

where $E_0$: energy of a direct sound, $\gamma$: average reflection coefficient of sound at the wall. Then the contribution of i-th component to the total field is,

$$p_i = \gamma^i E_0 / E = \gamma^i (1-\gamma) \quad (i = 0, 1, 2, \cdots)$$

where $i=0$ corresponds to a direct sound. Inserting the above equation to eq. (1) and after some calculation the entropy of the reverberant field is simply expressed

$$H(\gamma) = - \left\{ \frac{\gamma}{1-\gamma} \log \gamma + \log (1-\gamma) \right\} \quad (0 \leq \gamma \leq 1).$$

It is easily shown that the entropy is monotone increasing with $\gamma$. It takes the smallest value $H(0)=0$ when $\gamma=0$ and attains the maximum one $H(1)=\infty$ when $\gamma=1$. The results mean the reverberant field in a room becomes more complex with increasing average reflection coefficient of sound at the walls.
CALCULATION OF REVERBERATION DECAY CURVES FROM IMPULSE RESPONSE OF A ROOM

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DERIVATION OF REVERBERATION DECAY CURVE

An output signal \( f_2(t) \) of a linear system is given by an impulse response \( h(t) \) and an input signal \( f_1(t) \) as

\[
    f_2(t) = \int_{-\infty}^{\infty} f_1(t-\tau) h(\tau) d\tau
\]  

(1)

If \( f_1(t) \) is given as

\[
    f_1(t) = \begin{cases} 
    1 & \text{if } \tau \leq 0 \\
    0 & \text{if } \tau > 0
    \end{cases}
\]

(2)

where \( \chi(t) \) is a stationary signal, then \( \langle f_2^2(t) \rangle \) (for \( t > 0 \)) becomes a reverberation decay response, where \( \langle \rangle \) means to take ensemble average. Now next relations can be derived.

\[
    \langle f_2^2(t) \rangle = \int_{-\infty}^{\infty} |G(f, t)|^2 \tilde{\chi}(f) df
\]

(3)

where

\[
    G(f, t) = \int_{-\infty}^{\infty} h(\tau) \exp(-2\pi j f \tau) d\tau
\]

(4)

and \( \tilde{\chi}(f) \) is a power spectrum of \( \chi(t) \).

MEASUREMENT OF IMPULSE RESPONSE

From input signal \( y_1(t) \) and output signal \( y_2(t) \) of a linear system, transfer function is given as \( H(f) = Y_2(f) / Y_1(f) \) where \( Y(f) \) is a Fourier transform of \( y(t) \). Then impulse response is obtained as an inverse Fourier transform of \( H(f) \).

REVERBERATION DECAY CURVE OF A ROOM

Waveform of sound from a source placed at one point in a room is thought as an input signal and sound wave at another point as an output signal. In practice, impulsive sound like pistol shot is emitted in a room and an output of a microphone placed near the source is thought as an input signal. Output of another microphone is thought as an output signal of general one input one output linear system, that is a room.

Then two signals are fed to a mini-computer and reverberation decay curves are calculated from equations (3) and (4) for arbitrarily chosen input signal \( \chi(t) \).

This means reverberation decay curves for very narrow band signal or particular sound such as human voice or musical note can be calculated from impulse response of a room.
DETERMINATION OF MATERIAL ABSORPTION BY HOLOGRAPHY.

In SABINE's laws, the absorption coefficients of material are related not only to the intrinsic absorbed energy $E_a$ of the media but also to the transmitted energy $E_t$.

To measure separately $E_a$ and $E_t$, we propose the following technic: In a KUNDT tube, the sample constitutes one of the external faces and the acoustical source is placed at the other face. Classical measurements of pression in the tube give the incident and reflected energies $E_i$ and $E_r$.

Normal velocities field is determined by holographic interferometry, in real time and stroboscopical technique; these methods are accurated and without any perturbation. By application of KIRCHHOFF-FRESNEL integral it is thus possible to calculate from those data the transmitted energy $E_t$.

The balance $E_a = E_i - E_r - E_t$ gives the real absorption coefficient of the material

$$\alpha_a = \frac{E_a}{E_i}$$

Tables of coefficients $\alpha_a$ for different materials for different frequencies will be presented in the paper. Optimization of the measurements and numerical treatment will be discussed in function of the different types of absorbers.


A NEW METHOD OF SOUND ABSORPTION COEFFICIENT MEASUREMENT IN ACOUSTIC TUBE

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INTRODUCTION

For the measurement of normal incident sound absorption coefficient of materials in acoustic tube, a new method that separates the incident wave and the reflected wave using a set of two microphones and two delay devices has been designed.

MEASURING METHOD

In Fig.-1, let \( M_1 \) and \( M_2 \) be the output of the two microphones which are set at the points of \( x=L+d \) and \( x=L \) from the surface of the tested material, and \( m_1 \) and \( m_2 \) be the delayed signal of \( M_1 \) and \( M_2 \) by \( t=d/c \) (\( c \) : sound velocity).

\[
M_1 = A \sin \omega \left( t + \frac{L+d}{c} \right) + A \sin \omega \left( t - \frac{L+d}{c} + \theta \right)
\]

\[
M_2 = A \sin \omega \left( t + \frac{L}{c} \right) + A \sin \omega \left( t - \frac{L}{c} + \theta \right)
\]

\[
m_1 = A \sin \omega \left( t + \frac{L+2d}{c} \right) + A \sin \omega \left( t - \frac{L+2d}{c} + \theta \right)
\]

\[
m_2 = A \sin \omega \left( t - \frac{L-d}{c} \right) + A \sin \omega \left( t - \frac{L-d}{c} + \theta \right)
\]

Then the incident wave and the reflected wave can be virtually separated by making \( u = M_1 - m_2 \) and \( v = M_2 - m_1 \).

\[
a = 2A \sin \left( \frac{\omega d}{c} \right) \cos \omega \left( t + \frac{L}{c} \right)
\]

\[
v = 2A \sin \left( \frac{\omega d}{c} \right) r \cos \omega \left( t - \frac{L+d}{c} + \theta \right)
\]

Consequently, the normal incident sound absorption coefficient and normal impedance can be known from \( u \) and \( v \).

Here, since the amplitudes of \( u \) and \( v \) depend on frequency, the distance \( d \) between the two microphones should be changed in several steps if the measurement is to be made over broad frequency range.

In this method, noises and pulses can be used for the sound sources as well as pure tones, and the length of the tube can be shortened as compared to the ordinary tube method.

As an example of the measured result, Fig.-2 shows the normal impedance of Glass Wool (5cm thick) measured by this method using pure tone signal.
Reverberation Room Tuning for Absorption Measurements

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The values of the absorption coefficients for a standard sample have been measured for nine different areas and configurations of fixed and stationary diffusing panels in the 250 m$^3$ reverberation room at NRC. Decay curves were recorded in eighteen one-third octave bands using a computer controlled real time analyzer. This system provides the means of studying spatial variations of reverberation time and decay linearity with good precision. A total of twenty-two microphone positions were used to sample the sound field in the room; 18 positioned in space throughout the room and 4 in corners. The position of the sample in the room was also varied. The data obtained are used to compare the room qualification procedures embodied in American Society for Testing and Materials method C423 with those in the draft ISO method R354. For our laboratory the two qualification procedures are not equivalent. The requirements in the draft method R354 were easier to satisfy, for example the requirements on decay curvature are much less stringent than those in C423. Measurement conditions acceptable to R354 did not satisfy the requirements of ASTM C423. Under such conditions the measured low frequency absorption coefficients were substantially less than those measured under conditions almost acceptable to C423. Rotating diffusers were found to be more effective at reducing spatial variations than fixed diffusers. The use of corner microphones did not eliminate the need for spatial averaging.
DEVELOPMENT OF POLYURETHANE FOAM ANECHOIC WEDGES.

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Anechoic wedges of a symmetrical shape have been reported extensively in the literature. Development work on anechoic wedges for the new anechoic room at The Acoustics Institute of the University of Auckland (reported in a companion paper) has resulted in a novel shape wedge which eliminates cutting to waste from expensive bulk material.

Preliminary work in selecting foams for extensive study of the wedge development involved measurement of acoustic flow resistance, and complex impedance in a Bruel & Kjær standing wave apparatus on many grades of polyurethane and polyester foams. It was found that air flow resistance is nonlinearly related to flow velocity through the range of typical acoustic particle velocities.

Composite wedges 1.2 metres long having a cut-off frequency below 63 Hz have been developed in both fire retardant and non-retardant grades of foam. Their novel shape shown in cross-section in Figure 1 continuously supports half the wedges out to their tips when laid in modules vertically as shown in Figure 2, and originally suggested by Professor A.H. Marshall. The arrows in the Figures show the slope of the wedge to the tip. The horizontally laid wedges are pinned with a short length of wire into the adjacent vertical wedges for support at the tips.

During development of the wedges systematic measurements of complex impedance and reflection coefficient were made on four full size wedges in a large standing wave tube under automatic control (reported in a companion paper). Tip length was successively increased then base length was reduced until an optimised absorber was achieved.

In the process of this development the sensitivity of reflection coefficient on the adhesive layer between composite materials at the base, introduction of wire spines to stiffen the wedges against tip sag, and mounting the wedges in a steel grid for support at 300mm from the chamber wall were studied. Each of these conditions resulted in an appreciable increase of reflection coefficient and cut-off frequency.

Figure 1. Figure 2.
FLAT WALLED GRADED DENSITY ANECHOIC LINING

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Materials used for the lining of anechoic rooms have varied over the years from layered sheet and blanket materials to shaped structures such as wedges, pyramids and blocks.

Design has relied on measurements of normal incidence absorption in an impedance tube, the more sensitive percentage pressure reflection being used as the measure. The criterion normally used to achieve suitable anechoic linings is normal absorption coefficient not less than 0.99 which corresponds to percentage pressure reflection not greater than 10.

As an aid to design the specific acoustic impedance of the lining which is more readily related to its physical properties can be used. In this case to meet the criterion above, the impedance measured at the face of the lining must lie within a limited range e.g. 0.818 > $\frac{R_s}{pc} < 1.020$ at $\frac{X_s}{pc} = 0$ and $-0.202 > \frac{X_s}{pc} < 0.202$ at $\frac{R_s}{pc} = 1.000$.

For a graded density lining, it is important that the front layer be highly porous, consisting mainly of air. The relatively recent availability of such durable materials has enabled the design of these linings to be considered.

It is important in such a system that sound reflections at the interfaces between the layers be reduced to a minimum. Hence, discontinuities such as air spaces between layers should be avoided.

Measurements of such a flat, multi-layered, graded density lining have achieved results comparable with well designed wedge linings with overall thicknesses slightly less than a quarter of a wavelength at the cut-off frequency e.g. 727 mm, 110 Hz.

These linings are more simply installed than the individually shaped systems with consequent reduction in costs.
SOUND INSULATION MEASUREMENT BY M-SEQUENCE CORRELATION METHOD

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INTRODUCTION

M-sequence correlation method is being used in various kinds of acoustic propagation measurements.
The authors have been studied the application of this method to architectural acoustic measurements, and have acertained that this technique is markedly useful for sound insulation measurements in the fields where enough S/N ratio can not be realized because of the influence of the background noises.

MEASURING METHOD

(1) Measurement of Pulse Response of a Room

In Fig.-1, the ensemble average of the squared pulse response of a room (echo time pattern) is obtained by measuring the cross correlation function $R_{ny2}(\tau)$ between m(t) (M-sequence signal) and the squared value of y(t) (output signal obtained when a noise source which is switched on and off by the M-sequence signal is applied to the room).

Fig.-2 shows an example of the pulse response of a room measured by this method.

(2) Measurement of S.P.L. Variation in a Room.

From the results of $R_{ny2}(\tau)$ measured at many points, steady state S.P.L. variation in the room can be derived by integrating them from $\tau = 0$ to infinity. cf. In the same way, reverberation decay process can be derived by integrating $R_{ny2}(\tau)$ from $\tau = t$ to infinity, according to the integrated impulse response method.

(3) Measurement of Sound Insulation between Two Rooms.

As shown in Fig.-3, the S.P.L. difference between two rooms (sound insulation) can be obtained by the same way as mentioned above.

As an advantage of the cross correlation method, the influence of the background noises can be reduced by taking long enough averaging time.

REFERENCE

N. Aoshima and J. Igarashi: The M-sequence signal applied to the correlation measurement of the wave propagation.
(6th I.C.A. in Tokyo, 1968)
FACTORS AFFECTING THE MEASUREMENT OF SOUND TRANSMISSION LOSS WHEN USING CORRELATION TECHNIQUES.

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The application of correlation techniques to in situ measurements of Sound Transmission Loss was investigated. An experiment was conducted using a test panel subjected to band limited (600 Hz to 6000 Hz) random white noise emanating from a loudspeaker. The acoustic field was monitored by two microphones placed on either side of the panel. The signals from the microphones were processed to give auto and cross correlograms, and then Fourier Transforms taken of segments of these correlograms were used to compute frequency response estimates and hence Sound Transmission Loss.

The Sound Transmission Loss estimates were found to be dependent on several major factors, these being:

- the identification of the required correlation peak from other contaminating signals
- the magnitude of contaminating signals
- the spatial response of the test environment
- the solid angle subtended by the panel at both the loudspeaker noise source and the receiving microphone
- the acoustic measurements in the near field of both the noise source and the test panel
- the frequency and amplitude resolution of the Sound Transmission Loss estimates obtained by using finite correlation time segments.

These factors are discussed and several solutions are postulated.
IDENTIFICATION OF FLANKING PATHS IN BUILDINGS

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INTRODUCTION
In recent years there has been an increased awareness of the need to regulate noise control in terms of acoustic isolation rather than the sound transmission loss of individual elements. If a partition whose attenuation characteristics seem satisfactory does not give adequate isolation there would also be a need to identify the faulty areas. The present method as described in ASTM E336 is cumbersome and time consuming.

EXPERIMENTAL METHODS
Various methods outlined in the literature have been reviewed and other methods have also been investigated.
(i) The most simple technique involves the use of a directional microphone to find out how the sound levels vary as the room is surveyed. This can best be done using a high frequency source. (see Fig. 1).
(ii) Other methods require the recognition of the path length difference between the direct signal from source to receiver and that via the flanking paths. A cross-correlation technique described in (1) & (3) was investigated. The signal from a microphone near the source is delayed with the signal from a microphone in the receiving room. A peak occurs when the time delay corresponds to the direct path and another when the delay corresponds to the flanking path. The time difference between the peaks identifies the path length difference.
(iii) A similar technique described by Raes (2) using an impulse source was looked at. Again two peaks appear due to the direct path and the flanking path.
(iv) A method using a sweep frequency generator as a source was developed. Because of the different path lengths there should be a constant frequency difference reaching the receiving microphone which sets up a beat. By knowing the sweep rate and sweep range the path length difference can be calculated.

CONCLUSIONS
The cross-correlation and the impulse techniques give accurate results but require quite sophisticated equipment. The directional microphone method is very simple and gives reliable results with a minimum of equipment.

Fig. 1 - Polar plot of directional microphone test with the microphone placed in the center of the receiving room. The large peak points to a poor construction joint and the smaller peak to the edge of a door.

REFERENCES
(1) K. W. Goff, J.A.S.A. Vol. 27, Number 2 (1955) pp. 236-246
(2) A. C. Raes, J.A.S.A. Vol. 27, Number 1 (1955) pp. 98-102
POWER-BASED MEASUREMENTS OF SOUND INSULATION

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INTRODUCTION
According to ISO Standard R 140, the transmission loss, \( R \), for a wall between two rooms is defined as:

\[
R = 10 \log \left( \frac{W_1}{W_2} \right) \quad (\text{dB})
\]

where \( W_1 \) is the sound power incident on the wall, and \( W_2 \) is the sound power radiated from the wall into the receiving room. Under the assumption of diffuse sound fields, the transmission loss can be evaluated from the following formula, given in the Standard:

\[
R = L_1 - L_2 + 10 \log \left( \frac{S}{A_2} \right) \quad (\text{dB})
\]

where \( A_2 \) is the absorption in the receiving room determined from Sabine’s formula.

Measurement according to eq (2) means in reality that the sound power, \( W_2 \), emitted into the receiving room, is measured using the reverberation time method. However, it has been shown in B & K Technical Review No. 3–1978 that such measurements of sound power give lower values compared to the free-field method at low frequencies. Therefore it may be expected that the measuring results of transmission loss using the classical method give too high values at low frequencies.

ALTERNATIVE MEASURING METHOD
To avoid the error explained above, a reference sound source may be used to determine the sound power \( W_2 \). The following formula can be derived for the transmission loss:

\[
R = L_1 + 10 \log \left( 1 + \frac{\lambda \cdot S_1}{8 \cdot V_1} \right) - L_2 - L_{WR} + L_{R2} - 6 + 10 \log \frac{S}{S_0}
\]

where \( L_{WR} \) is the (known) sound power level from the reference sound source, \( L_{R2} \) is the sound pressure level from the reference sound source placed in the receiving room, and \( S_0 \) is a reference area of 1 m².

**MEASURING RESULTS**
Measurement of transmission loss for a wall between two small rooms (\( V_1 = 31 \text{ m}^3 \), \( V_2 = 27.5 \text{ m}^3 \)) has been carried out using both methods, and the results are shown in the figure. Also the result of sound power measurement in the receiving room using the reverberation time method relative to measurement in a free field above a reflecting plane is shown.

It can be seen that the difference in sound power determination gives approximately the same difference in transmission loss. It should be noticed that the room in fact is too small for correct measurement of sound power at the lower frequencies. As a conclusion the alternative method described above gives better values, as it is based on the more accurate method of sound power determination using a reference sound source.
IMPROVEMENT THE PROCESSION THE EQUIVALENT ABSORPTION AREA FOR SOUND INSULATION MEASUREMENTS

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Total absorption of the receiving room is determined on the basis of reverberation time measurements $T$ when measuring sound insulation SRI. Measurements on site are carried out in very similar rooms, they are non-furnished and with concrete, plastered or gypsum walls, the floors being covered with parquet or vinyl-asbestos tiles. In all cases we can expect roughly the same average coefficient of absorption.

Results of site measurements $T$ of three Yugoslav laboratories were statistically treated /1/. Value $T/V^{\alpha}$ was also statistically treated. When calculating with with estimated total absorption, it was established that minimum error appears if instead of statistical average $T$ mathematical expectation $E$ of random variable $T/V^{\alpha}$ is taken, where $0.3 < \alpha < 0.7$.

The well known formula

$$ R = L_1 - L_2 + 10 \log \left( \frac{S}{A} \right) $$

yields to

$$ R = L_1 - L_2 + 10 \log \left( \frac{\text{6 SE}}{\text{0,163 S}_{\text{tot}}} \right) $$

for $\alpha = 1/3$. Here: $L_1$ and $L_2$ - average sound pressure levels in the source and receiving room, $S$ - surface of the dividing wall, $A$ - equivalent absorption area, $S_{\text{tot}}$ - total surface of the receiving room and $E$ - statistically established value /1/, /2/. Value $E$ is frequency dependent but constant for all the rooms. The formula (2) leads to an error of $+1.5$ dB at low frequencies and $+0.75$ dB at high with probability of 68.8% with respect to (1).

Smaller error may be obtained if a pre-defined absorption is placed into the receiving room.

2. B. Budisavljević, Statistics of reverberation time measured data XXIII Yugoslav conference ETAN, 11-15. juni 1979, MARIBOR, YU.
A Simplified Method for Laboratory and Field Sound Transmission Loss Testing of Building Partitions.

LIN, Hwee
FRICKE, Fergus

Introduction
Standard sound transmission loss tests, including the ASTM E597-77T\(^1\), do not appear to be simple or quick enough to be used for acceptance testing of buildings. An alternative, which is cheaper and easier to use, is proposed.

Description of Method
A sweep generator capable of sweeping through a frequency range of 100 Hz to 5000 Hz, is used to provide a source signal. The rapidly swept signal (sweep time ≈ 0.5s) is amplified and fed to a loudspeaker situated opposite the wall to be tested. One microphone only at approximately 1.5m high at the centre but in from the test wall is required in each room. Sound pressure levels in each room can be obtained simultaneously, if two measuring channels are available, or sequentially if only one sound level meter is available, and an overall sound transmission loss (e.g., dB(C)-dB(A)) or the transmission loss in a particular frequency band can be obtained.

Results
The results are summarised in Fig. 1. The regression lines in all cases do not pass through the origins as a 10 log S/A correction is not made for the simplified method. (The 10 log S/A correction is approximately 1 dB for the laboratory and 2 dB for field tests carried out).

Conclusion
The simplified method has been shown to correlate very well with standard method. It is also highly repeatable (Fig. 2). It is definitely a much simpler method to use while compared with other proposed methods\(^2\).

\[ y = 0.998x - 1.10 \quad r = 0.997 \]

\[ y = 0.997x - 1.89 \quad r = 0.97 \]

\[ y = 0.96x + 5.61 \quad r = 0.995 \]

\[ y = 0.96x + 3.10 \quad r = 0.97 \]

Fig. 1. Comparison of Sweep & Standard Test Results.

Eisenberg\(^2\) Gesele\(^2\) Sweep Method
2.4 2.4 1.29 (97c)
093(896-868)

Fig. 2. Repeatability Study

Acknowledgement
The financial support of Buchan Fell Committee is gratefully acknowledged.

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1. ASTM E597-77T, Tentative Recommended Practice for Determining a Single-number Rating of Airborne Sound Isolation in Multunit Building Specifi-
borne Sound Isolation in Buildings, A review undertaken for working
Group 51 on acoustics of International Council on Buildings (CIB).
Luftschalldämmung offener poriger Materialien

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Unter den Eichen 87

1.Absorbierende Materialien endlicher Dicke mit großer innerer Dämpfung besitzen eine Dämmung, die sich aus der Differenzbildung \( D_T \) der Schallpegel vor und hinter dem Material ergibt.

\[
D_T = \left( \frac{20 \cdot \omega \cdot c_0}{\sqrt{2}} \right) \cdot \sqrt{1 + \left( \frac{x}{\omega s_0} \right)^2 - 1} \cdot x_1 \quad \text{in dB} \quad (1)
\]

\( w = 2 \pi f, \quad c_0 = 340 \, \text{m/s}, \quad s_0 = 1,2 \, \text{kg/m}^3, \quad x_1 \quad \text{Materialdicke} \)

Das Schalldämm-Maß \( R_T \) kann unter Berücksichtigung 1. der Dämpfung im Material, 2. der Mehrfachreflexion an den Grenzflächen und 3. bei Bezug auf senkrechten Schalleinfall mit Hilfe der hyperbolischen Funktionen berechnet werden.

2. Messungen wurden nach einem Sondenverfahren und nach der Zweiwurmmethode durchgeführt. Auszugsweise sind Ergebnisse der Differenzmessung (D) und -berechnungen (\( D_T \)), der Schalldämm-Messungen (R) und -berechnungen (\( R_T \)) dargestellt.

Nach Gleichung 2 wird das theoretische Schalldämm-Maß \( R_T \) ermittelt, wie es im Diagramm dargestellt ist.

\[
R_T = 20 \log \left[ \cos h j k' d + \frac{j}{2} \left( \frac{\omega}{s_0} + \frac{Z_0}{\omega} \right) \sin h j k' d \right] \quad \text{in dB} \quad (2)
\]
A method is developed for analysing the influence of: (a) the statistical properties and interference pattern of the source, (b) the imperfect boundary conditions and (c) the internal damping on the acoustic wave motion in enclosures with transient and random excitations. Using a generalized normal-mode method, eigenfunction solutions are obtained for such wave motion where previously no analytical solutions have been found because of the boundary imperfections and non-uniform internal damping. Accurate estimates of the eigenfrequencies and their explicit dependence on the internal and boundary impedances are found by a generalized variational analysis, and a pole-diagram is obtained which shows geometrically in the complex frequency domain the location of the eigenfrequencies of the system with respect to the natural frequency of the excitation, where the natural frequencies of the source for any particular mode of the system are obtained by transforming the cross correlation of the source for any particular mode of the source into the complex frequency domain by means of the Fourier transform. Viewed in this fashion, the interactions between source and system can be classified and the influence of boundary and internal impedance factors readily evaluated. Examples and the wide applications of the method are presented.
Godot is a software system for computer-aided room acoustics design. It is able to generate an audible simulation of sources having arbitrary directional characteristics radiating within arbitrarily-shaped rooms, and can incorporate diffraction effects.

A polyhedral representation - a hierarchy of room, wall, edge, face, and vertex nodes - is used to model the room under consideration. Sound beams are defined by their axis directions and directivity factors, and may emanate from one or more sources. These beams are traced as they traverse the room, and their "hits" upon a receiver position are noted. The choice of which beams to trace can be made on the basis of their likelihood of striking the receiver within a given time interval. A perspective projection from the image of the source onto a "window" perpendicular to the beam axis is used to determine which edges of the room fall within a beam and hence can generate diffracted waves.

Each beam striking the receiver will be represented by a chain of source, reflection, and secondary source nodes, and from this chain the power transfer function, the time delay, the direction of hit, and the attenuation due to spherical divergence can be determined. The power transfer function is used to derive the autocorrelation coefficients associated with the path, and linear least-squares prediction is used to determine the coefficients of an autoregressive digital filter which will match the frequency response of the path. Each path from source to receiver may be simulated by inserting a time delay prior to the filter and a directional positioning module after it. The parallel channels of the system which represent paths to the receiver may then be driven by arbitrary digitized program material; after conversion to analog form, the signal may be used to give an audible simulation of the conditions being considered.
Initial design data for an enclosure such as an auditorium, a sound studio, etc. is the prescribed acoustic response of the enclosure, in the form of a numeric criterion. It has become accustomed to use the reverberation time criterion, specified as an ensemble of values of $T_R$ for each one of $n$ octave bands which compose the total relevant frequency range. The tolerance field of these values is usually also given.

The frequency characteristic of the total absorption is then deduced from the prescribed $T_R$ frequency characteristic, on the basis of the Eyring-Millington formula. The total absorption consists of: a) fixed basic absorption, and b) additional absorption at the designer's option. The additional absorption should in each given octave band supplement the basic absorption up to the value that will satisfy the initial criterion within the limits of the tolerance field. It is therefore necessary to use optimization procedure to compute the additional absorption all along $n$ octave bands. An ensemble of computer programmes in Basic has been made for that purpose.

The optimization procedure is performed in two phases:

1. The characteristic of the required total absorption, defined upon the discrete variable $x_n = n \cdot \log (f_0)$, $n=0...N$, is approximated by the polynomial $P(x)$ of the $N$-th order, which is a continuous frequency function.

2. The calculation of variation evaluates the optimum of the $x$-solved integral of the function $\theta(S_1...S_{k-1}, x, Kr)\text{,}$ the form of the function depending on an optimization criterion. The symbols stand for:
   
   $f_0 = \text{nominal i.e. center frequency of the lowest octave band,}$  
   $S_1...S_{k-1} = \text{single surfaces of additional absorbers,}$  
   $x = k \cdot \log f = \text{continuous variable}$  
   $Kr = \text{optimization criterion}$

As the result of the above computation the values of $S_1...S_{k-1}$ are obtained which fit optimally the prescribed frequency characteristics of the total absorption and reverberation time, respectively.
Towards an automated computer managed acoustic information and learning system

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Research into acoustic design information systems and design education was initiated in 1968 by Powell, Harman and Jones because it was considered that although the acoustic technology was sufficiently advanced to make possible acceptable design, much social survey research of the time had shown that noise was one of the most unsatisfactory aspects of the urban environment. To some extent the problem continues to exist because technological advances have barely kept pace with the improved standards we have increasingly come to expect of our environment. In order to improve standards I believe it is much more important for researchers to learn how to communicate existing acoustic knowledge to designers in a more efficient manner and also how to develop techniques and understanding that will enable them to apply this knowledge simply and quickly in their normal design work.

With this in mind an acoustic design guide was compiled for use by designers in the preliminary stages of the design of offices and domestic dwellings. In order to evaluate and develop the guide subjects carried out a series of design tasks both with and without the guide. Behavioural observations were made during the tasks and the initial guide modified accordingly. The outputs from these tasks showed the use of the guide increased the number of technical decisions made by the designer from about 20-80% in a given time depending on the designer's previous training. There was also evidence that the guide also improved the quality of their acoustic design of offices.

The guide at this stage was like a manual programmed learning text and it was clear that a sophisticated design method was not possible when the designers had to carry out their own computations. Therefore work was initiated with a view to making an efficient and interactive translation of the relevant parts of the guide in terms of computer software and micro-computer hardware. Using the "number crunching" capacity of the computer and its logic, the aim was to modify the guide so that with the same ease of understanding, the designer could produce acoustical solutions of high technical fidelity. A computer system, linked to a microfiche information retrieval device, seems to provide the most effective means of sorting and presenting design information as it allows a ready means of obtaining essential fast feedback on design decisions; a feedback which seems to facilitate learning, decision making and therefore the design process in general.

Such a system has now been developed at Portsmouth School of Architecture and is proving to be an invaluable aid in the acoustic education of young architect designer and an adequate mechanism for his retrieving relevant design information.
F. BIOACOUSTICS
ACOUSTIC CHARACTERISTICS OF SLOW AND FAST MUSCLES

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The determination of acoustic characteristics of muscle tissues is considered as an important line of investigation as it enables (1) the computation of the elastic constants, (2) the interpretation of ultrasonic images, and (3) the assessment of the degree of anisotropy.

The acoustical characteristics namely the velocity of propagation of longitudinal and shear waves together with the absorption of longitudinal waves along the direction of the muscle fibre and in a direction perpendicular to it, are determined in order to assess the degree of anisotropy, in slow (heart) and fast (leg) muscles of three animals (ox, sheep and goat), using the pulse transmission technique. From these measurements it is noticed that the velocity of propagation is more and the absorption is less along the direction of the fibre, while it is vice-versa in a direction perpendicular to it.

Further, it is interesting to note that the velocity in a fast muscle is relatively higher than in the slow muscle tissues and the absorption in fast muscle is relatively less than that in slow muscles.
RESTRICTION OF MECHANICAL ACTIVITY PRODUCED BY HIGH SOUND PRESSURES ON VARIOUS MUSCLE TYPES. EXPERIMENTS WITH UNDERWATER SOUND.

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Gastro-intestinal disorders often occur in man after an exposure to sound pressure levels of more than 100 dB in the range of lower frequencies. This fact has induced us to investigate the influence of sonic load on the contractile function of isolated intestines of laboratory animals. Segments of the ileum obtained from guinea pigs or rabbits were positioned in a 500 ml organ bath, the bottom of which was the vibration membrane of an underwater loudspeaker. Two mechano-electrical transducers were connected to the segment for simultaneous measurements of both intraluminal pressure and longitudinal tension development at 32.5°C. The maximum sound pressures, measured with a quartz pressure transducer were found to be about 30 mbar at frequencies of 70-130 Hz. During an exposure to sound load lasting 1-30 min the spontaneous contraction of ileum segments ceased or was reduced to about 30-40 % of the normal. When the loudspeaker was switched off contractility immediately or gradually returned to control levels. This depression of contractile activity produced by sound load was also demonstrated in electrically driven ileum segments. Other smooth muscle preparations (seminal vesicle, portal vein, and aortic strips) showed a similar relaxation. In further experiments the effects of sound load on the sustained contractile state of the guinea pig ileum was investigated. A potassium contracture was evoked by raising the K concentration in the bathing fluid from 4.7 to 43 mmol/l. At maximum state of contracture a sound load of 120 Hz caused prompt reduction of active tension and the intestine remained at a relaxed level during sonic exposure. On release of sound load the intestinal segment recovered its contractility in two phases: a first rapid and a second slow one. Striated muscle preparations (frog rectus abdominis) showed a similar response to sound load during K contracture. On the basis of our previous findings it appeared that the effects of sound pressure on muscle might be mimicked by direct vibration. Therefore, we have made attempts to investigate the excitation-contraction coupling of muscles exposed to vibration. The action potential of guinea pig papillary muscles was recorded by conventional microelectrode technique. The tendinous side of the muscle was connected to an isometric tension transducer and its base attached to a vibrator which produced oscillating movements of 0.2 mm at 100 Hz. The isometric tension development by electric stimuli (0.1 Hz) was reduced by about 60% during vibration, while resting potential, overshoot, and duration of the action potential remained unchanged. Thus the question arose whether the observed restriction of contractile force is caused by an alteration in the energy production of the cell or whether the energy utilization is reduced by uncoupling of excitation and the contractile system. For this purpose papillary muscles or ileum segments were frozen in liquid nitrogen, after a 15 min period of vibration or sound load, and their content of high-energy phosphates was analyzed. The muscles relaxed by sound or vibration did not show any change in ATP or creatine phosphate content compared with normally contracting muscles. These findings suggest that sound-induced vibration of muscles can disturb the sliding process of the myosin and actin filaments while the splitting of ATP at the cross-bridges proceeds without interruption.
The influence of acoustically induced shear stresses on biological systems.
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The investigation of the interaction between ultrasonic waves and biological tissue has been intensified in recent years as ultrasound has become increasingly important in medical diagnosis and therapy.

Although diagnostic ultrasound is generally considered to present no significant hazard to biological tissues, experimental results indicate that ultrasonic intensities as low as 100 mW/cm² (spatial-average, time-average) can give rise to uncontrollable changes in some biological systems which may include destruction.

The aim of this paper was to obtain information which would be useful in the assessment or prediction of conditions under which bio-effects occur and thus establish safety limits for ultrasonic exposure. This paper presents theoretical and experimental results based on a model, which assumes the existence of gaseous microbubbles in tissue, aimed at illuminating mechanisms of the tissue: ultrasonic interaction for the establishment of safety limits.

The inhomogeneities represented by the bubbles may, when exposed to ultrasonic irradiation, result in effects such as microstreaming and locally induced shear and normal stresses which may cause cell dysfunction through membrane injury. The theory developed leads to a non-linear differential equation comparable to that of Lauterborn based on the Noltingk-Neppiras theory. Bubble resonant frequencies are calculated for a typical tissue environment, and shear stresses evaluated from the solution of the equation for the pulsation amplitude.

Limited experimental results of the intensity thresholds at which effects occurred were compared with theoretical predictions, for both pulsed and continuous irradiation. The partial confirmation of the use of the model for predicting safety limits for ultrasonic exposures will be discussed with particular reference to the implications for routine diagnostic and therapeutic procedures.
Estimation of the Acoustic Characteristics of Continuous Medium
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Ultrasonic pulse-echo systems have been used widely in diagnostic medicine; however, they cannot give quantitative information concerning the acoustic property of tissue. Since the determination of the acoustic property will be sufficient to discriminate among the different tissue types and their pathological states, currently there is considerable interest in ultrasonic tissue characterization. In this paper, the basic theory which represents a relation between the waveform of reflected echo signals and the acoustic characteristics of the medium was introduced first, and then the approximate method which can estimate both of the impedance and the attenuation coefficient from the waveform of the echo signals was introduced.

From the echo signals reflected by the medium, we get the normalized frequency response function \( H(\omega) \) of the medium as follows:

\[
H(\omega) = \int_0^\infty \exp(-Q(t,\omega))g(t)\exp(-j\omega t) dt
\]

where

\[
g(t) = \frac{1}{2} \frac{d}{dt} \ln Z(t),
Q(t,\omega) = \int_0^t \alpha(x,\omega) dx
\]

and \( Z(t) \) and \( \alpha(t,\omega) \) denote acoustic impedance and attenuation coefficient as a function of acoustic travel time \( t \) and/or frequency \( \omega \).

It is assumed that \( Q(t,\omega) = Q(\omega) |_t \), that is, attenuation is proportional to frequency and that the total round-trip acoustic travel time \( T \) and the total attenuation \( A = -Q(T) \) of the medium are known beforehand. Taking the echo signals from both sides of medium, that is, from the side 1 and side 2 as shown in Figure 1 and denoting the Fourier transforms of the signals obtained from the side 1 and 2 by \( H_1(\omega) \) and \( H_2(\omega) \) respectively, we get the first approximated relations as follows.

\[
\int_0^\infty g(\tau) \exp(-j\omega \tau) d\tau = F(\omega) / [1 + \exp(-A/|\omega|)]
\]

\[
\int_0^\infty Q(\tau) g(\tau) \exp(-j\omega \tau) d\tau =
[(1 - \exp(-A/|\omega|)) F(\omega) / (1 + \exp(-A/|\omega|) - G(\omega))] / [(1 + \exp(-A/|\omega|)) |\omega|]
\]

where \( F(\omega) = H_1(\omega) [\exp(j\omega T) H_2(\omega)]^* \), \( G(\omega) = H_1(\omega) + [\exp(j\omega T) H_2(\omega)]^* \).

As the right side quantities of Eqs. (1) and (2) are known, \( g(t) \) and \( Q(t)g(t) \) can be obtained by inverse Fourier transforms of these equations. If \( g(t) \) and \( Q(t) \) are known, we can estimate the profiles of impedance and attenuation coefficients, that is, \( Z(t) \) and \( \alpha(t) \).

As resonance-type ultrasound transducers which are commonly used in practical pulse-echo systems have band-limited frequency characteristics, actually we analyzed the echo signals by taking it account.

![Fig.1 Estimation method](image)
APPARENT AND REAL THRESHOLDS FOR BIOLOGICAL EFFECTS OF ULTRASOUND

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Reported in literature thresholds for biological effects of ultrasound vary from less than 0.1 w/cm² to greater than 100 w/cm². It is difficult to imagine that biological objects having so much in common in their structural and functional properties may differ as much in sensitivity to the external physical factor. Primary physico-chemical phenomena in sonicated biological objects must have similar nature in different objects. So, thresholds given in literature are only apparent ones and it is doubtful that they actually represent values of real thresholds directly related to physico-chemical shifts produced by ultrasound in a biological system.

Apparent thresholds for biological effects of ultrasound correspond to intensities where local homeostatic systems of the object become unable to compensate the ultrasonically produced disturbances. Apparent thresholds depend on both the parameters of ultrasonic field and the activity of regulatory and reparatory systems in the object. So the real thresholds for ultrasonically induced bioeffects can be much lower than those reported in literature and it can be revealed in three different ways. The first is the application of long term ultrasound which reduces thresholds by increasing so to say, the signal-to-noise ratio, by exhausting homeostatic system. The second possibility is to use the specially constructed mode of pulsed ultrasound where pulse duration and pulse intervals are chosen in such a way as to make the dynamic regulatory systems in the object ineffective. The third possibility to obtain experimentally real thresholds, not shifted by regulatory processes, is the investigation of ultrasound action on specialized receptor tissues.

The examination of literature data on bioeffects of ultrasound has shown that apparent and real thresholds may differ by a few orders of magnitude.
BIOACOUSTICS OF FLYING INSECTS

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The wingbeat frequency of an insect may be regarded as a species specific parameter along with the other aerodynamic parameters. Laboratory data of the flight sound enables one to determine the wingbeat frequency. These parameters are useful in the (1) computation of the energy requirements of flying (complimentary to metabolic investigations), (2) identification of species, and (3) protection of crops. Substantial progress has been made and interesting results are obtained by recording and analysing the wingbeat frequencies of six species of insects (*Periplaneta americana*, *Cicada* sp., *Tesseratoma javanica*, *Chrysocoris purpureus*, *Apis* sp., *Bombus* sp.), in the frequency range 25-125 Hz. These results are also explained theoretically by considering the rate of mass-flow of air induced downward when the flier is in its normal state of hovering.

Thus the study of bioacoustics of insects with reference to their wingbeat frequency is considered as a fruitful line of investigation.
ELECTRO-AcouSTIC ANALOGS IN THE STUDY OF ANIMAL AUDITORY SYSTEMS

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The use of electric network analogs to assist in the analysis of the auditory systems of simple animals is fruitful for the understanding of many aspects of system behaviour, particularly frequency response and directionality [Fletcher and Thwaites Quart. Rev. Biophys. 12 25-65(1979)]. Among the most suitable systems for analysis are those in which two relatively simple ears are coupled acoustically through a mouth cavity, a trachea or an auditory canal. This acoustic analysis is, of course, only a first stage towards understanding the total system response, which may be modified very significantly by the frequency response of the neural transducers and by neural interaction at a higher level in the auditory chain.

When an appropriate analog network has been derived for a particular auditory system, it becomes of importance to determine numerical values for the various impedances involved by making acoustical measurements on real biological specimens. This also checks the physical reality of the model. Recent progress with this type of analysis in our laboratory will be described.
In the Tenth International Congress on Acoustics, L.R. Gavrilov and E.M. Tsirul'nikov presented a paper titled "Transmission of Auditory Information to Man by Means of Focused Ultrasound." The authors are affiliated with the Acoustical Institute, USSR Acad. Sci., Moscow USSR.

The paper discusses investigations that have demonstrated that auditory information can be transmitted to man by means of focused ultrasound having a frequency of some megahertz and amplitude modulated by informational signals which included music and speech. It has been shown that auditory sensation may be evoked by various causes: a) by demodulation of AM-oscillations, producing mechanical sound waves which travel through the head tissues and reach the ear labyrinth, b) by direct action of ultrasound frequency oscillations upon the ear labyrinth structures when the acoustical axis of the transmitter passes through the labyrinth, c) by action of ultrasonically evoked in tissues electrical pulses on the extralabyrinth nerve structures, when the focal region lies in the skull bones, outside the labyrinth.

Different causes of auditory perception by means of ultrasound underlie the attempts of creating hearing aids for individual patients who lack normal receptor cells but retain other intralabyrinth elements, say ganglion spiralis, as well as for those who lost their intralabyrinth elements but retain the functioning auditory nerve fibres.

References
MINIATURE HYDROPHONES FOR THE MEASUREMENT OF ULTRASONIC FIELD

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Ultrasonic diagnostic equipment is widely used in the field of medicine. Doppler method and pulse reflection method and others are put to practical use. Especially pulse reflection method has been popularized. In the equipments of this field, ultrasound of 1.5 - 15 MHz in frequency and 1 mW/cm² to 10 W/cm² in intensity (mean value or peak value). For the evaluation of the performance of the equipment, measurement of the sound field generated by the equipment is significant.

IEC SC29D has been deliberating on the standards of hydrophones for the measurement of sound field of ultrasonic medical equipment. Outline of the draft of the standards is as follows. Operating frequency range required is more than 2.5 octaves between 0.5 - 15 MHz. Frequency response, sensitivity, sound pressure range, temporal stability and others are stated. Transducer element approximately 0.5 mm in dimension is recommended.

This paper reports on miniature hydrophones for the measurement of sound field which suit the standards. Fig. 1 shows the constructions of the hydrophones. As transducer elements, PVDF and PZT are used respectively. Since the capacitance of the transducers is small, an impedance converter is employed.

Miniature hydrophones which were made this time have wide frequency range and broad directivity. Accordingly they can be used in the measurement of sound field of pulse ultrasound.

Reference:

Fig.1 CONSTRUCTIONS OF THE HYDROPHONES
Performance comparison of ceramic and polymer miniature transducers for sensing ultrasonic fields

Danish Institute of Biomedical Engineering;*  University of Surrey, U.K. x

Lewin*  Peter A.
Chivers x  Robert C.


In the past few years miniature ultrasonic probes have gained attention and recognition as convenient tools in the field of ultrasonic dosimetry.

This paper describes the development and construction of two different miniature piezoelectric transducers for sensing ultrasonic fields. The first type of miniature probe employs as an active element a piezoelectric ceramic disc having a diameter less than 0.4 mm, and a natural frequency of about 22 MHz.

The second probe, of similar dimensions, uses a piezoelectric polymer (P F2) film as the sensing element.

The development of the transducers will be presented together with a comprehensive comparison of their performance. Design considerations include the backing impedance and its effect on the (overall) absolute sensitivity and the frequency response of the sensors. Long term stability studies have been undertaken and some of these results will also be presented. Details of the procedure for the calibration of the miniature transducers will briefly be discussed with attention to the limitations of the technique.

The use of the probes for making measurements of both the spatial and temporal behaviour of acoustic pressure radiated by ultrasonic biomedical transducers will be illustrated by several practical examples, as will the application of the sensing probes to in vivo measurement of the acoustic pressure.

Once calibrated the probes may be used conveniently (if carefully) in any hospital by a physicist (or technician) for accurate, rapid assessment of medical transducer beam profiles and radiated acoustic pressure amplitude.
ON THE EXPOSIMETRY OF REAL-TIME ULTRASONIC SCANNERS

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CHIVERS *
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The advantages of real-time scanners, in terms of the provision of dynamic diagnostic information, and rapidity and ease of use, imply the continuously increasing widespread use of such devices. The major variants - linear arrays, phased (including focussing) arrays, and revolving multielement transducers ('spinners') - each demand distinct treatment within a general discussion of performance assessment.

The paper presents such a generalized approach, briefly, in terms of conventional parameters such as; frequency, beam-profile, pulse waveform, resolution, intensity, peak acoustic pressure and focussing properties. The main discussion is directed towards the relationship of measurements of these parameters to the exposure of a pre-specified volume of tissue irradiated by the different designs of machine mentioned above.

Selected measurements on different types of machine are reported and are discussed in relation to the precautions to be taken in the practical measurement of these parameters and in the interpretation of the results. The preliminary design of a mobile test facility will also be presented.
TRANSCLUDER ARRAY DESIGN IN MEDICAL ULTRASOUND.

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Real-time systems for ultrasonic visualization of internal structures in the human body are widely applied now in medical centers all over the world.

The first generation of linear array systems were based on transmission and reception on one or a few elements of the array. Nowadays beamforming techniques and even dynamic focusing have been introduced. These techniques use a larger number of individual elements for the completion of one scan. In order to minimize the complexity and the cost of a medical ultrasound system the design of an array with as few elements as necessary is required.

Normally the array's of medical ultrasound systems have an array periodicity far above half a wavelength.

With these array configurations grating lobe artefacts are introduced in the images. In technical terms the effects of the width of individual elements and the total number of active elements in an array will be discussed for medical ultrasonic imaging.

A number of techniques exist for reduction of the grating lobe effects. Attention will be paid to the influence of the shape of the individual elements on beamforming and grating lobe.

Anti-aliasing techniques for effective reduction of grating lobe, as obtained with particular element forms, will be described.

These techniques are successfully introduced now in a series of ultrasound systems.
A SURVEY OF ULTRASONIC THERAPY MACHINE FIELDS

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Since the publication, in 1963, of the first standard on ultrasonic therapy machine performance and the procedures to be used in its assessment, considerable refinements have taken place in the development of methods of measurement of the ultrasonic field parameters.

The paper reports a survey of ten commercially available therapy machines (for use in physical medicine), in which particular attention has been paid not only to the relationship of manufacturer's data to parameter values, but also to the limitations of the techniques used for the measurements.

Measurements of total power output made using a commercially available radiation force balance are compared with values of total power output derived from measurements of peak pressure amplitude made with a purpose-built independently calibrated miniature hydrophone probe. Three methods for assessing the effective area of the transducer - (using the hydrophone, a liquid crystal display and Schlieren techniques), have been used and the results show general agreement.

The hydrophone has been also used to compare the waveforms emitted by the equipment.

The data from these different measurements have been used as the basis of a comparison of the values of power output and transducer area quoted by the manufacturer and those found in practice. It is shown from the results that although considerable care has to be taken in the measurement procedures and the interpretation of the results, that the relationship of the quoted and measured values is more disparate than warranted by the current status of measurement techniques. The situation is thus similar to that reported by the comprehensive FDA test of 1974 (Stewart et al).

No significant differences were observed between new machines and those which had been routinely used for many months. In the case of the latter the discrepancies may be due to poor maintenance protocols.

The practical problems of coupling are discussed briefly in terms of the acoustic properties of different commercially available coupling media and a method for the instantaneous indication of the effective applicator contact area.

Reference

MEASUREMENT METHOD OF THE SENSITIVITY OF ULTRASONIC DOPPLER FETAL DIAGNOSTIC EQUIPMENT

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INTRODUCTION

Ultrasonic Doppler diagnostic equipment is widely used in the fetal diagnosis. In 1978, JIS (Japanese Industrial Standards) draft of 'Ultrasonic Doppler Fetal Diagnostic Equipment' was made by the committee, when measurement method of overall sensitivity of the Doppler equipment was investigated. This paper reports on the outline of this measurement method.

METHOD OF THE SENSITIVITY MEASUREMENT

Since the Doppler signal arises from moving reflectors, investigation of reflectors appropriate for the sensitivity measurement was made and a small steel ball set up to the recording pen of pen motor was employed. Fig. 1 shows fundamental setup for the measurement method of the overall sensitivity of Doppler equipment. Small steel ball (2.38 mm φ), the moving reflector, is attached to the end of the arm of pen motor which is driven by a function generator. Hence the amplitude, waveform and frequency of the motion of the reflector can be set with ease. Oscilloscope and electronic voltmeter are connected to the audio frequency output terminal (terminals of loud speaker) of Doppler equipment under test. Measurement and observation of the output voltage and waveform of Doppler signal together with detection by ears. Maximum overall sensitivity of Doppler equipment is defined from the total attenuation between transmitting and receiving when defined level of Doppler signal from the reflector is obtained in the defined least S/N.

RESULT

By the measurement method, calibration of the sensitivity of Doppler equipments was made. By using the calibrated equipments, detections of the fetal heart beat in the 12th week of pregnancy were made in several institutions. It was found that these equipments could detect the fetal heart beat at the sensitivity of 73 – 87 dB.

REFERENCE:

Quantitative Analysis of A-mode Ultrasound Waveforms of the Abdomen

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We have implemented a computerized data analysis system for ultrasonic tissue characterization. One important feature of this system is its ability (via a fast ADC) to record both the phase and amplitude of A-mode waveforms selected through regions of interest on a conventional B-mode ultrasonogram. The measuring system is well calibrated and characterized so that data obtained by it are essentially independent of the instrumentation used for acquisition. One section of the system, designed specifically for one-dimensional or A-line analysis, is built around PDP 11/40 and PDP 11/60 computers. This paper will describe the results of comprehensive measurements taken in vivo on several organs of the abdomen, especially the liver as well as measurements taken on tissue phantoms. The data have been subjected to various signal processing procedures including several correlation techniques and several forms of spectral analysis with the intent of developing one or more methods for the quantitative characterization of tissue. These various signal processing will be compared and evaluated and the problems associated with a characterization scheme noted. One specific procedure, based on the power spectrum of the cross-correlation of the incident and reflected A-mode waveforms, appears particularly promising.
Applications of holographic interferometry in otology
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Characteristic changes in the vibratory pattern of the human tympanic membrane caused by pathological alterations of the mechanical properties of the middle ear can provide the possibility of a differential diagnosis in cases of conductive deafness without opening the tympanic cavity. This could be proved on human temporal bone preparations using holographic interferometry, a laser-optical technique for threedimensional, contactless, high resolving vibration analysis. For studies on the transfer function of the middle ear the used double-exposure holographic method turned out to be capable of investigating vibrational unsymmetries and phase-opposition of different parts of the tympanic membrane.

In order to provide a better optical access to the eardrum the outer ear canal was removed in these investigations on human temporal bone preparations. To overcome the difficulty in simultaneously taking holograms and applying the vibration eliciting sound pressure through the intact outer ear canal in living man a special closed acoustic system was developed. For phase investigations of the sound pressure oscillation it was proved that this system has no resonances at the frequencies used. Thus, it was possible to detect a vibration pattern of the tympanic membrane in patients with regular middle ear status characteristic depending on amplitude, frequency and phase of the sound pressure oscillation, which corresponds to that found in temporal bone preparations. First results on patients with pathologically altered middle ear mechanics are described. Increasing feasibility of this technique for clinical use could be gained by using flexible fibre optics for easy positioning of the acousto-optical system in relation to the patient, and by generating holograms on a thermoplastic film material within 10 s without wet development of photographic plates.
A REAL-TIME ULTRASONIC DIAGNOSTIC SYSTEM FOR SIMULTANEOUS IMAGE DISPLAY

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High speed electro-scanning ultrasonic diagnostic equipment is now widely used in clinical examinations. Since still images of a dynamic organ like the heart and a fetus can be obtained, its utility has expanded for cardiology. The system presently used, however, does not permit quantitative observations.

Recently various digital image processing technology has been developed for ultrasonic diagnostic systems. The authors have developed a digital scan conversion unit with a frame memory for B, C and F mode methods. By utilizing this technology, we have developed a new ultrasonic diagnostic system in which a real time image and some still images are mixed and displayed on one display unit. A real time image of a heart and the images of extended and contracted moments, for example, are displayed simultaneously on one display unit, so that differences between the extended and contracted shape or a valve displacement of a heart can be observed in the displayed image.

SYSTEM CONFIGURATION

A block diagram of the developed system is shown in Fig.1. A B mode image obtained from the reflected echo sensing unit consists of 114 scanning lines. A 1 line analog echo signal is sampled at 256 points and sampled data is converted into a 4 bit digital echo signal by the A-D converter. These signals are sent to three digital frame memories.

In each frame memory which consists of 256 x 128 pixels, the scanning speed of the reflected echo sensing unit is converted to standard NTSC TV scanning speed.

The contents of a frame memory are frozen at any time with a signal from its own input controllers. Images frozen at different moments, for example, the extended and contracted moments of a heart are read out from frame memories by output controllers. The display area for stored images can be varied at will, and read out digital echo signals are multiplied by weight factors for mixing in the output controller. Three images are added in the mixing circuit through output controllers and a new image consisting of three image information is obtained. The new digital image signals are converted to analog signals at the D-A converter.

These analog signals are mixed with TV synchronous signals, converted to RF signals and displayed on a home TV set. On the other hand, another display method was used for color displays. Each digital echo signal from three output controllers is converted to an analog signal for R, G or B input to the color TV monitor. These analog echo signals and TV synchronous signals are sent to R, G and B terminals of the RGB monitor, and a color image consisting of three red, blue and green images is displayed.

CONCLUSION
(1) Real time dynamic and still images can be observed simultaneously.
(2) Dynamic images can be frozen at any time.
(3) Images containing two or three image information can be observed.
(4) These images can be displayed on a standard NTSC TV and RGB monitor.
MEASUREMENTS OF ULTRASOUND VELOCITY IN SOLUTIONS OF BIOLOGICAL MOLECULES FOR INVESTIGATION OF THEIR STRUCTURAL PROPERTIES AND MOLECULAR INTERACTIONS.

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Recent development of precise methods of ultrasound velocity measurements in small volumes of liquids opened new possibilities in acoustical investigation of biomolecular problems.

There are many important molecular characteristics of solutions which can be investigated by acoustical methods except well-known relaxational properties of substances obtained by ultrasound absorption and velocity dispersion measurements. The value of ultrasound velocity in solution is defined by the totality of inter- and intramolecular interactions. In the region of small concentrations in case when there is no interactions, velocity of ultrasound increases with concentration as an additive function of solution componenets. Infringement of additivity reflects solute-solvent interaction resulting in changes of acoustical properties of solution components or in solvation phenomena. Nonlinear dependence of ultrasound velocity on concentration reflects solute-solute interactions.

We have measured ultrasound velocity in dilute aqueous solutions of aminoacids, proteins, nucleic bases, nucleosides, nucleotides and vesicular lipid membranes at different temperatures and pH. Measurements of pH dependences allowed to obtain quantitatively the hydration effects of ionization of different atomic groups of aminoacids and nucleic acid derivatives. Contributions of different sites of investigated molecules to solute-solvent interactions are separated. The thickness of hydration layer is evaluated. Hydration effects and association constants of stacking interaction of nucleic bases are obtained from concentration dependences of ultrasound velocity in solution. Intrinsic compressibilities of globular proteins and bilayer lipid membranes are calculated.
TECHNIQUE FOR PRECISE MEASURING THE ULTRASOUND VELOCITY IN 0.1 ml SAMPLES OF BIOLOGICAL LIQUIDS AND SOLUTIONS

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Traditional methods of ultrasound velocity measurements in liquids are usually inapplicable to investigation of biological solutions because of insufficient sensitivity and large volume of measuring cells.

In this presentation a device for ultrasound velocity in 0.1 ml volume of liquids with the precision of 10⁻⁶ in the frequency range of 6-9 MHz is described. The device is based on the fixed path interferometer or so called resonator method. In this method the velocity of ultrasound in the liquid placed in the resonator cell is determined from the value of frequency of the given resonance of resonator. Ultrasound velocity is linear function of the given resonance frequency.

The resonator cell unlike cells described in literature, contains no elements for parallelism adjustment, due to which drifts of the resonance frequency may occur, especially during temperature dependence measurements. The parallelism of the rigidly-fixed transducers is provided by the special construction of the cell. The transducers are made of the lithium niobate discs with diameter of 10 mm and resonance frequency of 10 MHz.

Resonator cell is connected to the electronic circuit which provides the generation of electrical oscillations at the frequency, corresponding to the maximum of the choosen resonance. The frequency is registered by digital frequency-meter.

To lower the requirements to the temperature stability in the cell, the differencial sistem is used. The velocity of ultrasound is determined from the difference of frequencies of resonances in two resonator cells, one of which is filled with investigated liquid and the second with the reference solution. The difference of temperatures in two cells does not exceed 10⁻³°C, so the velocity changes in the liquid equal to 1-2 mm/sec can be easily detected. The time for one measurement is 3 minutes.

The device is used for biochemical investigations.
Ultrasonic properties of normal and diseased human liver.
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Hill, Christopher R. and Bamber, Jeffrey C.
I.C.R., Royal Marsden Hospital, Sutton, Surrey, U.K.

Ultrasonic speed, attenuation coefficient and back-scattering crosssection per unit volume have been measured, as functions of frequency and orientation, in excised specimens of normal and cancerous human liver. Acoustic speed is found to be a good index for distinguishing in vitro between specimens of tumours and normal liver. Attenuation coefficient and backscattering crosssection at a given frequency are of comparable value only when the data are corrected for variations between one subject and another. From the limited data yet available it appears that livers infiltrated by diffuse malignant disease possess ultrasonic propagation properties markedly different from normal.

From measurements of the water, fat and collagen contents of the examined tissues, it is found that sound velocity decreases with both increasing water and fat contents. Water content shows an inverse relationship to both attenuation and backscattering coefficients, whilst a direct relationship is found between these acoustic characteristics and the fat content. Collagen appears to be of secondary importance in this situation.
Non-thermal action of ultrasound on cells under heat stress
Institute of Cancer Research.
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Mammalian cell suspensions (V79 and HeLa) maintained at
temperatures in the hyperthermia range 42-45°C, have been
irradiated with continuous 3 MHz ultrasound at a spatial
average intensity of 3 W cm⁻². An increase in rate of cell
killing was observed, relative to that to be expected from
temperature alone. On the evidence that the irradiation
did not give rise to cell lysis, and that irradiation for up
to six hours at 37°C did not lead to measurable cell killing,
the effect is believed not to be due to cavitation. Arrhenius
parameters for the heat inactivation survival curves have been
determined and lend additional support to the belief that the
effect is non-thermal in nature and appears, rather, to be
due to "direct" action of ultrasound on cells already
weakened by heat stress.
DEFORMATION BEHAVIOUR AND VIBRATION PATTERN OF THE HUMAN SKULL
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Holographic interferometry as well as accelerometry have been applied to study both the deformation behaviour and the vibration pattern of the human skull.

In a first experiment a special device was constructed to expose a specimen of a human skull to a shock. Immediately after the application of the shock (0 - 500 μs) the deformation of the skull was measured both by means of double-pulse holography and with an accelerometer, connected to a storage oscilloscope.

In the second experiment the skull was stimulated continuously using a bone vibrator. The different vibration patterns of specified regions of the skull were visualized by means of holographic interferometry. The dependency of deformation behaviour and vibration pattern on both a transient and a continuous sinusoidal stimulation was studied.

Effects due to rigid body motions which often occur in experiments can only be eliminated if experimental setup and fixation of the object are interferometrically stable. A fixation, however, influences the vibration pattern of the investigated object. It shall be demonstrated how the effects of rigid body motions can be eliminated using a Moiré technique.
G. ULTRASONICS, QUANTUM ACOUSTICS
AND PHYSICAL EFFECTS
OF SOUND
Acoustic Testing of Poles

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Wooden poles are traditionally tested in situ by "sounding"—striking with a hammer near their base and making a subjective assessment of the pole from the sound emitted. The efficiency of the acoustic testing of poles has been assessed by investigations of the propagation of flexural, longitudinal and torsional waves in the pole. Wave propagation is samples of poles suffering typical service damage such as wet rot and termite attack has shown that it may be possible to distinguish the damage using these acoustic techniques.
Measurement of depth of steel bar in concrete with electromagnetic impact driving method

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(1) INTRODUCTION

It is desirable to develop the non-destructive technique to detect the location of a steel bar which is buried in comparatively shallow places such as in the depth of less than 15 cm below the concrete block surface.

The authors devised a new detection method of measuring the location of a steel bar within concrete on the concrete surface through an electromagnetic impact driving method and examinations about its possibility were performed.

(2) ELECTROMAGNETIC IMPACT DRIVING METHOD

A steel bar in concrete and occurred impulsive electromagnetic flux at a spiral coil set on the concrete surface are interlinked (as interlinkage). Then, the impulsive electromagnetic attraction force acts between the driving coil and a steel bar in concrete. As its result, a steel bar hits the concrete block and an impulsive sound wave is radiated into the concrete. Impulsive sound wave was radiated from a steel bar in concrete, which was caught with a piezoelectric type receiver cemented on the concrete surface.

(3) MEASUREMENT AND RESULT

The driving coil is a flat spiral coil (inner dia.: 12.5 cm, outer dia.: 15 cm) cemented on a bakelite plate. The receiver, consisting of two PZT disks stucked together, was buried into urethane resin bar, because the duration of electromagnetic induction wave and a propagation time of radiated impulsive sound wave are seperately measured. Experiments on the detection of a steel bar at depth 10.5 cm under the concrete surface were performed under the condition of the driving current frequency $f_e=38$ KHz and driving energy $E=8$ joule. Consequently, impulsive sound wave radiated from a steel bar was observed at 135 $\mu$ sec after beginning of electric discharge. Lag time in urethan resin bar (110 $\mu$ sec) being taken into consideration, sound velocity in concrete turn out to be 4176 m/s. This value was in good agreement with the sound velocity in concrete measured with pulsed sinusoidal wave.

As its result, it is clear that the electromagnetic impact driving method is useful for the detection of a steel bar buried at depth of less than 15 cm below the concrete surface.
Detection of Micron-size Particles and Bubbles in a Flowing Liquid
MicroPure Systems, Inc.

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An experimental analysis has been carried out on the scattering of an ultrasonic pulse from microparticles and microbubbles in the size range 0.01 < ka < 1.0, where a is the radius of the contaminant and k is the propagation constant of the host liquid. The system consists of a 7.5 MHz ultrasonic test cell, in which the wall of the cell serves as an acoustic concentrator, a commercial pulser, which emits an eight-cycle pulse, and a receiver with a dynamic range of 46 dB.

The gain due to the focusing can be as high as 60 dB, so that the overall gain of the system is sufficient to detect particles below one micron in diameter. Experiments have been carried out on latex particles; these particles are available in six sizes, ranging from 0.39 to 14 μ in diameter. All were detected. The smallest identifiable air bubbles thus far generated in this research have been 30 μ in diameter, but scattering from smaller, uncalibrated bubbles has been observed.

The cell can be inserted in a continuous flow system so that on-line monitoring of randomly occurring contaminants is possible.
ULTRASONIC POWER MEASUREMENTS IN THE MILLIWATT RANGE

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Ultrasound has been widely used in the field of medical diagnosis for more than two decades now. Despite this fact no procedure for power measurements in the milliwatt range which is easy to handle and calibrate, is available yet.

The determination of the total radiated ultrasonic power of medical devices in the watt range is usually performed by measuring the radiation force using the float technique /1/. The float consists of a cone-shaped reflector with a stem dipping into a liquid which is not miscible with water and which has a density larger than 1, e.g. tetracarbonchloride. The radiation force displaces the float until its stem moves into the heavier liquid far enough to develop an equal force. The main advantages of this method are, first, that the calibration can be carried out very easily by dropping weights onto the float, and secondly, that the float is self-centering. Unfortunately, at power levels in the milliwatt range, the radiation force is only of the order of \(10^{-6}\) Newtons and the use of a float system as a measuring device creates a lot of problems due to adhesive and surface tension forces acting on the stem of the float.

It is the aim of this work to investigate the applicability of the float technique to the milliwatt range. To increase the sensitivity of the system, the difference in density between the two immiscible liquids and the stem diameter should be as small as possible. Reid /2/ has proposed a water-heptanoic acid system. However, in this case the transducer, the radiated power of which should be determined, has to be immersed in heptanoic acid. So far, our concern has been to decrease the diameter of the stem and to reduce the disturbing forces in the boundary layers.

Measurements have shown that these disturbing forces are considerably diminished when Teflon-coated stems are used. The sensitivities achieved of two systems measured were 0.70 mm/mW and 0.22 mm/mW. Assuming a smallest immersion length of 1 mm and a largest of 100 mm, the range over which measurements can be performed is about 1.4 milliwatt to 140 milliwatt for a stem diameter of 0.55 mm, and 4.6 milliwatt to 460 milliwatt for a stem diameter of 1 mm. The reproducibility of the measurements was about \(\pm 5\) %.

The mean values of the experimental results were compared with other standard methods at a frequency of 1 MHz and an agreement was found within a measuring uncertainty of \(\pm 10\) %.

/1/ IEC-Publication 150, Testing and calibration of ultrasonic therapeutic equipment, 1963

Design of low frequency high-power transducer
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How to decide a clamping-bolt diameter, that is one of the important problems for designing a low frequency high-power transducer, is discussed using an idealized model of the transducer shown in Fig.1 (a and b, radii of the clamping-bolt and of the transducer; 2c and 2h, the size of clamping-nut and the height of the transducer, respectively).

The clamping-bolt and nut has to apply compressive force to all the interfaces among components such as metal-blocks, piezoelectric elements and electrodes in order to propagate acoustic waves across the interfaces. The axial static stress $\sigma_z$ due to the clamping-bolt and nut, however, distributes along the radial direction as well as z direction, and has negative value for certain ratios among h/b, c/b and a/b. Then the condition of $\sigma_z > 0$ gives one criterion for deciding the clamping-bolt diameter.

To obtain available ratios of c/b and h/b, calculations of $\sigma_z |_{eq}$ are carried out by computer under the condition of c 1.6a, and the result is shown in Fig.2, where hatched area is available. This result is experimentally proved to be reasonable.

Fig.1. Fig.2.
SOUND PROPAGATION IN SUPERCOOLED LIQUIDS

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Supercooled water under atmospheric pressure has recently aroused considerable interest because of an anomalous behavior of some physical properties as density, isothermal compressibility, specific heat, viscosity, etc. (1). As sound propagation is concerned, ultrasonic (2) and hypersonic (3) measurements also show divergences of both sound velocity and absorption from projections of the behavior in the stable region. Different explanations have been proposed to interpret such results. In addition it is not clear if these anomalies are connected with a thermodynamic singularity related to a peculiar structuring of the metastable liquid water (cooperative formation of an open hydrogen-bonded network) or with the limit of mechanical stability of such liquid phase (1). In the latter case one would observe similar anomalies for all the liquids when appropriately supercooled.

Previous ultrasonic velocity and attenuation measurements (4) in supercooled liquids of different type show the existence of anomalies which qualitative behavior appears to be similar to that observed in water. In order to get a wide set of experimental data we are performing accurate measurements in benzophenone and diphenylamine in a large range of temperature and frequency. Classical phase comparison method and pulse technique are used respectively for sound velocity and absorption measurements. They however require large amounts of the sample and this constitutes a serious limitation to the degree of supercooling attainable. Preliminary results obtained in benzophenone \(T_c=48.2^\circ C\) in the temperature range between 38°C and 87°C, can be summarized as follow:

a) no sound velocity dispersion is found between 5 and 65 MHz;
b) on approaching the metastable region by decreasing temperature, sound velocity shows positive deviations from the linear behavior exhibited in the stable phase;
c) for all the frequencies, in the range between 5 and 195 MHz, the sound absorption coefficient (as expressed by \(\alpha/f^2\)) shows considerable increase in the metastable region;
d) \(\alpha/f^2\) is higher at 5 MHz.

REFERENCES

The propagation of elastic surface waves in a system consisting in only one layer deposited on a half-space substrate was studied by a number of authors (1,2,3,4). Velocity of that quasi-Rayleigh wave is found when cancelling the 6 x 6 boundary conditions determinant. The numerical calculation using large computer programs is the only way to find accurate solutions, although in many practical cases of thin films deposited on either isotropic (2) or crystalline (4) substrate, considerable simplifications in good agreement with experiments can occur as long as \( h/\lambda << 1 \) (\( h \): film thickness, \( \lambda \): wavelength).

We study here both the theoretical and experimental case of layered media for any thickness of \( N \) different layers on an isotropic half-space substrate.

THEORETICAL STUDY

The elastic waves velocity in a \( N \) different layers system is calculated, when respecting compatible conditions, and the numeric calculation program is written. It consists in cancelling a \((4N+4) \times (4N+4)\) determinant which is the mathematical representation of boundary conditions for stress and displacements between the different layers and on the free surface. In calculating the deformation tensor the components of the displacement-vectors are also calculated for any depth, depending on the vibration frequency and on the layers thickness: the results give the location of vibration energy in the system.

This problem is illustrated for several layered Al-Ni systems. The corresponding propagation velocity dispersion curves and energy location are given.

EXPERIMENTAL STUDY

In a first time variable thickness layers of Nickel (0-130 \( \mu \)m) are deposited on Aluminium substrate by adapted sputtering. In a second step, we transform some of the previous devices by deposition of Aluminium layers (thickness 0 to 35 \( \mu \)m) produced by an electron beam gun apparatus.

Absolute velocity measurements in the (3 MHz - 20 MHz) frequency band are performed by the knife-edge method: reflexion of a laser-beam on the vibrating surface (5-6). Our experimental results are in very good agreement with theoretical calculations.

Modellisation of gradient properties for a cemented steel by a 5 layers system is also introduced and compared with a previous simplex model (7).

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(5) R.L. WHITMAN et al, Appl. Optics, 8, 1567 (1979)
(7) FLAMBART, Rapport interne du CETIM, Juin 72, Congrès Mesucora, 7, 37, (1973).
SCATTERING OF BG WAVE BY A GROOVE ON THE SURFACE OF A 6mm CRYSTAL

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Using surface perturbation technique, the scattering of BG wave by a groove on the surface of a 6mm crystal converts into a generalized Lamb problem: not only an equivalent mechanical source distribution, but also an equivalent electric source distribution exist on the crystal surface. An analytical solution is obtained.

Some characteristics of the reflected surface wave and the scattered bulk wave are obtained for both metallized and free surfaces. Table 1 lists the calculated energy reflection coefficient of surface wave $\Gamma_s$ and energy scattering coefficient of bulk wave $\Gamma_B$ for PZT ceramics and ZnO, CdS crystals.

<table>
<thead>
<tr>
<th>Materials</th>
<th>PZT-5H</th>
<th>ZnO</th>
<th>CdS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Metallized surface</td>
<td>$\Gamma_s$</td>
<td>$15.2 \cdot (h/\lambda)^2$</td>
<td>$0.62 \cdot (h/\lambda)^2$</td>
</tr>
<tr>
<td>$\Gamma_B$</td>
<td>14.1</td>
<td>4.6</td>
<td>2.4</td>
</tr>
<tr>
<td>Free surface</td>
<td>$\Gamma_s$</td>
<td>$1.2 \cdot 10^{-5}$</td>
<td>$7.4 \cdot 10^{-3}$</td>
</tr>
<tr>
<td>$\Gamma_B$</td>
<td>$2.1 \cdot 10^{-5}$</td>
<td>0.53</td>
<td>0.25</td>
</tr>
</tbody>
</table>

$h$ - groove depth; $\lambda$ - BG wave length

From the analytical solutions it can be seen that, for materials with strong electromechanical coupling and high permittivity, e.g. the PZT piezoelectric ceramics in Table 1, the reflection coefficient is very large for the metallized surface and very small for the free.
Transient behavior of nematic liquid crystals resulting from acoustical excitation.

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Objective. The purpose of this work was to investigate the rise and decay times respectively following the application and removal of acoustic surface wave excitation.

Experimental. The liquid crystal cell was similar to that used previously by the authors (1), and permitted of various thicknesses of liquid crystal being used on the same substrate. The source of surface waves was an interdigital transducer of frequency 20 MHz. By observing the cell through cross-polars the change of director orientation obtained in optical transmission was measured with a photomultiplier, whose output was displayed on a chart recorder.

Definition of rise and decay times. The method of defining these parameters is indicated in Fig. (1).

Results. Both $\tau_R$ and $\tau_D$ are very sensitive to cell thickness and vary considerably with the applied acoustical intensity. However, in general it is found that $\tau_R$ decreases while $\tau_D$ increases with increasing acoustical intensity. For cell thicknesses greater than 100 $\mu$m, $\tau_D$ increases initially very sharply with increasing acoustic intensity and then shows a tendency to saturate. For thin liquid crystal cells only a small variation of $\tau_R$ and $\tau_D$ is found.

The variation of the rise-time $\tau_R$ with cell thickness is ill-defined but shows a trend to a decrease with increasing cell thickness at low acoustical intensities.

The variation of the decay time $\tau_D$ with increasing cell thickness $L$ is, for low acoustical intensities, given empirically by

$$\tau_D = DL^\delta$$

where $\delta = 2.5$

APPLICATION DES ONDES DE SURFACE A L'ETUDE DE LA PHYSISORPTION

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INTRODUCTION

On se propose de suivre la condensation de gaz rares à basse température et sous pression réduite, sur un quartz piézoélectrique parcouru par une onde acoustique de surface en mesurant les variations de vitesse de propagation provoquées par le dépôt.

DISPOSITIF EXPERIMENTAL

Un quartz de taille Y, muni de peignes interdigités émetteur et récepteur d'ondes de Rayleigh, placé dans une enceinte ultravide à la température T = 79°K, est utilisé comme surface de dépôt.

Cette ligne acoustique est employée en dipôle de réaction d'un amplificateur maintenu à la limite d'accrochage et réalise un oscillateur dont les variations ΔF de la fréquence de boucle F (= 20 MHz) sont directement proportionnelles à celles de la vitesse de l'onde de Rayleigh. On fait croître par paliers la pression de gaz à adsorber (Ar, Kr, Xe). A chaque équilibre on note la variation de fréquence ΔF correspondante. La sensibilité du procédé (sing-around) est excellente : 10⁻⁷ (1).

RESULTATS

Admettant l'égalité à l'équilibre des taux d'adsorption et de désorption et celle des chaleurs d'adsorption pour toutes les couches, sauf la première (2), supposant en outre que les variations ΔF sont proportionnelles à la densité d'atomes déposés, il vient pour un gaz de pression de vapeur saturante p₀ à la température T :

\[ y = \frac{p}{(p₀ - p)ΔF} = \frac{1}{CΔF'} + \frac{C - 1_p}{CΔF'} \]

ΔF' : variation de fréquence pour une monocouche adsorbée ; C : constante liée à l'énergie d'adsorption de la première couche et à la chaleur de liquéfaction du gaz.

Les variations expérimentales de y en fonction de p/p₀ donnent pour l'argon une droite dont la pente et l'ordonnée à l'origine permettent de déterminer

\[ C = 32,10^3 \text{ et } ΔF' = 185,7 \text{ Hz}. \]

En l'absence d'échanges gazeux, la stabilité de fréquence est de 1 Hz ; la méthode s'avère donc capable de déceler un dépôt inférieur au centième de monocouche d'argon.

REFERENCES

(2) EMSCHWILLER G., Traité de Chimie-Physique "Méthode Brunauer, Emmet et Teller", p. 806.
RAYLEIGH WAVE ATTENUATION PRODUCED BY METAL FATIGUE

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PRINCIPLE

During fatigue cycling we measure, as a function of flexure degree, the acoustical attenuation of S.A.W. impulses propagating on the specimen surface submitted to an unidirectional bending stress.

An optoelectromechanical selector run by the testing machine drives the pulse emission (level $V_1$) on a chosen degree of flexure $F$ (0 to 15). After propagation on the sample we receive a pulse (level $V_2$). 30 μsec later. After amplification and detection the levels $V_1$ and $V_2$ are digitalised and added, for the same state of flexure $F$, over 50 successive mechanical periods.

We calculate
\[ \alpha = 1 - \left( k \sum_{i=0}^{50} \frac{V_i}{V_0} \right) \]

which fairly gives the attenuation expressed in 0.01 dB (1). The results are available either under analogic form for recorder or digital form. The later results are edited by internal printer of transferred to a magnetic tape by way of a microcomputer (HP 9825 A); at last the results accumulated on tape could be statistically treated and correlated.

RESULTS

Measures related here have been made on XC 55 steel at constant vibrational amplitude versus the number $N$ of cycles. With an amplitude smaller than the limit of fatigue the attenuations $\alpha_0$ at the plane position and $\alpha_m$ at the maximum flexure remain constant except sometimes during the first thousands cycles (stabilisation of the material).

With an amplitude superior at the limit several phases have been observed
1. The same phase of material stabilisation.
2. $\alpha_0$ and $\alpha_m$ remain nearly constant: no damage is acoustically observable
3. $\alpha_m$ is increasing, sometimes by steps, whereas $\alpha_0$ does not change: the first damage is produced but microcracks are often not optically observable.
4. $\alpha_0$ is increasing: the variations of $\alpha_0$ and $\alpha_m$ could be correlated at the propagation of cracks and $\alpha_0 N$ is connected with the progression rate in the length and depth of cracks. During phase 4 the form of the curve $\alpha$ versus $F$ (at $N$ given) is modified: nearly sinusoidal at the beginning it becomes roughly square when the cracks are important.

REFERENCE

ACOUSTIC MODES OF 127.86° ROTATED Y-CUT LiNbO₃ EXCITED BY IDT
Chinese Physical Society and the Chinese Acoustical Society

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On the substrate of 127.86° rotated Y-cut LiNbO₃, we have observed an acoustic mode propagating along the surface in some directions excited by IDT which is different from normal Rayleigh wave mode. It is also found that this mode can not be removed by the absorbing materials attached to the surface. As the propagating directions are about 30° - 50° departed from the X axis, this mode can be excited intensely and detected conveniently by method of the optical probing. The propagating velocities of this mode are found always faster than the velocities of corresponding Rayleigh wave in the same directions. The properties and propagating parameters of this mode have been measured and studied by the optical and electrical methods. According to these results some brief discussions are also given.
A phase interference method, which is used to measure the relative change of velocity of Surface Acoustic Wave, is presented. In this method we don't measure the absolute value of SAW velocity and the propagation length, but only the frequency and relative length ratio. It has the advantages of using simpler measuring equipment with higher accuracy over the conventional pulse technique and phase comparison method. We have got the relative change of velocity of Surface Acoustic Wave on metallized surface of yz-LiNbO$_3$, $\frac{AV}{V} = 0.02217$ with accuracy within 0.1%.

If it is possible to measure the length of propagation path accurately, the method can also be used to measure the absolute value of SAW velocity accurately.

The test data of $\frac{AV}{V}$ for several orientations of LiNbO$_3$ and quartz are given in the paper.
The desorption of gases from solid surfaces provides an important means of studying the dynamics of gas-solid interactions at the interface. However, most of the desorption methods presently used by investigators of surface phenomena are applicable to limited number of materials and especially dielectric surfaces are not amenable to most surface analysis methods.

A new technique was first described by Krischer and Lichtman with preliminary results on quartz [C. Krischer, D. Lichtman, Physics Letters, 44, 2, (1973)]. This technique seemed to be equally applicable to metallic and dielectric surfaces, but further work was needed to identify its major cause and determine its potential as an experimental tool in study of surface adsorption-desorption. In this paper results of experiments utilizing this new technique on single crystal quartz, and nickel and aluminum thin films will be discussed.

Implementation of the method requires the deposition of transmitting and receiving surface wave interdigital transducers on a piezoelectric substrate. The material to be examined is deposited in the acoustic beam propagation area, and when a surface wave is excited certain adsorbed species are observed to desorb.

Fig. 1. Block diagram of the experimental system.

We have examined CO, CH₄, CO₂ desorption from quartz and H₂O and H₂ desorption from nickel surface in addition to CO, CH₄ and CO₂.

Our experiments show ultrasonic energy to be the major cause of desorption and indicate that with this method it is possible to gather surface data on several materials of interest on which surface analysis is completely lacking.
ON THE ULTRASONIC SIGNAL NONLINEAR ATTENUATION IN THE MEDIA WITH LOWFREQUENCY NOISE

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Developing the experiments described in [1] we observed the dependence of the strong additional signal attenuation on the space length interaction, noise intensity and signal frequency. The theoretical calculations with taking into account the finite value of the spectral signal line width show that at the definite conditions it is necessary to introduce for the theoretical data obtained in [2] the corrections which depend in particular on the noise bandwidth. The obtained theoretical results are improving essentially the conformity experiment and theory results.

At the figure we show the dependence of the additional ultrasonic signal attenuation $\Delta$ in the water upon the parameter $\xi = \frac{\varepsilon \cdot \omega \cdot x}{c_s}$ ( $\varepsilon$ - nonlinear parameter, $c_s$ - sound velocity, $\xi$ - noise intensity, $\omega$ - signal frequency, $x$ - space coordinate). The spectra of noise are concentrated in the interval from 1 to 1,6 MHz, $\Delta \omega_{signal} / \Delta \omega_{noise} = 0,07$, I - calculations according to [2], II - calculations according to authors data, $\circ$, $\Delta$, and $\square$ experimental data for $\Delta$ in dependence on $x$, $\xi$ and $\omega$ correspondingly ( $x = 1 \div 40$ cm, $\omega_{2\pi} = 8 \div 14$ MHz, $\xi = 2,5 \div 7$ cm/sec).

References
Surface-longitudinal waves in solids

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It is shown in [4] that there is a additional longitudinal wave (apart from Rayleigh wave) on the boundary between an elastic half-space and thin liquid layer (or in the case of impedance load at the boundary of half-space). This wave is analogous to elastic leaky surface waves in a crystals [2]. This wave propagates along the boundary and its phase velocity is equal approximately the phase velocity of the longitudinal wave $C_0$. The main displacement component of the wave is parallel the direction of wave propagation. This component decays slowly both with depth $Z$ and the direction of propagation $X$. The volume component of this wave is transversal (shear) wave spreading from the boundary with the angle $\approx 45^\circ$. The displacements of this component are small and it increases slowly along wave fronts with distance from the boundary.

The purpose of this paper is on experimental observation of this wave. The experiments were performed with thick plate of fused silica. The surface of this plate was loaded by thin layer of oil. Narrow beam of ultrasonic longitudinal waves with frequency $\approx 6.7$ MHz was incident on this surface at a grazing angle $\Theta = 3^\circ$. Sound field in silica sample with the layer of oil and without the layer (free surface $Z = 0$) was investigated in detail by light diffractoin method.

It is shown experimentally that the component distributed in half-space qualitatively in the same way as the investigated longitudinal surface leaky wave appears in sound field of the sample when the thickness of the layer has such a value that it gives the necessary load on the surface. It demonstrates the generation of the wave in our experiments.

THE NATURE OF ZERO SOUND IN THE FERMI LIQUID $^3$He

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Zero sound in $^3$He is often described as being collisionless sound in which deformations of the Fermi Surface propagate as a sound wave. There is a phenomenological description, realized belatedly, which recognizes that zero sound is a viscoelastic effect. Until now, no one has used this description to obtain absorption and dispersion relations for all $\omega t$ values. These relations will be obtained and it will be shown that they agree remarkably well with experimental findings. In contrast with the results of Landau's Fermi-liquid theory which are formidable and largely opaque, those obtained here are simple indeed. Zero sound is thus the sound wave which propagates in viscoelastic $^3$He at large $\omega t$ and it is consequently an elastic wave. $^3$He is the only ideal viscoelastic substance, having a single relaxation time.
ACOUSTICAL PROPERTIES OF SINGLE AND MULTIPLE VISCO-ELASTIC MATERIALS

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This paper summarizes recent progress in wave propagation in fluid, elastic and viscoelastic layers. This work finds application in underwater acoustics, NDT, geophysics, SAW devices, medical acoustics, etc. The role of absorption in viscoelastic layers can be accounted for by introducing complex propagation constants into elastic theories. This necessitates the need for four independent material constants, one of which (the shear loss) is difficult to obtain experimentally. This difficulty has been overcome by introducing a physical model relating the shear to longitudinal absorption. In the case of inhomogenous layers (frequently encountered in practice) a theory which predicts effective material constants, including dynamic effects, is introduced. These constants can then be used in computer codes to calculate the transmission (T), reflection (R), and absorption coefficients for single and multiple layered configurations. The effect of finite thickness and fluid loading on the acoustic properties has recently been clarified by the application of resonance theory which readily allows the interpretation of resonances in T and R. Resonance widths and amplitudes are modified by the presence of absorption loss, and the interaction between resonances accounts for an apparent violation of the coincidence rule in viscoelastic layers.
A significant trend in recent studies of acoustic wave propagation relates
to amplification or attenuation of acoustic waves in piezoelectrics, chiefly on account of strong electromechanical coupling in such media. While theoretical investigations in this direction are undertaken by resorting to linearization of nonlinearities on account of space charge, amplitudes of the signal wave have not been adequately reckoned. On the other hand not-so-weak amplitude of signal waves would justify the consideration of parametric interaction of waves so as to ascertain loss or gain. The present paper is thus an attempt to look into these aspects and to seek, in particular, if there is a gain of energy of signal waves. The problem is made theoretically tractable through the use of the basic equations formed by the equations of electricity and of mechanical motion aided by constitutive equations of piezoelectric media. These inevitably lead to involved partial differential equations connecting mechanical displacement and perturbed electric field. The simple degenerate case i.e. when the frequency of the signal wave is half that of the pump wave is then considered and this permits the consideration of mechanical displacement and perturbed electric field as linear combinations of signal wave and pump wave. The relevant amplitude equations of two coupled waves are then obtained by assuming amplitudes to be independent of time but characterized by a slow spatial dependence. The pump energy loss due to non-linear interaction is neglected; when the amplitude of the initial pump wave is taken to be very large compared to that of the signal wave. The gain of energy of signal wave for the above degenerate case is investigated at crossover i.e. where the electron drift velocity is equal to sound velocity - an assumption which facilitates, doubtless, the solution of the problem. A gain of energy of the signal wave, as the present analysis shows, is perceptible at least at initial stages because of a pump wave whatever be its amplitudes. The investigation goes further in depth so as to reveal the possibility of a certain appropriate region for which there will always be gain of energy, accompanied by other regions for which there are also gains conditioned to spatial variations.
Existence of Negative Group Velocity in $S_1$ Mode of Lamb Waves

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**INTRODUCTION**

Previously we reported an experimental evidence of negative group velocity in $S_1$ mode of Lamb waves on Al plate(1). In this paper are described the behavior of phase velocities of true Lamb and leaky Lamb waves of $S_1$ mode, and the existence of negative group velocity in several plate materials.

**EXPERIMENT**

A stroboscopic schlieren technique was used to observe an ultrasonic pulse reflected from an Al plate of 1 mm thick immersed in water. Zero reflection occurred at specific angles of incidence from which phase velocities of Lamb mode were determined.

**RESULTS AND CONCLUSIONS**

Experimental results are shown in Fig.1. Phase velocities thus determined are very close to true Lamb velocities at least above 5 km/s. Negative beam displacements were observed in the region of positive gradient in $S_1$ mode implying negative group velocities. Broken lines in Fig.1 represent theoretical phase velocities of leaky Lamb waves which are separated into two branches in $S_1$ mode. Negative attenuation is predicted in the upper branch.

Theoretical Lamb velocities of several other materials are shown in Fig.2. Existence of negative group velocity seems to be a general behavior of $S_1$ mode. Also in $S_4$ mode negative group velocities are predicted in some of the materials.

**REFERENCE**


Fig.1 Theoretical and experimental phase velocities of Lamb waves on Al plate.

Fig.2 Theoretical phase velocities (upper) and group velocities (lower) of $S_1$ mode on several plate materials.
Investigation of Ultrasonic Wave Propagation in Bituminous Coal

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INTRODUCTION

The present study of acoustical properties of coal is motivated by the pos-
sibility of developing an acoustical echo-sounder for use in long-wall coal
mining. Such a sounder would function as the sensing element in a control
system which would guard against cutting into the substrate to which the
coal is attached. In view of the scarcity of data concerning wave speeds
and attenuation factors for coal, and of the variations in reported values
for these quantities, a series of laboratory experiments is being conducted
using samples of coal from the Pittsburgh seam in the United States of
America.

EXPERIMENTAL TECHNIQUE AND APPARATUS

The experimental method selected for the determination of acoustical proper-
ties of coal is a modification of the echo cancellation technique developed
by McSkimin¹. A sample is suspended in a water bath between two transducers.
Short wave trains are sent from both transducers. The signal received at
one of the transducers is displayed on a digital oscilloscope. The goal of
the measurement is to adjust the times at which the pulses are launched and
their amplitudes so that the transmitted pulse cancels the pulse reflected
from the face of the sample nearest the receiving transducer. The relative
gain G and delay phase angle φ can be used to determine the wave speed v_c
and attenuation factor α from

$$Ge^{-iφ} = (-i/2)(β - β^{-1}) \sin(k_c l)$$

where β is the specific admittance for coal normalized by that for water,
and k_c is ω/v_c + iα. In order to achieve the necessary control of amplitudes and time delay a two-channel digital pulsed sine wave generator has
been constructed².

EXPERIMENTAL RESULTS

Exploratory measurements have been made using a comparison of travel time
and relative amplitude for fixed transmission path with and without coal sam-
pies. The frequency used was 100 kHz; sample thicknesses ranged from 1.4 cm
to 2.5 cm. Representative values of wave speed and attenuation factor are
2.50 x 10³ m/sec and 1.08 Nepers/cm. More extensive measurements, using the
method discussed above, are continuing.

ON THE DISPERSION OF VELOCITY OF SOUND IN GRANULAR MEDIUM

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The velocity of sound in granular media such as sand and soil has been obtained at various frequencies by two different methods: (a) by direct measurement and (b) by computation from the phase constant values derived from impedance measurements.

The observed dispersion in velocity is attributed to the grain size of the aggregate and the viscous effects of the filling fluid following Rayleigh's classical model of loaded string. The influence of porosity on the dispersion of velocity in granular media has also been discussed.

The difference in the extent of dispersion obtained by the two methods is explained on the basis of difference in the group velocity in the case of pulse method and the wave velocity obtained in the case of continuous wave method.

The higher velocity values observed in soil as compared to that in loose sand are also explained on the basis of porosity and density of the medium.
Reconstruction of Moving Sound Sources from Acoustical Hologram recorded by using One-dimensional Detector Array

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Imaging acoustic radiation sources of a moving object such as an airplane or a train may be possible by means of acoustical holography where holograms are recorded using a one-dimensional detector array as shown in Fig.1. Reference Signals from an object under study have to be used for recording a hologram in order to detect any moving objects which may radiate white-noise like sounds, because temporal reference signals are applicable only to monochromatic or quasimonochromatic sound sources (1).

The hologram of a moving object recorded in such a way is different from a conventional hologram. Observation period for obtaining a hologram of moving objects is so short that averaging process can not be used and broad spectra of sources may degrade reconstructed images.

Then the effects of the sound spectra on reconstructed image from the hologram are studied. Reconstructed images of sound source at a constant velocity by using a one-dimensional detector are also obtained both theoretically and experimentally with satisfactory agreement.

This technique is proved to be a useful one to map moving acoustic sound sources because non-monochromatic sources as well as monochromatic ones can be detected.

Reference

Fig. 1. Geometry for recording holograms.
Reflectivity CT Using Impulsive Sound

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INTRODUCTION

Authors have developed a new imaging method which is capable of obtaining high resolutions both in lateral and longitudinal directions. This method required to transmitting and receiving transducers employed in the imaging system wide directivity and capability of handling an impulsive sound which has wide band in frequency domain. We reconstructed a clear image using the transducer satisfying these requirements.

RECONSTRUCTION OF IMAGE

Computer reconstructing algorithm for imaging in our method is different from that employed in X-ray CT which visualizes transmissivity distribution, in point of visualizing reflectivity distribution by using traveling time and magnitude of reflected impulsive sound. The reflectivity distribution is calculated at every point in a view by compensating delay time and attenuation so as to focus the back-projected sound at the point. Large aperture of the receiver array and use of the impulsive sound make it possible to obtain both high lateral and range resolution.

RECEIVER AND TRANSMITTER

We made a receiver array consisting of seven PVDF transducers arrayed linearly on a concave brass plate. It had the flat response in wide frequency range and the wide directivity. We scanned it along the circumference as shown in Fig.1 to complete aperical surface array consisting of 7x7 receiving points. In order to generate a precise impulsive sound, we designed a new transmitter. It radiates the sufficiently intense impulsive sound to wide direction in the water.

EXPERIMENT

Figure 1 shows the arrangement of the system. A micro-computer controls whole system, and sets up data for computer processing and display. The computer is used for both reconstruction and improvement of the image.

Fig.1 Block diagram of experimental setup
ULTRASONIC IMAGING SYSTEM USING ORTHOGONAL WAVE FRONTS

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A new ultrasonic imaging system which uses a priori knowledge about the class of objects under observation is proposed. The construction of the system is shown schematically in Fig. 1.

First, a set of orthonormal functions and the corresponding eigen values are derived by solving the second kind of Fredholm integral equation which has the second order spatial moment function of the class of objects under observation as its kernel.

Next the signals to be transmitted from the array transmitter corresponding to each wave fronts are derived by taking into account the process of the wave propagation in the medium and they are stored.

Then imaging is carried out by detecting the responses from the object for these orthogonal wave fronts transmitted in the order of the magnitude of the eigen value and reconstructing the image as the orthogonal function expansion with the responses as their coefficients.

This new method has special features of higher signal-to-noise ratio and faster image formation comparing with the conventional imaging methods which use essentially the beam forming technique.

A result of the reconstructed image by the practically constructed system is shown in Fig. 2. In this case the object which consists of two metal rods can be imaged by using only two or three wave fronts.

The details of the principle, results of numerical analysis of the special features of the system such as the high signal-to-noise ratio imaging as well as the concrete construction of the system and other results will be shown at the meeting.

Fig. 1

Fig. 2
Scanning Acoustic Microscope Composed of Plane-Wave and Focussing Transducers

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INTRODUCTION

A mechanically scanned acoustic microscope (SAM) has been developed by Quate, et al. So far, a pair of focussing transducers are employed to compose for a transmission-mode SAM as shown in Fig.1.1,2) The transmission-mode SAM is expected to be usefully accepted in the field of bioacoustics. In this paper a new construction is proposed for a transmission-mode SAM.

CONSTRUCTION AND EXPERIMENT

Typical constructions of our SAM are as shown in Fig.2. A specimen is illuminated by plane acoustic-waves in a water cell which are launched from the bottom of a stage by a thin film transducer. Acoustic waves transmitted through the specimen are detected by a receiving transducer focussing at the top surface of the specimen. The receiving transducer is scanned mechanically in a raster pattern to compose acoustic images. Experiments have been made at a frequency around 200 MHz. Several acoustic images observed by our SAM will be demonstrated.

CONCLUSION

A plane-wave transducer and a focussing transducer are satisfactorily combined to compose a SAM for transmission mode. Such an immobile stage as shown in Fig.2 is confirmed to be more favorable to mount the specimens, especially living biological tissues.

REFERENCES


Fig.1. Confocal pair of focussing elements to compose transmission-mode SAM. (a) Lenses, (b) Concave transducers.

Fig.2. Newly proposed constructions of transmission-mode SAM.
Schlieren photographs of the scattering of very short pulses of 3 MHz ultrasound from cylinders of aluminium, glass and polystyrene in water are presented. At $ka \sim 120$ ($k$ is the wavenumber, $a$ the cylinder radius) two types of waves are apparent: a Franz circumferential (creeping) wave, and geometrically scattered waves resulting from specular reflection and from waves transmitted into the cylinder. The positions of the geometrical wavefronts are in good agreement with the predictions of ray-tracing. The photoelastic technique is used to visualize the waves transmitted into the glass cylinder and multiply reflected inside it. Superimposition of the photoelastic and Schlieren photographs shows the complete process of scattering in both the cylinder and the water.
ACOUSTO-OPTIC MEASUREMENT OF ULTRASONIC BEAMS REFLECTED FROM FLAT BOUNDARIES

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Bounded ultrasonic beams reflected from liquid-solid flat interfaces frequently show phase incoherence. These phase shifts are clearly manifest in Schlieren images of the reflection. Measurements of the overall reflection coefficient, which can be calculated for different geometries, are thus made difficult.

Light diffraction techniques have been used in the past to determine the amplitude of sound fields; it is shown under what conditions of beam reflection it is still possible to use an acousto-optic technique to determine the reflection coefficient as a function of angle of incidence.

Work supported by the Office of Naval Research and the National Science Foundation.
Nonlinearities in the diffraction of light by adjacent ultrasonics

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THEORETICAL BASES

The diffraction pattern of a lightbeam diffracted by two parallel adjacent ultrasonic beams, with frequencies $\nu_1$ and $2\nu_1$, and showing a phase-difference $\delta$ has been calculated [1].

$$\rho > 1 \rightarrow J_{\pm 1}(\xi) = I_{\pm 1}(\xi) \left[ 1 \pm \cos \delta \sin \frac{\alpha_2 \xi}{2} \right]$$  \hspace{1cm} (1)

$$\rho < 1 \rightarrow \psi_1(2\xi) = \sum_{q=1}^{\infty} J_{\pm q}(\xi) J_q(\alpha \xi) e^{-iq\delta}$$  \hspace{1cm} (2)

with $\rho = \lambda^2 / \nu_0 \mu_1 \lambda^2$.

EXPERIMENTAL RESULTS

Out of the first expression one [2] derived a very simple method for measuring $\alpha_2 = \mu_2 / \mu_1$, the ratio of the two ultrasonic pressures. This experiments were performed in the Raman Nath region $\rho < 1$ and seemed to be very sensitive in measuring $\alpha_2$. Under certain restrictions for $\xi$, $\zeta < 1.5$ the mathematical model, being of the form (1) predicted very well the experimental results. Higher values however for $\xi$ disturbed this simple relation. In other words the dependence of $J_{\pm 1}$ against $\delta$ was no more cosinusoidal. Nonlinearities were asumed.

THEORETICAL EXTENSIONS

A mathematical model considering higher harmonics as well in the first as in the second medium was derived. These nonlinearities changed the mathematical formulation in such a way that it was impossible to find a solution in analytical form. Only numerical results were possible. Coming back to the expression (2) without neglecting higher terms, it was possible to derive an expression for the intensities of the first order of the following form

$$J_{\pm 1} = A + B \cos \delta + C \cos 2\delta$$  \hspace{1cm} (3)

This expression now described very well the experimental results for all values of $\xi$. Further in the linear case and $\rho > 1$ we could generalize formulae (1) for a fundamental tone and its $n$th harmonic. An identical expression for the intensities of the diffracted lightwaves was found.

REFERENCES

INTERACTION OF LIGHT WITH TWO ADJACENT PARALLEL ULTRASONIC BEAMS
PROGRESSING IN OPPOSITE DIRECTIONS
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INTRODUCTION
The Leroy's theory [1] of diffraction of light by two adjacent ultrasonic beams with frequencies $f$ and $nf$, respectively, where $n$ is a natural number, progressing in the same direction has been experimentally proved in the case of the ratios of frequencies 1:2 [2,3] and 1:4 [4]. The intensities of light in diffraction orders have depended on the shift in phase $\delta$ between the two ultrasonic beams. Now, we have been interested in the diffraction of two adjacent ultrasonic beams progressing in opposite directions Fig. 1. The case for frequencies $f$ and $2f$ was considered. Some theoretical relations were derived and their experimental verifications have been performed.

RESULTS
Relations for light intensities $I_n$ in diffraction orders were found [5] by calculations and expressed by the formulas in which components determined by the products of Bessel functions of arguments $a_1$ and $a_2$ are multiplied by a modulating factor with the frequency of $nf$, where $n$ - natural number, $a_1 = \frac{2\pi f}{\lambda}$, $a_2 = \frac{2\pi 2f}{\lambda}$, and $\lambda$ - amplitudes of changes in light refractive indices caused by the first and the second ultrasonic beams of frequencies $f$ and $2f$, respectively Fig.1.

Some experimental results in comparison with calculations for light intensities in $O_{-1}$ and $+2$ diffraction orders are exemplified in the Fig. 2.

DISCUSSION
By selecting appropriate parameters $a_1$ and $a_2$ for ultrasonic beams one can obtain various distributions of intensities. The influence of the phase shift $\delta$ between two ultrasonic beams on the light intensities is compensated in the case of opposite propagation of the adjacent beams.

REFERENCES
DIFFRACTION OF INTENSE LASERLIGHT BY AN ULTRASONIC WAVE
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The interaction between an intense laser beam and an ultrasonic field in a liquid medium is studied theoretically. The third order polarization leads to a system of nonlinear coupled wave equations, relating the fundamental light wave to its third harmonic. This system is closely related to the one established by Bloembergen et al., but contains periodically varying coefficients, due to the ultrasound acting on the liquid. In the case of large ultrasonic wavelengths the nonlinear differential system is solved exactly by a generating function method. The spectrum consists of two types of lines: ordinary diffraction lines to be found at the same place as in a linear medium, and intermediate diffraction lines due to the third harmonic generated in the acousto-optical interaction region. In contrast with recent theories (Jozefowska, Kosmol and Sliwinski, Mertens and Leroy) we find that the sum of all the calculated intensities of the diffraction lines is exactly equal to the intensity of the incident laser beam. A comparison with those earlier theories is also made.
Optical fibres for the detection of acoustic signals offer a higher sensitivity especially in liquid media than piezoceramic detectors 1), which are often used in the field of optoacoustic absorption spectroscopy (CAS) of liquid samples 3,4,5,6). The chopped monochromatic light absorbed by the liquid and thermalized via radiationless transitions causes periodical pressure fluctuations in the OAS-cell. The application of optical fibres instead of a piezoceramic detector as a transducer for the generated acoustic waves results in an improved detection limit as long as the noise level predominates the synchronous background of the OAS-system.

MOVING PULSED LASER THERMAL SOURCES OF SOUND

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Some results of the theoretical and experimental investigation into moving laser thermal sources of sound are represented in the paper. The sound field of these sources is analyzed for various velocities of motion and final trajectories of scanning, i.e., for the final durations of monochromatically modulated laser pulses as well as those of unmodulated pulses. Special attention is given to the case of scanning by the sources in the medium at the acoustic velocity. The conditions for maximum sound amplitudes are determined.

The source-scanning provides the following: 1) in the certain conditions the scanning leads to a considerable gain in the conversion coefficient of optical energy into the acoustic energy due to the optoacoustical effect; 2) it opens up new possibilities for controlling spatial and temporal characteristics of the sound pulse generated. Thus, by varying the speed of scanning and the modulation frequency of laser beam intensity within a laser pulse width one can obtain a source of sound pulses tunable over a wide range of modulation frequency, wave packet duration as well as direction of radiation. In the experiments a YAG-Nd\(^{3+}\) laser (\(\lambda = 106\mu\) m) was used. The optical pulse energy was 1.5 J, its duration was 0.2 ms, the light beam radius on the surface of water was \(a=0.25\) cm.

The pressure \(P=1.5,10^{3}\) dyn cm\(^{-2}\) was registered at a distance of 4 m from the end of the optical trajectory in the direction of scanning at the angle \(\theta = (a/L) = 0.1\) rad to the surface of water, with \(L=30\) cm trajectory length and with the velocity of scanning equal to that of the sound.
On Transitional Radiation of Sound by an Optoacoustical Source

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The radiation of sound arising in transition of a localized domain of thermal or strictional stresses developing under laser radiation over a sharp interface of two media is considered. The transitional radiation in the two media is shown to be a sum of two sound pulses: one caused by disappearance, at the interface, of the source moving in the first medium and the other - by a new source moving in the second medium. Basing on transitional radiation one can determine the form, dimensions and velocity of the moving optoacoustical source, in particular, the shape and duration of laser pulses, the shape of a strictional acoustical signal being described by the derivative of the temporal dependence of a thermo-optical acoustical signal. The conditions under which transitional radiation forms are analyzed, the effect of the Mach wave excited by the supersonic source motion included. Various configurations of optoacoustical sources are investigated and numerical estimations are given.
The optoacoustical concentrators suggested in the references $^1,^2$ make it possible to generate acoustical pressures in the focal region unobtainable in the area of optical radiation absorption in the nonevaporating generation conditions.

When the optical radiation is incident upon an acoustically rigid surface of the radius $R$, which serves the boundary of the liquid absorbing a laser pulse, in the focal point of the concentrators there forms temporally antisymmetric acoustical pulse. A maximum possible concentration coefficient is $K = \mu R$, provided $\tau \mu c \ll 1$ where $\mu$ is the optical absorption coefficient of the liquid, $\tau$ is the pulse duration, $c$ is the acoustic velocity in the medium involved.

In the experiments a spherical concentrator of $R=2.3\text{cm}$ filled with a solution of the dye in the ethanol with $\mu = 30\text{cm}^{-1}$ was investigated.

The characteristics of the exciting laser pulse are:
$\tau = 3.10^{-8}\text{s}$, pulse energy $E = 1J$, the beam radius on the concentrator surface $a = 1\text{cm}$. The amplitude of the pressure in the concentrator focus is $p = 2.10^7\text{Pa}$. The fields were registered by a piezoelectric receiver and shadow photography. Acoustical nonlinearities of the medium showed up beginning with the energy $0.1J$. The experiments concerned have revealed that optoacoustical concentrators have bright prospects for obtaining intense acoustical fields and investigating nonlinear acoustical interactions.

References
Study of optical properties of solids by photoacoustic spectroscopy

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A modulated beam of electromagnetic radiation falling upon a solid within a closed-volume gas cell, can generate an acoustic wave. This effect previously applied to gases (1,2) is now used as a new spectroscopic technique for solids. It is complementary to conventional optical absorption and reflection spectroscopy but offers some advantages over these techniques. A photoacoustic cell having cylindrical configuration has been designed to study the experimental factors which significantly influence the performance of the device such as the modulation frequency, nature of the filler gas, gas pressure, cell volume etc. Fig: 1 is a schematic diagram indicating the construction of the cell. The usefulness of this technique has been demonstrated by the investigation of a variety of solid samples. Some preliminary studies are described concerning the measurement of optical parameters e.g. optical absorption coefficient of different solids using the spectrophone technique.

References
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Fig: 1 Photoacoustic cell
A Top lid; B Infra-red transmission window; C Main body of the cell; D Specimen holder; E Gasket; F Microphone with preamplifier; G Backing; H Needle-valve (not shown)
INTRODUCTION - Laser and laser systems have been used to initiate vibration by laser light momentum [1], and to accelerate metal foil by front surface ablation [2]. An area which has not been well investigated is that of generating vibrations in a structure by front surface material vaporization due to the absorption of impulses of laser light on a target material, e.g. lead, located on the surface of the structure. The momentum of the vaporized material sets the structure into vibration. This technique has many possible uses, e.g. multiple point impulsive loadings of a structure, this being accomplished through the use of beam splitters and multiple targets on the structure. The purpose of the experiments reported here was to obtain order of magnitude data regarding the response of a beam to the above type of loading, these results are reported below.

EXPERIMENT - A cantilevered steel beam, 310 mm long x 12 mm high x 0.93 mm thick, had a lead target attached at the free end of the beam. The target was exposed to a one microsecond long pulse of laser light from a 15 Joule pulsed CO₂ laser, wavelength of 10.6 μm. A Vibra Metrics model M1000 accelerometer was also located near the free end of the beam to measure the vibration of the beam created by the front surface vaporization of the target. The acceleration signal was then displayed on a calibrated UV recorder, and the amplitudes of the first two modes were obtained from this display. Electric discharge noise (electrical field) was apparent for the first several cycles for some data plots so that the given results were obtained from the time histories for later cycles by assuming the damping to be small.

RESULTS - Observed fundamental frequencies and amplitudes of vibration were obtained directly from the time plots. The calculation of the energy and impulse stored in each mode was obtained by using "mass loaded cantiler beam theory" and obtaining the unknown coefficients in the series solution, by assuming an impulsive point loading and equating the maximum velocity to that measured by the accelerometer for each mode. A result is given below and other data will be presented at the conference.

- Fundamental frequency observed - 7 Hz, Predicted - 6.4 Hz.
- Measured maximum velocity first mode - 0.19 m/s.
- Impulse stored in first mode - 3.02 x 10⁻³ N·s.
- Energy stored in first mode - 2.1 x 10⁻⁴ Joules.
- Efficiency - First mode energy/Laser light energy - 1.6 x 10⁻³ % (assuming 90% absorption by lead).

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ACKNOWLEDGEMENT - I wish to thank Messrs. D. Hannan, R. Tobin, and L. Mathias for their assistance.
An ultra-wideband guided wave acoustooptic mode converter utilizing tilted-finger chirp transducer in doubly confined structure

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Acoustooptic (AO) deflectors and mode converters constitute a key active element in future integrated optical signal processing and communication systems. In particular, the high-bit rate polarization multiplexer which requires a mode converter (MC) with large conversion bandwidth is very important. A 30MHz bandwidth device has been demonstrated (1), however the interdigital transducer (IDT) used to generate surface acoustic waves (SAW) in this conjunction has equal width electrodes. Its narrow acoustic bandwidth limits that of the MC.

In this paper, analysis of an AO mode conversion using collinear interaction between laterally confined optical guided waves and SAW is given. This structure is illustrated in Fig.1. The SAW zig-zag propagation in acoustic waveguide where Ti-diffused layers employed as fast region, is described. A different type of the tilted-finger chirp transducer (TFCT) as shown in Fig.2 is incorporated to generate large bandwidth SAW. The acoustic frequency and mode conversion phase matching conditions are considered. It is shown that a 500MHz half-power conversion bandwidth with SAW centred at 1GHz can be obtained.

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(1) L.N. Binh. "Collinear Acoustooptic Mode Converter: Planar and Doubly Confined Structures". To be presented in this Congress.
COLLINEAR ACOUSTO-OPTIC MODE CONVERTER : PLANAR AND DOUBLY CONFINED STRUCTURE

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An analysis is briefly given of the optimisation of the mode conversion of guided optical waves resulting from interaction with collinearly travelling surface acoustic waves of Raleigh type on a planar Ti-diffused LiNbO₃ waveguide. The effect of electro-optic and acousto-optic index fluctuations together with the phase matching condition and interaction lengths are considered. A number of devices have been fabricated and tested to demonstrate optimisation of mode conversion in planar structure. A 50% conversion efficiency is obtained for TE₁-TM₁ conversion in LiNbO₃ with an acoustic input power of 10mW. These planar type structures result in a very narrow bandwidth of less than 1MHz.

Bandwith limitations of the above structure have been overcome by a method in which the optical and surface acoustic waves have been interacted in a doubly confined waveguide structure (1). This arrangement is shown in Fig.1. The conversion half-power bandwidth is increased to about 30MHz with a 300 Å Ti-film diffused 12.5 m wide optical channel and a 1000 Å Ti-film diffused layer in the region for guiding surface acoustic waves. The required acoustic input power for a 50% mode conversion of the fundamental optical guided mode is reduced to about 1.2mW.

Figure 1. Schematic diagram of mode converter using doubly confined structure

(1) L.n Binh and J. Livingstone
A wideband acousto-optic TE-TM mode converter.
NOVEL TRANSUDER DESIGN FOR THE DETECTION OF NON-BONDS IN ADHESIVE BONDED MULTIPLE-LAYERS BY THE ACOUSTIC IMPEDANCE METHOD

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In the usual acoustic impedance method of detecting non-bonds, two adhesive bonded thin layers act as the load of a specially designed transducer. A variation of the state of binding is reflected as a change in the acoustic impedance of the composite material and hence as a change in the load impedance of the transducer. By measuring the resulting change of certain characteristic behaviors of the transducer, the state of binding can be specified. We have also demonstrated that the Fokker bond tester can be considered as another version of the acoustic impedance method.

In order to detect non-bonds in a composite material consisting of more than two adhesive-bonded layers, a novel transducer is designed on the basis of our previous works and is shown in the accompanying figure. Theoretical and experimental results show that the transducer can be used to detect non-bonds between any two layers of such composite material.

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2 Li Ming-Xian, Physics, 2 (1972), 79.
3 Ying Chong-fu and Li Ming-Xian, Acta Acoustica, 1 (1979), 45.
The phonon-electron interactions in the crystal lattices of copper have been studied along with the angular forces. The phonon dispersion curves have been drawn in [100], [110] and [111] directions and compared with the experimental points. The agreement is found to be very good. \( \Theta_p - T \) curve for temperatures above 80K also shows very good agreement.
ACOUSTICAL PHASE MEMORY IN FERMION SYSTEMS

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Up to now, beams of fermions (electrons, protons, neutrons and \( \alpha \)-particles) have been of no use for quantum counters and experiments in quantum acoustics. Our aim is to point out some new possibilities in this field of physics. 1) Quantum counters must work without any perturbation. So the temperature of the counter must be \( kT < \hbar \omega \) where the quantum to be detected is \( \hbar \omega \). Quantum interference must be used in the detection scheme to exploit all the power of quantum mechanics. Interference of a fermion with itself may be used. The periodicity is \( \check{\omega} \). If a beam of fermion passes a target of polarized nuclei there will be a Fermi contact nuclei–nuclei interaction depending on the distance between the nuclei. So sound perturbation will cause time dependent Fermi interactions. Polarized spin-systems may be of use for counting of fermions in a beam. The interaction may be of resonant nature. Sound waves modulate the distance between the beam of neutrons and protons. Resonant transitions between the spin states of the neutron will be induced. The effect is the acoustic nuclear resonance of neutrons. The transition probability is as high as in the case of acoustical nuclear resonance in antiferromagnets. As a result the polarization of the beam will change.

The second effect is the spin echo of neutrons. To induce this effect we need coherent sound excitation. A superpositional state of the neutron spin will be formed. But as the time goes the coherence of the spins will be destroyed by inhomogeneous magnetic fields. The second sound excitation will radically change the situation and after a time interval the coherence will be restored.

Fermion beams may be put into a superpositional state. One of the states may be disturbed by means of an acoustical deformation. An interference pattern will reveal this effect. But echo formation can destroy the effect. As is known, gravitational waves can generate deformations in crystals. We show that neutron beams are suited to detect relative deformations of the order of \( 10^{-18} \).

A theory of the acoustical excitation resonance effects on neutron beams is developed. Phase memory effects are discussed. The result is that phonon neutron technique has very high sensitivity in the energy and time domain. A coherent preparation of the target will stimulate the effect of neutron superscattering. To get the space coherence and energy magnetic fields and terasound pulses must be used. Numerical calculations show the reality of such experiments.

Some new possibilities arise by the exitation of positronium atoms by means of hyper and terasound. The detection is possible by means of lifetime measuring and correlation experiments.
ULTRASONIC MEASUREMENTS OF STRUCTURAL RELAXATION IN AQUEOUS ELECTROLYTE SOLUTIONS IN THE RANGE 0.01...5 GHz At -50...+20°C.
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We measured ultrasonic absorption and sound velocity in aqueous solutions of electrolytes\(^1\) (esp. alkali halides) and could show that structural relaxation expected for pure water at frequencies near 80 GHz is shifted down to the measuring range (0.1...5 GHz) by increasing the concentration of ions as indicated\(^2\) and lowering the temperature\(^3\). Results for a 15.3 m LiCl-solution are shown in the figure for various temperatures. At low frequencies the absorption is proportional to shear viscosity; it is increasing with decreasing temperature stronger than given by a constant activation energy. At high frequencies and lower temperatures the values of reduced absorption \(\alpha/\nu^2\) are below the classical Stokes value, indicating that both volume and shear viscosity have undergone relaxation. The Kramers-Kronig-relations applied for complex propagation of sound waves predict the accompanying dispersion of sound velocity. Though we are able to fit the data by various two-parameter relaxation distributions, we prefer the Montrose-Litovitz-theory for fitting\(^4\). The width of the distribution is temperature-independent, if a finite limiting value \(\alpha/\nu^2\) \(\omega\to\infty\) is properly taken into account.

1. K. Tamm, 6\textsuperscript{th} Int. Congress on Acoustics, Tokyo (1968)
STUDY OF ULTRASONIC VELOCITY AND OTHER RELATED PARAMETERS IN
CHOLESSTEROL-1-HEXAGONAL MIXED LIQUID CRYSTALS

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Measurements of ultrasonic velocity and specific volume are carried out
at four concentrations of cholesterol + 1-hexadecanol mixture, besides
the non-mesomorphic pur compounds. For the pure compounds, the
parameters specific volume ($v$), ultrasonic velocity ($V$) and adiabatic
compressibility ($K_{ad}$) varied linearly with temperature. In the mixture
at all the concentrations there was observed a measurable change in
specific volume in the vicinity of the phase transition, whereas the
adiabatic compressibility showed a sudden jump at transition. The dip
in ultrasonic velocity and the jump in adiabatic compressibility
increased with increasing concentration of cholesterol in the mixture.
The parameters molar sound velocity and molar compressibility are also
estimated. The temperature coefficient of volume expansion and adiabatic
compressibility are estimated which are found to attain high values near
transition showing the presence of pre-transitional effects. This
behaviour is interpreted on the basis of de-Gennes theory, generalized
by Bendler. The critical exponents for volume expansion and temperature
coefficient and adiabatic compressibility are estimated both in the
isotropic and smectic phases.
INVESTIGATION OF ULTRASONIC BEHAVIOUR IN CHOLESTERYL MYRISTATE AND ETHYL P-AMINO-BENZOATE LIQUID CRYSTALLINE MIXTURES

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Investigation of ultrasonic velocity, specific volume and other related parameters is carried out in liquid crystalline mixtures of cholesteryl myristate and ethyl P-aminobenzoate over a wide temperature range in seven different concentrations. Anomalies are observed in all the parameters, at the isotropic-cholesteryl and cholesteryl-smectic A phase transitions. The magnitudes of the velocity dip, the specific volume change and the adiabatic compressibility jump are found to be decreasing with increasing concentration of ethyl P-aminobenzoate. The interesting aspect of the study is that the smectic A-Cholesteric transition is found to change from first order to the second order at a particular concentration of isotropic compound EPAB. It is found that smectic A cholesteric transition becomes second order with the interaction parameter \( \chi = \frac{T_{NA}}{T_{NI}} = 0.84 \) which is in good agreement with Lee's theory.

It is also observed that \( \frac{\Delta \rho_{ad}}{\rho_{ad}} \) bears linear relation to the interaction parameter \( \chi \) and also to the concentration of cholesteryl myristate.
STUDY OF VIBRATIONAL RELAXATION IN CHLOROFORM BY HIGH-RESOLUTION BRAGG REFLECTION TECHNIQUE

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INTRODUCTION

Diagram of the vibrational energy levels suggests two relaxation regions are expected in liquid chloroform, although previous studies have found only one. To settle this open question, we made the measurement of velocity dispersion over a range from 3 MHz to 6 GHz using high-resolution Bragg reflection and Brillouin scattering. A cooperative work of these two techniques gives the best way available to investigate the ultrasonic and hypersonic properties in liquids.

HIGH-RESOLUTION BRAGG REFLECTION

In this particular work, we used a modified electronic system in which the receiver is automatically tuned to the beat signal at every frequency. Increase of the stability in the received signal has enabled to use a narrower bandwidth and therefore improved the s/n ratio by a factor of ten.

DOUBLE RELAXATION PROCESS

The observed results shown in Fig.1 cannot be described by any curve of single relaxation. We determined therefore the best-fitted curve of double relaxation represented by solid lines.

Analogously to the double relaxation in dichloromethane, we assigned all the vibrational modes belonging to group I in Fig.2 to the relaxation at lower frequency and those in group II, to the relaxation at higher frequency. Under this assumption, we calculated the contribution of each group to the vibrational specific heat and compared the result with the relaxing specific heat. Agreement between the experimental and the theoretical values confirms validity of the hypothesis proposed here.

ULTRASONIC ABSORPTION IN THE CRITICAL SYSTEM O-CHLOROANILINE–n-HEXANE

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The absorption coefficient values for ultrasound ($\alpha/r^2$) in O-chloroaniline–n-hexane system are presented. The measurements were made over the full range of concentrations at 15°C, 17°C, 19°C, 21°C and 23°C and in the frequency range 3 – 15 MHz. The absorption is found to be a strong function of concentration and rises to a maximum at $\cong 0.54$ mole fraction of n-hexane. The anomalous absorption observed near the critical concentration has been explained in the light of Fixman’s theory. Using the approximate thermodynamic data the Debye short-range correlation length and the friction constant have been calculated. The composition dependence of absorption has been explained using Flory-Huggins approximation together with Fixman Theory.
ULTRASONIC INVESTIGATION OF THE AQUEOUS SOLUTION OF
POLYACRYLAMIDE


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The shear viscosity of concentrated aqueous solution of polyacrylamide is so high that, according to the classical theory of the acoustic absorption, ultrasonic vibration cannot propagate in it at all. However our experiment with ultrasonic wave of frequency 11.4 MHz finds that it has ultrasonic absorption with the same order of magnitude as water does. This paper provides a qualitative explanation about the great difference between theory and experiment from the molecular structural feature of this kind of solution.

The measurements of the ultrasonic absorption of acrylamide aqueous solution with various concentrations have been done, too. By comparing the experimental results of the two solutions it can be considered that excessive ultrasonic absorption of the polyacrylamide solution comes from the characteristic behaviors of the solute macromolecules themselves.

The ultrasonic velocity and density of these two solutions are also investigated. From them the concentration curves of compressibilities has been calculated and it is found that some atomic groups in the solute molecules exhibit the same hydrate in the aqueous solution independent of whether they are in polymer or in small molecular substance.
LIGHT SCATTERING STUDIES OF THE ULTRASONIC RELAXATION IN IONIC SOLUTIONS IN THE FREQUENCY RANGE 1-30 GHZ

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The high-frequency structural fluctuations in molecular liquids are usually beyond the reach of ultrasonic measuring techniques. However, these fluctuations are slowed down in the vicinity of ions of sufficiently high charge density.\textsuperscript{1,2} Lowering the temperature reduces the relaxation frequency still further. The combination of these two effects brings the dispersion region into the frequency range which is accessible with Brillouin light scattering techniques. By varying the scattering angle and the wavelength of the laser light a wide frequency range can be covered. We have measured the ultrasonic absorption and the sound velocity in aqueous solutions of alkali halides in the temperature range 100-300K for various concentrations. The results for an 8m solution of LiCl are shown in the figure for four different temperatures. As can be seen, for all temperatures shown their is a dispersion of the sound velocity and an accompanying variation of the reduced ultrasonic absorption $a/\nu^2$ ($\nu =$ frequency) indicating the presence of relaxation processes in this frequency range. The relaxation strength increases with decreasing temperature while the mean frequency of the relaxation processes shifts to lower frequencies.

CORRELATION BETWEEN HALL'S THEORY AND EYRING'S HOLE THEORY

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Ultrasonic studies have been made in associated and unassociated liquids as a function of temperature and pressure. The absorption measurements were made as a function of temperature using a pulse technique. The accuracy of the measurements are ±2 percent and are in good agreement with the available results.

Hall's theory of structural relaxation and Eyring's hole theory have been used to explain the excess ultrasonic absorption in these liquids. Hall's theory has thus far been quite successful in explaining the absorption in associated liquids. Eyring's concept of holes has been found very useful for explaining the excess absorption in unassociated liquids.

According to the two state model proposed by Hall, water is considered to consist of two states, an open packed state and a relatively closed packed one. Eyring visualises the two states as one containing holes and the other holeless. Thus to start with, both the theories assume a two state model.

Hall's theory has also been used in explaining the excess absorption in unassociated liquids and Eyring's hole theory has been used to explain the excess absorption in associated liquids. The results are encouraging.

Considering that both the theories, Hall's theory of structural relaxation and Eyring's hole theory, can be used to explain the excess absorption in associated and unassociated liquids, an attempt has been made to correlate the two. The results are encouraging. The correlation has been tested for few liquids.
The dynamics of acoustic cavitation bubbles is investigated in order to obtain the values of the main parameters which characterize the strength of the various collapses. To this end, the classical Kirkwood-Bethe-Gilmore equation is solved, with the help of a numerical integration, in the case of a single spherical bubble surrounded by an infinite compressible liquid. The minimum value of the bubble radius, which is reached at the end of the collapse, is used to calculate the maximum value of the pressure inside the bubble. Taking into account the distributed radii of cavitation nuclei, the main distribution features of the pressure pulses produced by the various collapses are deduced. In particular, the influence of the ultrasonic pressure amplitude is indicated. When this increases, the pressure pulse-height begins to increase then decreases after a maximum value has been reached. Then, a slight further increase of the ultrasonic pressure amplitude induces a sharp variation of the pressure pulse-height of which is now higher than its first maximal value.

The theoretical results concerning the distribution of the pressure pulse-heights are then compared with the actual histogram of the experimental pressure pulse-heights. This histogram is obtained by means of a miniature pressure probe connected to a multichannel analyser. The probe is plunged into the cavitation field produced by an ultrasonic transducer. About $2 \times 10^5$ pulses are analysed during each run. For low level ultrasonic power, the histogram takes the form of a peak. When the power increases regularly, a second peak appears which is due to pressure pulses higher than those which produced the first peak. These experimental observations agree with the previous theoretical analysis.
A study of the cellular precipitation process in Pb rich Pb–Sn alloys containing 6.1 and 12.1 wt. percentages of Sn has been carried out using the composite oscillator technique. From the longitudinal and shear ultrasonic velocities and densities, a set of constants such as Young’s modulus, rigidity modulus, bulk modulus, Poisson’s ratio, average sound velocity and Debye temperature are estimated. The above parameters and internal friction are found to exhibit interesting variations during the precipitation process. The results are interpreted in terms of the kinetics of the cellular precipitation occurring in the alloys system.
FORMATION DE COACERVATS A PARTIR DE MOLECULES SIMPLES PAR ACTION DES ULTRASONS.

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PRUDHOMME

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On sait que les ultrasons peuvent donner naissance en milieu aqueux à de nombreux composés organiques dont la nature dépend des gaz dissous: ammoniac, hydroxylamine, aldéhyde formique, acide cyanhydrique, acides carboxyliques, hydrocarbures saturés et insaturés... Ces substances peuvent à leur tour synthétiser sous l'action des ultrasons, des molécules dites d'intérêt biologique "car on les retrouve dans la composition de la matière vivante. Beaucoup de ces réactions peuvent aussi se produire sous l'action des décharges électriques, des rayonnements lumineux et des radiations ionisantes. La formation de coacervats (globules, sphères, microsphères) a été aussi observée sous l'influence des rayonnements lumineux à partir de solutions aqueuses de formaldéhyde et de sel ammoniacaux. Parmi ceux-ci, il faut citer les globules obtenus par BAHADUR en faisant réagir, à la lumière, un mélange aqueux d'aldéhyde formique, de phosphate diammonique et de différents sels. Ces globules microscopiques de couleur bleue, dont la frontière externe présente une structure en double couche, s'associent en colonies et peuvent bourgeonner en donnant naissance à une seconde génération de globules.

Nous avons voulu dans ce travail savoir si l'énergie ultrasonore pouvait, de même que l'énergie lumineuse, synthétiser ces coacervats. Pour cela, nous avons soumis le mélange de BAHADUR à deux fréquences ultrasonores respectivement: 80 KHz/10W/cm² et 20 KHz/8W/cm². L'expérience a été faite à l'obscurité et en présence d'argon comme gaz dissous. Quelque soit la fréquence utilisée nous avons obtenu, après dix heures de traitement, des globules très comparables à ceux formés avec l'énergie lumineuse. Ces globules sont colorés bleus à 800 KHz, gris-verts à 20 KHz. L'analyse chromatographique montre comme avec les globules synthétisés sous l'action de la lumière, l'existence d'acides: glycocollé, α et β-alanine, méthyl-alanine, acide glutamique. D'autre part, le rapport carbone-azote est très voisin de celui des microsphères de BAHADUR. Les globules obtenus par ultrasonation présentent aussi une frontière externe dont la structure est en double couche. Ces microsphères une fois formées et en dehors du champ ultrasonore évoluent dans le temps: elles peuvent bourgeonner et donner naissance à d'autres globules .

En conclusion, de même que les autres formes d'énergie: décharges électriques lumineuses, rayonnements ionisants, les ultrasons sont capables de synthétiser certains coacervats. A l'époque de la terre primitive, de nombreux phénomènes naturels: tonnerre, séismes, raz de marée, chutes d'eau... ont produit des vibrations acoustiques de grande intensité. Ces apports énergétiques, quoique peut-être moindres que ceux des décharges électriques et des rayonnements, ont vraisemblablement joué un rôle non négligeable dans l'apparition des composés primordiaux de la matière vivante.
ULTRASONIC ATOMIZATION OF LIQUIDS

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INTRODUCTION

The atomization of liquids by ultrasound has been investigated in detail for a long time. As the basic research for the spray drying, this experiment was to study the effects of vibration amplitude, frequency, and viscosity of solution.

EXPERIMENTAL METHODS

Piezo-electric transducers were used at the frequency of 20 kHz and 40 kHz with the exponential horn. A titanium nut was connected on the tip of horn with varying the vibration amplitude. The solution was introduced on the nut and the viscosity of solution was changed. The atomized particles were collected on the surface of oil and 600 particles were counted under the microscope to measure the particle size distribution.

RESULT AND DISCUSSION

Fig. 1 shows the arithmetic average particle size when the flow rate was varied from 20 ml/min. to 80 ml/min. It can be seen that the smaller the flow rate, the smaller the average particle size was obtained at the frequency of 20 kHz with the tip amplitude of 23μm. The same experiment was operated at the frequency of 40 kHz and the average particle size showed almost the same trend curve as shown in Fig. 1.

Fig. 2 shows the effect of vibration amplitude when the milk solution was atomized. It can be seen that the average particle size was measured to be smaller with increasing the vibration amplitude.

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**Fig. 1** Flow rate vs. arithmetic average particle size.

**Fig. 2** Vibration amplitude vs. arithmetic average particle size.
ULTRASONIC WELDING OF PLASTICS

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INTRODUCTION

In ultrasonic welding of thermo-plastic materials, it was found that the stiffness of the anvil, where workpiece is placed, plays important role. High-quality welding was performed with small-amplitude vibrations in a short time when the stiffness was set appropriately.

EXPERIMENTAL PROCEDURE

Figure 1 shows the anvil with a steel elastic ring which provides arbitrary stiffness. The stiffness was controlled by changing the inner diameter of the ring, whose outer diameter was 8 cm. A conventional anvil, which was fixed directly on the base without the use of the ring, was also prepared for comparison. Polyethylene cylindrical rods of 1 cm diameter and 3 cm length were used as workpiece. In the experiment, each two pieces was welded at the vibration frequency of 19.7 kHz under the static pressure of 40 kgf/cm². The correlation between the time spent for welding and the vibration displacement (r.m.s.) of the tool was measured with the stiffness as parameter.

RESULTS AND CONCLUSION

Figure 2 shows experimental results obtained. The time spent for welding was shortest when the stiffness was set to 0.16×10⁶ N/m, whereas it took longest when the conventional anvil, whose stiffness was regarded as infinite, was employed.

It was concluded that the stiffness of the anvil, which is set appropriately according to workpiece, is effective in decreasing both the time and vibration amplitude required for high-quality ultrasonic welding.
AN EXPERIMENTAL MODEL OF PILOT PLANT FOR ULTRASONIC DEPURATION OF CARBON BLACK SMOKE

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INTRODUCTION
The agglomeration of solid particles suspended in a gas is one of the many interesting effects produced by high intensity ultrasound waves. This phenomena, that was initially studied by Brandt, Freud and Hiedemann in 1937, has recently taken a renewed interest in connection with environmental problems. Specifically the sonic treatment of fine-particle emission from industrial sources appears as a problem of present interest in many countries. During the last years we have developed different researches in this field (1,2) using a new powerful ultrasound generator. These works have conducted to the design and performance of an experimental model of pilot plant for depuration of carbon smoke from hydrocarbon combustion. In this paper we describe the characteristics and performances of this prototype.

EXPERIMENTAL
The carbon black smoke is an aerosol that appears in all these processes in which hydrocarbons are burned. Therefore it is one of the most important from the industrial point of view. On the other hand the size of the particles is generally less than about 3 microns which corresponds to the particle range adequate for sonic treatment.

The apparatus is schematically shown in Fig. 1. The carbon black smoke is made using a commercial fuel-oil burner 1. The smoke concentration can be varied by varying the fuel-air ratio. The smoke temperature and humidity can be adjusted by water in the cooling tower 2. The acoustic chamber 4 is a resonant cylindrical metallic tube of 2.6 m long and 0.23 inside diameter. The sonic generator is constituted by a high-directional and high-efficient ultrasonic transducer for 20 kHz (3) driven by an electronic generator with a motional feedback system. The degree of coagulation is measured in the measuring chamber 3 and 6 by the change in light transmission of the smoke and taking samples of the particles to analyse them by electron microscopy. A cyclone system 5 and an exhaust fan 7 complete the apparatus.

The high collection efficiencies obtained with moderate rates of energy consumption demonstrate the feasibility of our ultrasonic system for industrial applications. The experimental set-up here presented represents a new test facility to obtain commercial design data.

ULTRASONIC VIBRATION PRESS OF METAL POWDER -- ON THE EFFECTS OF VIBRATION PHASE DIFFERENCE BETWEEN VIBRATION PUNCHES --

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Ultrasonic vibration press of hard metal powder (WC-Co alloy powder) is studied. The upper and lower compacting punches are driven by 20 kHz upper and lower vibrating systems independently. The vibration phase difference between upper and lower compacting vibration punches is varied 0--180°.

Fig.1 shows the arrangements of vibration systems and compacting specimen.

The upper and lower vibration systems are composed of 8 PZT volt-clamped Langvin type longitudinal transducers (40 mm diameter), radial-longitudinal vibration direction converters, and stepped horns to support the vibration systems at their nodal positions. The vibration punches are 1/2 wave length longitudinal resonance bars which have vibration loop at their compacting tips. The upper and lower vibration systems are driven by 1 KW power amplifier independently. The upper and lower vibration amplitude of punches used is 0--6 microns (peak to zero value). The vibration phase difference between upper and lower vibration punches is controlled by the phase shifter connected between a main oscillator and a power amplifier.

A hydraulic static pressure source and a press frame unit are used to compact and press the specimen powder. The static pressure at specimens is varied 0--2500 kg/cm, and measured by a strain gauge unit set at the lower part of vibration press unit.

The variations of specimens height, density and dimensions changes are measured at various compacting conditions.

FIG.1 ARRANGEMENTS OF VIBRATION SYSTEMS
H. UNDERWATER SOUND
SOUND PROPAGATION IN LIQUID WITH PHASE TRANSFORMATIONS

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It is shown that attenuation and dispersion of sound in liquid containing inhomogeneities in the form of new-phase nuclei (vapor bubbles or solid nuclei) exceed essentially appropriate values for the liquid containing the same inhomogeneities but under conditions of absence of phase transformations on their surfaces. As it is known [1], the wave number $k_n$ of the sound propagating through the liquid containing inhomogeneities is defined as $k_n^2 = k^2 + 4\pi n \zeta$, where $k = \omega/c$ is wave number in pure liquid, $n$ is the number of inhomogeneities of radius $R$ in a unit liquid volume, and $\zeta$ is the scattering amplitude of the wave which is defined from the solution of the problem on linear spherical oscillations of the nucleus in the sound field [2-3]. Sound attenuation $\alpha = \text{Im} k_n$ and sound dispersion $\Delta c/c = (\alpha/c)/c = \text{Re} [(k-k_n)/k_n]$, may be written as

$$\alpha = \frac{c}{2} \varepsilon \omega (\rho' - \rho) \text{Im} \frac{K(R, \omega)}{Q(R, \omega)},$$

$$\Delta c = \frac{c}{2} \varepsilon \left[ 1 + c^2 (\rho' - \rho) \text{Re} \left( \frac{K(R, \omega)}{Q(R, \omega)} \left( 1 - \frac{\rho'}{(\rho' - \rho) K(R, \omega)} \right) \right) \right],$$

where $\varepsilon = 4\pi n R^3/3$ is volume concentration of the new phase in liquid, $\rho'$ and $\rho$ is new phase and liquid density and $\zeta$ is the liquid compressibility. Function $K(R, \omega)$is the own compressibility of new-phase nuclei which take account of additional change of nucleus volume caused by surface processes of phase transformations [3]. Function $Q(R, \omega)$takes account of the influence of inertial and viscosity forces and surface tension upon nucleus oscillations.

To illustrate the abovementioned we would like to point out that in the water containing solid phase nuclei with $\varepsilon = 10^{-5}$, $R = 8 \times 10^{-3}$ cm, at frequency $2 \times 10^3$ c.p.s. sound attenuation is $\alpha = 5 \times 10^{-6}$ cm$^{-1}$. If, however, phase transformations are absent, then under the same conditions $\alpha = 4 \times 10^{-9}$ cm$^{-1}$.

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OPTIMUM FREQUENCY IN DUCTED PROPAGATION

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There is always an optimum frequency associated with ducted propagation in the ocean. Three general types of sound-propagation ducts can be identified: 1) surface ducts, 2) deep sound channels, and 3) bottom-limited ducts. Item three is the most general ocean environment and includes the former two. The optimum frequency is a result of competing propagation and attenuation mechanisms at high and low frequencies. In the high-frequency region, there is simply increasing volume and scattering attenuation with frequency. The lower frequency portion is more complicated. With increasing wavelength, the efficiency of the duct to confine the sound field decreases. Hence propagation and attenuation mechanisms outside the duct (e.g., in the sea floor) affect the intensity of the sound in the duct. The simplest example is compressional attenuation in the ocean bottom. Even though this attenuation decreases with frequency, the overall attenuation of sound in the water column will increase with decreasing frequency because the dominant effect is the greater penetration into the bottom with decreasing frequency (increasing wavelength). Another important mechanism that affects the optimum frequency is the rigidity of the sea floor allowing for the existence of shear waves in the bottom. Losses in the water column will increase with increasing shear speed and decreasing frequency. Hence, the optimum frequency is increased with increasing shear speed. Shear speeds less than, say 200 m/s, have a negligible effect. The optimum frequency is also affected by the sound-speed profile in the water and the bottom, which is the cause of the duct in items one and two and in item three essentially regulates the sound interaction with the bottom. The presence of an optimum propagation frequency in the ocean is demonstrated with both experimental and theoretical data. A parametric study using a propagation module based on wave theory (1,2) has been made to determine the optimum frequency for some typical ocean environments. An example displayed here shows contoured propagation loss versus frequency and range for a given source/receiver combination. In this particular case the optimum frequency is seen to be around 350 Hz. Comparison between theoretical and experimental results indicate that we now have a fairly good idea of the competing mechanisms that determine the optimum frequency of a duct.

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(Both published by SACLANT ASW Research Centre, La Spezia, Italy)
INTRODUCTION AND APPROACH

This theoretical paper compares convergence-zone and shadow-zone propagation losses as calculated by ray theory with those calculated by normal mode theory. It presents initial results of a comprehensive study to establish the low-frequency limits to the validity of various corrections to geometric ray theory using normal mode theory as a control.

The input environment for mode theory is a 4-layer sound-speed profile where each layer is of the form \( C^{-2} = a + b z \). The first and second layers model the deep-water channel. In the third layer \( a_3 \) and \( b_3 \) are complex and are chosen such that \( C \) and \( \Re(\frac{dC}{dz}) \) are continuous at the layer 2-3 interface. The fourth layer is a half space with \( \Re b_4 \) positive. The third layer introduces attenuation in the form of complex sound speeds to limit the number of contributing modes and it matches conditions in order to minimize reflections from the interface. (Mode computations made without this third layer in the model depart markedly from ray theory because of reflections from the slope discontinuity at the channel bottom). The ray theory approach only treats the first and second layers. The amplitude and phase of the field, associated with each ray theory caustic, are evaluated by uniform asymptotics. (In the shadow zones this procedure uses rays with complex parameters). These fields are then combined to obtain the resultant propagation loss.

RESULTS

A 200m source and 500m receiver depth produced an initial caustic at 46.1 km and interior caustics at 54.3, 54.9, 58.0, and 61.5 km. The range of the ray which grazes the ocean surface is 56.0 km. The ray results begin to deteriorate before 56 km because the surface treatment is not rigorous. Propagation losses of ray and mode theory agreed to within 0.5 dB over the range intervals 39.8 to 51.2, 38.9 to 54.0, and 45.2 to 50.6 km for respective frequencies of 100, 25, and 10 Hz. Differences in propagation loss between ray and mode theory are not random but often contain bias and modulation components. The bias component at 100, 25, and 5 Hz is 0.2, 0.3, and -2.5 dB respectively. At 100 Hz at 41 km the modulation is ± 0.2 dB. The omission of an arrival, which has 32.7 dB more loss than the included arrivals, is sufficient to produce such a modulation. At frequencies of 10 Hz and below there are indications of range offsets between patterns. At 53 km the resultant propagation loss for 25 Hz was 10.8 dB less than that associated with the initial caustic only. These and other results lead to the following conclusions: (1) The method of combining the fields associated with several caustics is valid. (2) Other ray theory corrections are necessary particularly at frequencies below 10 Hz. (3) Shadow zone fields of interior caustics can be important. (4) The sub-bottom model of mode theory must be chosen carefully to minimize interface reflections.
Surface-duct propagation in waters near Australia

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Since 1968, a series of over 30 medium-range acoustic surface-duct propagation experiments have been conducted in ocean areas near Australia. These measurements were conducted at ten separate localities whose latitudes range from about 5 degrees to 35 degrees south. The main purpose of the programme was to test various surface-duct propagation models over a wide range of environmental conditions, and if possible, to develop improvements to the existing models.

The experimental method was to suspend two small hydrophones (one within the mixed-layer and the other below the layer) from a stationary ship while another ship detonated shallow explosive charges at ranges up to 6 km. All of the experiments have been analysed at frequencies of 4, 8 and 16 kHz; and some analysis has been done at 1 and 2 kHz.

The propagation models that have been compared with the data are: normal-mode theory; an empirical model by Baker; both the original and a modified version of the AMOS empirical model; and the Helmholtz–Kirchhoff theory of scattering from a rough surface.

The average relative energy within the mixed-layer at 6 km range varies from -77 dB at 4 kHz to -91 dB at 16 kHz, with standard deviations of about 4 dB. The modified AMOS model predicts these averages to within 3 dB, whereas normal-mode theory (for a flat surface) over-estimates the average energies by 12 to 19 dB. The correlations between individual AMOS predictions and individual results are generally insignificant; although there is a significant negative correlation between energy at 8 kHz and surface roughness. The average at 16 kHz is about 7 dB less than spherical spreading minus absorption; the cause of this attenuation is unexplained.

The average relative energy below the mixed-layer (in the "shadow zone") at 6 km varies from -87 dB at 4 kHz to -94 dB at 16 kHz, with standard deviations of about 6 dB. The averages of the predictions of the original AMOS model are within 5 dB of the experimental averages, while normal-mode theory under-estimates the energy by 7 to 15 dB. The ratio of the sound energy in the shadow zone to the energy within the mixed-layer tends to steadily increase with both surface roughness and sound frequency; whereas the Helmholtz–Kirchhoff theory predicts that, for most of the experimental results, the ratio should be a constant.
Predicting Convergence Zone Formation in the Deep Ocean (Category H-124)
Naval Underwater Systems Center (New London Laboratory)

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Convergence zones are formed at discrete range intervals in the world provided that certain environmental conditions are satisfied. In order to evaluate the effect of these environmental features on the convergence zone path, the concept of a "depth excess" was developed. Depth excess is defined for a given location as the excess in water depth over that just required for a single ray from the surface to reach the convergence zone. By computing the average depth excess for a given month and location, it is possible to portray various ocean areas as a series of average depth excess contours. Then, by specifying a minimum acceptable depth excess for adequate zone formation, an examination of the monthly average depth excesses in various areas, along with the variability about the average, permits a realistic estimate of the reliability of useful convergence zone formation as a function of month and area.
THE FAST FIELD PROGRAM (FFP) FOR RANGE DEPENDENT MEDIA

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The archetype model of an ocean stratified in depth (range independent) leads to an integral (Hankel) transform representation of the underwater sound field of a simple acoustic source. For numerical work, the asymptotic (ray) method or a residue (normal mode) series have been widely used to approximate this transform. Both have mathematical limitations, and being eigenvalue problems, can present significant computational difficulties. Current computer technology now permits direct numerical evaluation of the transform.* This is accomplished using a Fast Fourier Transform (whence the title). This technique has been utilized over a wide range of the various parameters involved for the range independent problem since 1967.

The application of this technique to the range dependent problem where the total wavenumber is separable but otherwise an arbitrary function of range and depth will be discussed and the results for several examples presented.

Modulation of Coherent Underwater Sound by Internal Motions of the Sea

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(Lecture will be read in English)

Volume Reverberation in Waters near Australia

R.A.N. Research Laboratory

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Since 1966, measurements of acoustic volume backscattering have been made during 15 cruises in deep-ocean areas near Australia. The areas in which most of the work has been done include the Coral Sea (CS), the Java Trench (JT), the equatorial Indian Ocean near Sumatra (IOS), the Great Australian Bight (GAB), and the Tasman Sea near Sydney (TS). Nine of the cruises were confined to the Tasman Sea. The purpose of the programme has been to develop a comprehensive data bank on the levels of volume backscattering that occur in the ocean areas around Australia.

The experimental method has been to suspend a small omnidirectional hydrophone from a ship and to detonate about 5 small explosive charges within a period of 5 to 10 minutes. The reverberation is measured at a particular instant of time to give the volume backscattering strength (S) averaged between the surface and the depth that corresponds to the time of measurement (namely, 1000 m in the GAB, and 750 m in the other areas). The reverberation signals have been analysed at frequencies in third-octave steps from 2.5 to 20 kHz.

The average daytime values of S in the JT and GAB areas are similar to each other and range from -82 dB at 4 kHz to -73 dB at 16 kHz. In the CS the result at 4 kHz is only -95 dB; whereas in the IOS, the average at 16 kHz is -70 dB. In the TS, the average result at 4, 8, and 16 kHz is -75 dB in each case. The geographic variation of the results correlates well with independent estimates of the organic productivities of the five areas.

The average night-time values of S in the JT, GAB and TS areas are similar to each other, and their values at 4, 8 and 16 kHz are all close to -70 dB. In the CS the result at 4 kHz is only -80 dB, and in the IOS the result at 16 kHz is -63 dB.

The diurnal variations in S for the CS, JT and GAB areas are similar to each other; the variations at 4 kHz are large (10 to 15 dB), but decrease to 3 dB at 16 kHz. In the IOS, the variation increases from 0 at 4 kHz to 7 dB at 16 kHz; and in the TS the variation is 5 dB at each frequency.

Deep scattering layers are prevalent during daytime in each area except the CS. The resonance frequency is usually about 5 kHz and the typical depth of the layers is 500 to 700 m. In the GAB area however, the resonance frequency is 4 kHz and the typical depth is 800 to 1000 m.
Acoustic High Frequency Scattering by Elastic Cylinders
Naval Ship R and D Center, Annapolis, MD 21402

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The complete bistatic scattering amplitude of an elastic cylinder in a fluid is calculated numerically in terms of its component amplitudes, namely reflected waves, refracted waves, and surface waves, in order to appraise their relative importance for various angular and frequency regions; in addition, the summed total amplitudes are evaluated for values of ka from 25 to 1000 (where k is the wave number in the fluid of the incident plane wave, and a the cylinder radius). All the different surface wave modes (i.e. Franz, Rayleigh, Stoneley, and Whispering-Gallery type) have also been obtained separately.

The individual wave amplitudes and their relative contribution to the total scattering amplitude will be shown as a function of the azimuthal scattering angle \( \phi \) for various values of ka. The summed scattering amplitude is displayed similarly; it is compared with the results of previous normal-mode calculations at low values of ka, and we show that at large values of ka, the geometrical limit is not reached. In the surface wave amplitudes, we demonstrate the existence of a series of narrow resonances corresponding to the excitation of eigenvibrations of the elastic cylinder.

All the above-mentioned individual wave amplitudes have been obtained by us from the normal-mode series, transforming the latter using the Watson transformation in which pole contributions furnish the surface waves, and saddle point contribution the geometric waves.

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SOUND FIELD OF BOTTOM SCATTERING

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Usually, the assumption of secondary sources lying on the sea bed and possessing definite directional character is taken for the analysis of bottom scattering. But in cases when transmitter and receiver are located at different depths, the results of such analysis contradict principle of reciprocity.

In the paper the sea bottom is treated as a plane boundary with random impedance, and the scattering field of point source is given by wave theory.

Discussion of results is given for special case of shallow water bottom reverberation.
Underwater Sound Reflection from Density Gradients

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The substantial density contrast observed in an estuarine pycnocline has been used in an examination of the mechanisms by which density gradients and steps in the water column reflect acoustic signals. Attention is directed to non-turbulent interfaces and two contributing reflection mechanisms are examined. The graded interface associated with molecular diffusion across an initially sharp fresh/salt water junction gives rise to a time-dependent reflectivity. The magnitude of this is compared with the backscattered signal expected from various concentrations of zooplankton, using recently reported estimates of zooplanktonic target strengths.

It is concluded from the results of a field experiment undertaken at the Moore River, Western Australia, that the 200 kHz acoustic echo observed at this site from the pycnocline associated with a salt wedge formation is due primarily to the graded interface mechanism. The effect of varying sound frequency on this conclusion is discussed.
The Use of Probability Density Function Estimators to Determine in-situ Acoustic Target Strength Values in Marine Acoustics.

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Peterson, Clay and Brandt (J. Acoust. Soc. Am. 60, 618-622 (1976)) have proposed a signal processing methodology which makes use of the PDF realisation of an ensemble of echo signals. This is used to account for several stochastic parameters in echo formation, and thus to arrive at an estimate of the acoustic target strengths of fish in the water column, which is independent of fish orientation and position within the sounder beam.

It is necessarily assumed that all echoes in the ensemble arise out of single scattering from mono-specific, single sized fish, and that echo fluctuations due to varying fish orientation to the sound beam are adequately represented on an ensemble basis by a Rayleigh distribution. It is also assumed that the transducer beam shape is accurately known. The extent to which non-ideality in all or each of these assumptions can be expected to vary the precision of the final target strength estimates made is difficult or impossible to evaluate in the field.

A Monte-Carlo computer model of echo formation has been developed, to supply various categories of test data for analysis by the Peterson, Clay and Brandt method. This technique is used to comment on the precision of the method and on the effects of relaxing the various assumptions involved in its development.
DESIGN OF A LABORATORY TANK FOR UNDERWATER SOUND TRANSMISSION

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The design of a laboratory tank for simulation of acoustical conditions in coastal seas is described. In physical modeling, if the linear dimensions of the sea are scaled down by a factor of, say, 1/100 in the model tank, the operating frequency in the tank should be scaled up by the reciprocal of the factor, viz., 100. The dimensions of the model tank, length 10 m, width 5 m and depth 2 m are chosen for operation with a scale down factor in the range 1/100 to 1/1000. The tank walls are constructed out of reinforced cement concrete 20 cm thick and are made thicker over short lengths at intervals of 2 m for additional strength. Glass windows 50 cm x 30 cm are provided in the longer walls on opposite sides for optical probing of the sound waves if required. By means of vertical partitions which can be introduced into the tank parallel to the shorter sides, the length of the tank can be reduced to 4 m, 6 m or 2 m. The tank is sunk about 1 m below the floor level of the laboratory for convenience of operation from the top. It rests on a layer of neoprene which lies on top of a sand and cinder bed. A styrofoam slab surrounds the tank on all sides and provides vibration isolation and thermal insulation to the water in the tank.

A gantry carrying a platform moves on rails fixed on brick walls parallel to the length of the tank. The platform can be moved along the length or breadth of the tank to position the transmitting and receiving transducers. It is hoped to achieve an accuracy of 1 mm in each of the three dimensions. Rows of electrical heaters which can be moved up or down produce vertical thermal gradients required for refraction studies. Preliminary experiments indicate that temperature gradients of 25°C/m are possible. Compressed air blown through small nozzles on the surface of water produces corrugations which are intended to simulate waves in the sea. The tank walls can be lined with butyl rubber mats to provide sound absorbing surfaces.

It is hoped that these facilities will enable controlled variation of several parameters which produce fluctuation in sound transmission in the ocean and help in the design of optimal signals for detection.
THE BEHAVIOUR OF AN UNDERWATER SOUND PHASED ARRAY FOR THE REALISTIC CONDITION

Harbin Shipbuilding Engineering Institute

FU HUNG CHOU

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The radiation pattern and mutual impedance of many underwater transducers in a large planary array are obtained when the array surface is comprised of closed packed radiation pistons. In this analysis, a realistic baffle condition is used, the baffle impedance dependent on the internal mechanical impedance of the array element. The scalar wave solution due to active transducer above a planar realistic impedance boundary is solved by Sommerfield's method. The complex acoustical radiation impedance can be calculated by Schoct's method. Each element's velocity of the phased array has different value because the mutual impedance is a function of position and phased angle. The method proposed here for computing the velocity of element is adopted by a matrix inversion procedure.
REAL TIME ACQUISITION AND PROCESSING OF UNDERWATER ACOUSTIC SIGNALS
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INTRODUCTION

The Institute for Acoustical Research has been engaged in long-term, long-range underwater acoustic studies requiring real time acquisition and processing of underwater acoustical signals. Studies, in a closely related field, at the Cooley Electronic Laboratory, University of Michigan, have resulted in unique signal designs, processing techniques, and hardware for our studies. The underwater signals usually require substantial processing gains to raise them above the noise. Signal processing includes digital filtering, pseudo-random linear maximal sequences, pulse compression, auto and cross correlation, and spectral analysis. The analysis includes both time and frequency domain processing. Multichannel high-speed digital computer-based receiving systems have been designed at the University of Michigan and at the Institute for Acoustical Research to acquire, record, and process the signals on-line in the field. These systems have been deployed at various field sites with considerable success during the past seven years.

THE MEASUREMENT SYSTEM

These systems provide multichannel acquisition of acoustical data at rates up to 50,000 samples per second, processing and quantifying parameters such as signal power, signal phase, noise power, multipath arrival structure, signal and noise spectra (FFT). The functional building blocks for each system include: 1) Signal conditioner: samples each input sequentially, low-pass filters, gain adjusts each channel, and converts from analog to digital form (10-bits), 2) Central processing unit: mini-computer with disc pak, and a digital to analog converter for graphic output, 3) Output: a teletypewriter terminal, magnetic tape, and a cathode-ray display, 4) Programs: loaded from magnetic tape or disc; keyboard choices of carrier frequency, number and sequencing of inputs, filter bandwidths, and recording formats, 5) Communication with CPU: by teletypewriter without interrupting computational flow, 6) Time: countdown from internal clock or external standard, 7) Language: programs generally written in assembly language or Fortran.

POST PROCESSING

The recorded magnetic tapes are machine processed in our laboratory to produce detailed statistical analysis in time and frequency domains and graphic plots. The systems have proven to be reliable and indispensable for assimilating a mass of acoustical data.
RANDOM DISC ARRAYS WITH DIRECTIONAL HYDROPHONES IN VERTICALLY DIRECTIVE SEA NOISE

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INTRODUCTION

Random arrays have been used extensively, especially in electromagnetic applications. Such arrays are generally "thin", i.e., spatially under sampled. This paper describes a study of acoustic random arrays over a wide frequency range from the (oversampled) low frequencies to the (undersampled) high frequencies. This work is a discrete extension of the continuous array studies reported previously.¹

Noise gains and directional responses were studied for horizontal disc random arrays with 16 omni-directional or directional hydrophones. The directional elements are pressure-gradient units with Limacon responses such as a cardioid. Isotropic and vertically directive noise fields are considered. A set of random arrays are investigated that all have the same relative shape. The size of each array is based upon a design frequency, $f_D$, for which the mean inter-element spacing is one wavelength.

RESULTS

The well-known gain of $10 \log_{10} (N)$ is obtained with omni-directional elements above the design frequency, implying that the elements' contributions are essentially independent for inter-element spacings in excess of one wavelength. The gain falls off and is linearly related to frequency over much of the decade below the design frequency. The gain improvement due to directional elements is especially significant (3 to 6 dB) at frequencies below 0.1 $f_D$ and above $f_D$. The gain improvement is as little as 1 dB and as much as 6 dB for all frequencies evaluated.

The directional responses are considered. The horizontal and vertical beamwidths for the random disc arrays agree well with the theory of solid discs. The wide vertical beamwidth of the edge-fired disc captures the multiple-signal and noise arrivals. Sidelobe levels vary significantly for these 16-element arrays but are nominally -12 dB. The front-to-back ratio of the directional response is significantly improved with directional hydrophones.

The bearing precision of disc arrays is discussed and compared with linear arrays of equal number of elements.

THE POTENTIAL CAPABILITY OF SUPPRESSING THE NEAR INTERFERENCE POINT SOURCE BY USING A MODE-SELECTIVE ARRAY IN SHALLOW-WATER

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In this paper we suggest that, in addition to the azimuth information the potential informations (the differences of range and depth between the target point source and the interference point source in shallow-water channel) may be utilized by using a mode-selective array filtering the mode instead of the concept of directionality. The processing gain of the mode-selective array for suppressing the near interference point source was estimated theoretically, and the effect seems apparent.

\[ G \approx \frac{A_L(\tilde{\gamma}_N, \tilde{\beta}_N)}{\frac{1}{n} \sum_n \hat{U}_n(\tilde{\beta}_N) e^{-\beta_n r_5} \hat{U}_n(\tilde{\beta}_N) A_n e^{ik_n r_5 - \beta_n r_5}} \]

Where, \( \tilde{\gamma}_s, \tilde{\beta}_s \) and \( \gamma_N, \tilde{\beta}_N \) are the coordinates of target and interference respectively; \( A_n \) - the "eigen-spectrum" of the weight function \( W(\tilde{\beta}_s) \); \( \hat{U}_n(\tilde{\beta}_N) \) - eigenfunction; \( \beta_n, \kappa_n \) - eigenvalue; 

- \( A_L \) - transmission anomaly for near field; \( H \) - water depth; \( \lambda \) - wave length. The gain was estimated for a shallow homogeneous layer with a boundary reflection loss described by "three-parameters model". (1)

(1) If \( A_n \) are matched to the target, 

\[ A_n = \begin{cases} \hat{U}_n(\tilde{\beta}_N) e^{-ik_n r_5 - \beta_n r_5} & \text{when } n \leq N_{\text{eff}}(\tilde{\gamma}_s) \\ 0 & \text{other} \end{cases} \]

we have:

\[ G \approx \left( \frac{H}{\lambda \gamma_N} \right) A_L \]

(2) If only the \( m \)-th mode is filtered, we have:

\[ G \approx \left( \frac{H}{\lambda \gamma_N} \right) A_L \cdot \frac{\hat{U}_n(\tilde{\beta}_N)}{N_{\text{eff}}(\tilde{\gamma}_s)} e^{-2(\beta_m - \beta_N) r_5} \]

(3) If target is set in the single mode region (\( \gamma_s > \gamma_N \)) and only the first mode was filtered, we have:

\[ G \approx \left( \frac{H}{\lambda \gamma_N} \right) \frac{A_L}{\hat{U}_n(\tilde{\beta}_N)} \]

Some recent work on parametric array in shallow water has demonstrated that by using a parametric array the first mode selection can be conveniently implemented. (1)

References


THE INFLUENCE OF MULTIPATH PROPAGATION EFFECTS ON THE GAIN OF A HORIZONTAL LINE ARRAY

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A propagation loss model based upon the parabolic equation technique is used to investigate the influence of the vertical arrival angle structure of a sound field in the ocean on the gain of a horizontal line array. Range dependent structure of the gain variation closely follows periodicities in the sound field such as a sequence of convergence zones. For some practical situations losses of gain as high as 10 dB are predicted.
CALCULATION METHOD AND PARAMETRIC ACOUSTIC ARRAYS INVESTIGATION

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Taganrog Radio-Engineering Institute, USSR

This paper presents the calculation method and the experimental investigation results of the created parametric devices. The results of the Zabolotskaya-Chochlov's modified equation solution for space spreading of secondary radiation permit to build nomograms for parametric aerials calculation, which takes into account attenuation and diffraction of primary and secondary fields /1/.

The nomograms building is carried out with the help of a computer. The nomograms were used for calculation of several types of the created parametric acoustic devices /2/.

Experimental investigations were performed in a damped basin and in an open pond that provided "free" field conditions. The initial frequencies from 20 to 2000 KC with initial converters apertures from 15 to 305 mm were used in the investigations and developments. Diagrams of initial radiators directivity were within the limits of 1-30°.

Measurements of parametric arrays difference frequencies were carried out in the range of 0,1-3000 KC. Diagrams of parametric radiators directivity with flat initial converters were received at 3db level within the limits of 2-8° in the range of 1-250 KC. Using slightly convex converters allowed to widen the directivity diagram of the parametric radiator up to 30° and to decrease the forming zone range from 1,5 to 0,5m. Side lobes in directivity diagrams are absent, which is an obvious advantage of parametric radiators.

Developments of sets of parametric radiator devices are made on the basis of these investigations. These devices were demonstrated at exhibitions, including the USSR Exhibition of Economic Achievements in 1974 and 1977, the Exhibition in 1978 in Budapest and in Prague in 1979.

REFERENCES.


Turbulent boundary layer which forms on a plane surface is known to possess a pressure distribution of spectral components which span both the subsonic and supersonic regions of spectral space. The supersonic components are radiating components and would thus generate sound directly to the far field. This radiation would be present even if the surface were of uniform impedance, e.g., a rigid surface. The subsonic components, on the other hand, are nonradiating and would thus not generate sound directly. However, if the surface embodies impedance nonuniformities, subsonic components may be converted into supersonic components. The quality and quantity of the conversion is a function of the characteristic impedance of the fluid, the impedance of the surface, and the nature of the surface impedance nonuniformities. To estimate the radiated sound, in addition to determining the conversion, one requires knowledge of the pressure distribution of spectral components in the turbulent boundary layer. The paper deals with the experimental acquisition of such knowledge. In particular, it deals with the limitations and difficulties that one may encounter in attempting to measure the pressure distribution of spectral components in a subsonic turbulent boundary layer. The paper focuses on the use of wavevector filters for such measurements. Techniques which may be used to alleviate some of the limitations and difficulties are discussed. These include steering, shading, adapting, and blanketing the wavevector filters.
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NAIVE MODEL OF THE ROTATIONAL NOISE OF THE SHIP SCREW PROPELLER

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According to Ross [1] rotational noise amplitude spectrum of the ship screw propeller operating in a circumferentially non-uniform velocity inflow field is given by

\[ p_m(r',x) = \frac{1}{2c^{3/2}r_o} \frac{mZn^3D^4}{r_o} \frac{r_o}{r'} |\cos x| k_{TM}^m, \quad m=1,2,\ldots \]

where \( p_m(r',x) \) is RMS value of the pressure component at the frequency \( mZn \) in the point \( (r',x) \), \( r' \)-distance from the propeller, \( x \)-angle measured from propeller axis, \( Z \)-number of blades, \( n \)-number of revolutions per unit time, \( c \)-sound speed in water, \( r_o \)-unit distance, \( D \)-propeller diameter, \( k_{TM} = T_m/(\rho n^2D^4) \), \( T_m \)-RMS value of \( m \)-th harmonic component of the fluctuating thrust.

The aim of this paper is to present a simple closed analytic model of \( k_{TM} \):

\[ k_{TM} = \frac{a \pi}{8V_o} \frac{ZJ}{R^3} \int_{R_o}^{R} \frac{i[mZ\Phi(r)+\rho_n Z(r)]}{V_o} dr, \quad m=1,2,\ldots \]

Here \( a=1.8\pi \text{ rad}^{-1} \) is a constant, \( J=V_o/(nD) \)-propeller advance ratio, \( V_o \)-ship's speed, \( R \)-propeller radius, \( R_o \)-hub radius, \( r \)-radial coordinate on the blade, \( s(r) \)-blade profile length at the radius \( r \), \( \Phi(r) \)-angle describing blade skewness: angle between the reference direction at the blade and the straight line connecting the propeller centre and the centre of lift and drag for the blade element at the radius \( r \); \( V_g(r) = |V_g(r)|/2 \) and \( \Phi_g(r) = \text{arg} V_g(r) \) are from \( V_g(r) = (2\pi)^{-1} \int_0^{2\pi} V(r,\theta)e^{-i2\theta} d\theta, \quad \theta=0,1,2,\ldots, \) where \( V(r,\theta) \) is axial component of the velocity disturbance at the point \( (r,\theta) \), with \( \theta \) being screw angular position. The direction of propeller rotation corresponds to the increasing values of \( \theta \) and \( \Phi(r) \).

Various assumptions were used and details omitted in the model construction. Such a naive model, being not very accurate, reveals the influence of screw and inflow field parameters rather directly, without numerical realization. This becomes obvious in the special case of non-skewed screw, \( \Phi(r)=0 \), operating in the radially similar inflow velocity field, \( v(r,\theta) = v(r)v(R,\theta) \), \( \theta \in [0,2\pi] \):

\[ k_{TM} = \frac{a \pi}{8V_o} \frac{ZJ}{R^3} \int_{R_o}^{R} \frac{1}{V_o} \int_{R_o}^{R_o} s(r)v(r)dr, \quad m=1,2,\ldots \]

REFERENCES
On board of many ships cavitation of propeller- and rudder-blades reinforces considerably vibrations and noise in the stern area. Remedial measures depend on the existing cavitation form, e.g. tip vortex-, hub-, bubble- or sheet-cavitation. Optical observation of the cavitation on board of a running ship is difficult, expensive or even impossible.

A German research program is looking out for an acoustical method for recognizing the existing cavitation form on a running ship-propeller. Measurements on a model of the fast container-ship "Sydney Express", which is known to have a highly cavitating propeller, have been carried out in the big circulating water tunnel of the Berlin Model Basin according to Froude and cavitation similarity with a miniature hydrophone lodged in the water-filled afterpeak of the model. The analysis of noise spectra and cavitation photos show a close correlation.

The corresponding measurements and observations on board of the "Sydney Express" have been achieved in November 79. In addition to underwater acoustic measuring in the water-filled afterpeak pressure fluctuations and aerial sound analysis have been carried out as well as photographic documentation. They are presented in this paper.
M-sequence Correlation Method for Measurement of Underwater Sound Propagation Time

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INTRODUCTION

Velocity of current going in and out through a strait is possible to be measured with difference of underwater sound propagation times along and against the current. However, ambient pulsed noises often interfere with the measurements of sound propagation times when a pulsed sound is used. Then, a correlation method is proposed for measuring sound propagation time, because the method is free from interference of ambient pulsed noises. As a practical method, two period M-sequence signal method is discussed here.

TWO PERIOD M-SEQUENCE SIGNAL METHOD

M-sequence signal is one of periodic pseudo-random signals with two values. A truncated M-sequence signal (Fig.1(a)) to two periods is transmitted and received. This received signal is correlated with one period discrete M-sequence signal (Fig.1(b)). This correlation with truncated discrete M-sequence signal enables faster processing with higher resolution. The correlated signal (Fig.1(c)) forms two peaks accompanied with low level noises. There are self-generated noises outside of the two peaks, and the level between the two peaks is almost zero. Then, an instantaneous amplitude of this correlated signal is compared with that of one period later, and the smaller one is plotted. In the result, the signal (Fig.1(d)) with one peak is obtained. This signal is similar to the received signal by pulse method.

EXPERIMENTS

A sound propagation test was performed in the sea by using above mentioned M-sequence signal (Period of the M-sequence: 127, Frequency of clock pulses: 50 kHz). Propagated signals up to the range of 3400 m were detected successfully. One example is shown in Fig.2.

Fig. 1 Two period M-sequence signal method

Fig. 2 An example (Range: 1300 m)
ANALYSIS OF SEA MAMMALS SOUND SIGNALS.

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Results of analysis of mammals sound signals, presumably belonging to humpback whales, are reported. The signals have been recorded by single hydrophone placed on the bottom of ocean shelf. Among the other sounds the most pronounced one were of the powerful frequency modulated pulses with duration near 1.5s in the range of 2kHz-200Hz. The signals were analysed by computer with high frequency resolution. The following characteristic features were revealed:

1. Character of modulation changed during the emission (short-time spectrum is shown on figure below). On the first stage (I) a frequency decreased approximately by hyperbolic law (duration 0.3sec.) then it failed linearly during 1 sec. (II). At the end of the signal a frequency rose shortly and then dropped to 200Hz (stage III in the figure).

2. Together with a principle change of signal frequency there are observed signal harmonics which are not the result of recording or processing nonlinearity but inherent in signals themselves.

3. All signals are characterised by remarkable similarity - the modulation law is precisely the same for each signal. Nevertheless careful analysis shown that the sounds belonged to two different animals.

4. In addition to direct signals up to three bottom-surface reflections was observed (dotted lines in the figure). Time delays of the reflections were used to determine a depth and a distance to the animals. In result it was found that one of the couple stayed moveless near sea surface at a distance of 650 m. off hydrophone and other swam with velocity 0.3 m/sec. near the bottom at a distance of 100-200 m.

Various suggestions on biological significance of FM signals are discussed. It seems that the most likely explanations are connected with animals need for location and navigation in the sea.

Literature:
EMPIRICAL MODEL OF THE WIND-GENERATED NOISE IN THE OCEAN.

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The investigations of recent years have shown that the spectral density of ambient noise, \( p^2(f) \), is the sum of the two components: the first one, dependent on the local wind speed by the power law, \( p^2(f) = a(f) u^n(f) \), and the second— independent of the wind speed, \( p_0^2(f) \) ("zero" noise). A great variety of results on the wind dependence of noise (I) is mainly caused by the processing procedures. Combining of results of two studies (2,3) in which the coefficients \( p_0^2 \), a and \( n \) were determined by the mean-square method with using of the long series of data allows to make the following conclusions:

1. There are two different mechanisms of the wind-generated noise, one of them prevails at frequencies under 40Hz, another— above 250Hz.
2. There is not a great amount of experimental data for the lower frequency range, and available data can be summarized by means of simple empirical formula: \( p^2(f) = 10^{-3} u^4/f^3 \) (p - dyne/sm\(^2\) Hz\(^{1/2}\), f - Hz, u - m/sec). The theories of the low frequency generation developed until now are in agreement with this formula.
3. The high frequency noise can be described by an empirical formula \( p^2(f) = bu^n/(f^2+f_0^2) \) where the coefficients b and n do not depend on frequency, "but can differ in various ocean areas and propagation conditions; \( f_1 = 1kHz \). That simple formula suggest an idea of occurrence of a common mechanism of noise generation in this frequency range, but at present there is no satisfactory quantitative theory.
4. Results of calculations by the given formulas show that there is a "plato" in the spectrum (40-200Hz) really observed in experiments. At not very high wind speeds "zero" noise mainly contributes into the total noise in this frequency range. Thus, it is not surprising, that maxima due to distant sea traffic are observed in this range.
5. "Zero" noise can be also described by the simple formula:
\[ p_0^2(f) = A/(f^2+f_0^2) \], \( f_0 = 100Hz \), coefficient A depends on place and conditions. The origin of "zero" noise is an outstanding problem, but it is probably caused the effects of wind in distant areas.

(2) Crouch W.W., Burt P.J., JASA, v-51, 3(pt1), 1972.
A REVIEW OF AMBIENT NOISE STUDIES IN THE PACIFIC OCEAN

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and

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In deep ocean areas with a relatively high shipping density, such as the North Atlantic, ambient noise is principally caused by wind generated noise at frequencies above 100 Hertz and by ship generated noise below 100 Hertz. A review of recent ambient noise measurements in the Pacific Ocean indicates that although some areas of the North Pacific are characteristic of high shipping density, most other areas especially in the South Pacific are significantly different. An analysis is made of the statistics of the noise, possible sources of noise both natural and manmade, and propagation conditions in various locations in the Pacific Ocean.
THE SOUND FIELD FROM A DISTRIBUTION OF SOURCES NEAR THE OCEAN SURFACE

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There are a number of possible mechanisms of low frequency noise generation related to wind action at the ocean surface. The theory presented in this paper relates distributions of sources, up to order quadrupole, resulting from near surface dynamics, to the received noise at depth.

Extending the theory of Lighthill, Curle (1955)* has shown that the density field at position \( \mathbf{x} \), due to the motion at \( \mathbf{y} \) in a bounded fluid is given by the expression

\[
\rho - \rho_0 = \frac{1}{4\pi \omega^2 \alpha_0^2} \int \frac{T_i(y, t - \frac{r}{c_0})}{r} dy \cdot \frac{1}{4\pi \omega^2 \alpha_0^2} \int_S \left( \frac{1}{r} T_{ij}(y) + \theta^2 \delta_{ij} \right) dS(y),
\]

where the symbols are as defined by Curle. By analogy with classical acoustics, the three terms in the expression may be considered to represent the effect of source distributions of quadrupoles, dipoles, and monopoles, respectively.

Considering the case where the sources are concentrated near the horizontal plane \( \eta = \text{constant} \) (for example, the ocean surface), we recast the problem in wave number frequency space such that the source strength distributions are transformed as their cross spectra, \( \Phi_{ijlm}(k_z, \omega, \eta) \) where \( k_z \) is the horizontal wave number. Here we have assumed horizontal homogeneity and stationarity. The sound pressure spectrum, \( P(\omega) \), at a depth \( z = \eta - z_0 \) can now be calculated. Taking as an example a source distribution of quadrupoles only, we show that, in the far field \( (\omega/\alpha_0 \gg 1) \)

\[
P(\omega) = \frac{1}{(2\pi)^2} \int \int \Phi_{ijlm}(k_z, \omega, \eta) H_{ij}(z, \omega, k_z) H^{*}_{lm}(z', \omega, k_z) \frac{dy_1 dy_2}{z' - z} dk_z
\]

where \( H_{ij}(z, \omega, k_z) = \mp 2\pi \int_0^\infty \frac{n_i n_j}{4\pi \alpha_0^2} \exp(-i\omega|z_0 + k_z \cdot \eta|) dy_1 dy_2 \)

The figure shows the non-dimensional "coupling" factor, \( \sqrt{H_{ij}^* H_{ij}} \) for \( i = l, j = m, 2\pi \)

plotted as a function of \( \omega/k_z \alpha_0 \) for \( \omega/\alpha_0 = 2\pi \). \( \omega/k_z \) may be interpreted as the phase speed of the source component at frequency \( \omega \) and wave number \( k_z \). As the "coupling" factors depend on the angle between the horizontal component of the source beam pattern and \( k_z \), only the maximum values have been plotted. The results show that by far the most efficient coupling occurs for \( \omega/k_z \alpha_0 \gg 1 \). Analogous results are obtained for dipole and monopole source distributions.

An exponential pressure pulse of the form

\[ p(t) = p_0 e^{-t/\theta} \]

is observed at the output of a sonobuoy as a result of an underwater explosion. The buoy receiver is linear but uncalibrated. Thus a direct measure of the peak pressure \( p_0 \) is not possible. A technique is described that permits quantification of \( p_0 \) under this set of circumstances. It involves a knowledge of wind speed (or an equivalent spectral noise measurement) and the total spectrum of the ambient background.
SENSORS AND MATERIALS FOR HYDRODYNAMIC ACOUSTICS MEASUREMENTS

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In each hydrodynamic situation, where acoustic measurements are to be performed, there is a number of conditions a sensor should fulfill. The main condition is that the hydrodynamic and hydroacoustic environment is not significantly changed by the presence of the sensor.

The subject of this paper is to present measured hydroacoustic properties of a developed set of the sensors and properties of materials used for sensors development. Sensors are designed for measurements on the models in towing tanks and cavitation tunnels as well as for measurements on the full scale objects. Sensors are divided into three groups:

- sensors for measurements in a flow: hydrophone inside the hollow aluminium hydrofoil, sipas (polyamide plastics) and steel hydrofoil with the hydrophone on it and the hydrodynamically profiled hydrophone,

- flush mounted sensors: mounted in the plexiglass and steel wall of the test section,

- sensors for measurements in calm water outside the flow; resolution and gain of elliptic acoustic mirror are investigated and water filled acoustically lined tanks for application in the cavitation tunnels are described.

Acoustical properties, such as transparency (for acoustic windows) and reflection and dumping (for tank lining) for some materials used in sensors development, are measured and presented. As the examples of the typical measurement results, Fig.1 shows directivity pattern for pressure transducer Kistler 6038 mounted in the plexiglass wall, and Fig.2 shows resolution curve for the elliptic acoustic mirror.
MEAN VELOCITY OF SOUND IN THE CADIZ AND ALBORAN SEAS. MONTHLY AND YEARLY CORRELATION

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INTRODUCTION
This paper presents some results obtained from the correlation between two oceanic masses of water united by a narrow channel. The interest of this analysis lies upon the well known different behaviour, under the acoustical point of view of both the Alboran and the Cadiz Seas united by the Strait of Gibraltar. In particular we have tried to evaluate the velocity of sound profile in one sea when such structure is known in the other.

ANALYSIS
From a well established statistics of thermal and density data, the velocity of sound/depth was computed; then we tried to show up the actual correlation among the velocity structure in both seas. In order to obtain representative results, the monthly mean traces were the reference variables. Three methods of comparison between profiles were considered: 1) The whole profile; 2) Layers of 0-200 m, 200-600 m, and greater than 600 m depths; 3) Samples of C(z) at equal depths in both seas. Fig. 1 shows some results, there the horizontal axis is the normalized difference between the actual statistical velocity of sound computed through the linear regression equation obtained for each different type of comparison. The straight line "I" is the monthly mean (April) which all of the curves refer; "II" is the velocity of sound using the 1st method; "III" is the result obtained by dividing the depth in 3 layers; "IV" represents the normalized velocity found relating the velocities in both seas, as equal depths; the ordinate axis is the depth in meters, the scale of the horizontal axis corresponding to the curves III and IV appear in the upper part of the graph.

RESULTS
From the example analyzed, that it can be considered a representation of the whole survey; the curve III shows the better approach; in this case the correlation coefficient "r" ranges from 0.80 to 0.98 for all of the three variables T. S. and C. On the other hand the adjustment got by linear, exponential, logarithmic and potential regression were very similar; we have chosen the linear regression for obvious reasons. Finally we can affirm, that if the meteorological and the wind generated waves situations in both seas are, to some extent, similar, then it is possible to have the C(z) profile in either sea knowing the C(z) function in the other.
THE INVESTIGATION OF THE FINE STRUCTURE OF THE SOUND VELOCITY FIELD IN THE PACIFIC OCEAN

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When we say "fine structure" we mean different types of micro- and meso-scale disturbances of the conservative and dynamic nature observed against the background of large-scale variability of the sound velocity field in the ocean. Investigations have been made by means of continuous vertical probing from a drifting ship using probes measuring sound velocity and dynamic medium parameters. A remote method based on echo sounding of the water thickness during ship run has also been used.

The probes used contain a cyclic meter for sound velocity measuring which is modified so that apart from sound velocity it can also measure the speed of the current. These probes include two of such meters with their sensors arranged along the vertical at fixed distances in a way allowing additional measurement of the local gradients of the parameters indicated. In addition to this, the probe contains a direction sensor determining the orientation of the device with regard to the magnetic meridian of the Earth and a depth meter. Another modification of the acoustic probe intended for measuring current speed uses a Doppler measuring device operating under conditions of continuous ultrasonic emission with a frequency of 5 MHz.

The analysis of the results of the measurements performed in different regions of the Pacific Ocean shows that in spite of an extremely great variety of the fine structure observed, one can make a classification reflecting the most general patterns and mechanisms of its formation. In particular, the areas of the ocean having different structure of water masses and located in different climatic zones as well as frontal zones and areas with complex dynamic characteristics have specific patterns of the fine structure of the sound velocity field. They differ sufficiently between themselves by quantitative parameters and by space and time variability.
The dynamics of the South Coral and Tasman Seas has recently been shown to be governed by the East Australian Current and the Tasman Front; the current flowing south along the coast of Australia and then turning east around 34°S to feed the front which has the form of a planetary wave with a zonal wavevector. This planetary wave spawns warm core eddies to the south and cold core eddies to the north in a regular manner, conforming to theoretical postulates that such systems are controlled by a conservation of potential vorticity in which the linear and nonlinear effects are approximately of equal magnitude. These features, which cause a perturbation to the ocean's regular sound speed structure, can be scaled in a canonical manner, allowing a deterministic evaluation of their effects on an acoustic wave propagating through them. These effects have been calculated in the frequency band 30 Hz to 3 kHz using range dependent propagation models and show significant departures from an unperturbed ocean; the results have a direct application in the inverse problem of acoustic tomography.
SOUND FLUCTUATIONS AND INTERNAL WAVES IN SHALLOW WATER

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A very sharp thermocline occurs in August in the shallow water region of Yellow-Sea near China coast. The temperature difference $\Delta t$ is about 10°C, and the thickness of transition layer is only a few meters. The experiments of sound fluctuations and the observations of internal waves were conducted in this region by using a flextensional transducer as a sound source set at the bottom. The CW signal of 1.6 kHz and the impulse signals of 500 ms and 5 ms were received at the fixed range of 3.6 km. In the meantime, the internal waves in the thermocline layer were measured by using a thermistor chain consisted of ten platinum temperature sensors. The fluctuation processes of temperature were recorded, and the example was shown in Fig.1. In Fig.1, the depth of the central sensor is 22 m, and the vertical interval between two adjacent sensors is 0.8 m. Correlation functions and power spectral of the temperature fluctuations were analyzed. The Väisälä frequency $\omega_z$ was calculated for the correspondent hydrographic conditions; the period is about 1 min. in the thermocline layer. Below this Väisälä frequency the spectrum characteristic is approximated to $\omega^{-3}$. The correlation functions and power spectral of the sound fluctuations were also analyzed, and the effects of the internal waves on the sound fluctuation were discussed.

The example of intensity fluctuations is shown in Fig.2.
Tenth International Congress on Acoustics

Surface Acoustoelectric Interaction and its Applications*

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Surface Acoustic Wave (SAW) propagating on a piezoelectric material carries with it an evanescent electric field which interacts with the carriers in a semiconductor placed adjacent to the piezoelectric substrate. This so-called acousto-electric interaction gives rise to a d.c. voltage and second harmonic generation. A theory for these acousto-electric voltages has been developed which includes the effect of these generated voltages on the semiconductor surface itself self-consistently.

This acousto-electric interaction has important applications in the following two fields: a) Real time signal processing and b) Contactless characterization of the electrical properties of semiconductors. Experimental results using LiNbO₃ substrate and different semiconductors demonstrating the different applications will be presented. SAW is generated using interdigital transducers centered at 50, 100 and 200 MHz. Some of the signal processing functions which can be performed are convolution and Fourier Transforms. For the semiconductor surface characterization one can effectively image the surface for point by point study using a long and a short pulse. Conductivity, type of carrier, lifetime are some of the properties which have been determined for different semiconductors.

* Partially supported by ONR Contract No. N00014-75-C-0772 and AFOSR Grant No. 77-3426.
FRONT-TO-BACK RATIO OF ULTRASONIC TRANSDUCER HAVING BACKING LAYERS FOR MISMATCHING
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Attempts were made to analyze the method of designing the condition in which intermediate layers—provided on the back face of piezoelectric elements had been made into layers for acoustically mismatching; and to make on an experimental basis an ultrasonic transducer having 2 backing layers.

As for underwater transducer having mismatching layers only on the back face, it was confirmed that the open transmission factor regarding the back incident waves could be approximated to the front-to-back ratio; besides the 2 backing layers were determined according to the Richard's key theorem.

The calculated results of the front-to-back ratio after determining 2 backing layers and experimental results are shown in figure. The data were obtained by measuring the beam patterns on elements of a planar test array over the frequency range from 180 to 620kHz. The back levels used in determining the ratio were taken as the peak level at the angle $\theta = 180^\circ$ in the increment $\pm 60^\circ$.

Since the transducer proposed in this paper is not constructed of rubber cork and similar compliant materials, that will withstand high pressure from deep ocean immersion.

Figure.

Comparison of experimental results with theory for front-to-back ratio of ultrasonic transducer having 2 backing layers.

($Z_2 = 31 R_w$, $Z_1 = 1.1 R_w$).

Aperture sizes in experiments:

- $1.9 \lambda \times 2.5 \lambda$: wavelength
- $1.3 \lambda \times 1.7 \lambda$: at $f = f_0$
- $0.9 \lambda \times 1.2 \lambda$: in water
INTRODUCTION
On montre une méthode pour mesurer la vitesse des ondes acoustiques dans un milieu poreux empli d'eau, et son application à la détection des ondes transversales. Cette méthode est le résultat de trouver les ressemblances des valeurs théoriques et expérimentales de la réflexion des ondes acoustiques tombant en incidence oblique sur un système composé par plusieurs couches.

Bases théoriques
D'après les calculs du coefficient de réfraction d'un système avec plusieurs couches développé par Spielvogel (1), nous avons réussi à déduire les courbes qui donnent la dépendance du coefficient avec les paramètres: fréquence et angle d'incidence (voir fig. 1). Étant donné qu'aux milieux fluides on ne propage pas des ondes transversales, et du même pour des milieux solides en incidence normale, il est assez simple identifier la présence des ondes transversales du chaque couche.

Techniques expérimentaux
On a fait un système réfléchissant composé par trois couches dont celle du milieu puisse être de l'eau ou du sable mouillé étant tout ceci plongé dans une cuve en eau douce. On émet des pulses avec durée inférieure aux 200 μs et fréquences compris entre 20 et 100 kHz.

Résultats et conclusions
On conclut que des milieux poreux tels que du sable mouillé d'eau, c'est à dire le cas assez courant au fond de la mer, sont capables de permettre la propagation des ondes transversales. On mesure aussi la vitesse de propagation de ces ondes là pour plusieurs types de sable.

Bibliographie
Laser Interferometer Probe for Underwater Pulse from Electric Arc.

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A laser interferometer is used to sense the sound field near a low amplitude underwater acoustic pulse produced by an electric arc. Experiments conducted in a small tank show the sensitivity of the method for examination of the sound field and the time lapse geometry of the water cavity which follows the electric discharge.
INTRODUCTION. Mobile and randomly distributed inhomogeneities which occur in the sea over a broad variety of space and time scales [1] will cause random time-varying amplitude and phase fluctuations in a sound wave propagating between a source and a receiver. The considerable interest recently shown in the application of parametric acoustic arrays to shallow water sound propagation [2-6] has rendered necessary a detailed study of the effects of medium inhomogeneities which will cause the source density function to vary randomly and scatter the nonlinearly generated difference-frequency signal resulting in additional fluctuations in the pressure measured during the registration of depth functions in shallow water. This paper aims at an illumination of the effects of thermal inhomogeneities - turbulent eddies and thermal 'patches' - on the nearfield and near farfield region of the difference-frequency wave.

THEORY. The deterministic model studies reported in [6] are being generalized to comprise random variations in the medium sound velocity and density in space and time. Coefficients of amplitude variations [7] are being evaluated in the wave regions considered for the nonlinearly generated difference-frequency wave and a linearly transmitted wave of equal frequency using the various coherence functions associated with amplitude and phase fluctuations and assuming complete longitudinal and transverse correlation of the fluctuations over the effective interaction volume of the primaries on a par with the assumptions leading to a farfield theory in [7].

EXPERIMENT. The experiments are performed in a shallow water test basin, 12 m long and 1.6 m in width using a constant water depth of 0.06 m over a medium sand seabed. The primaries transmitted by a 1 cm diameter transducer are formed through a 100% amplitude modulation of a 4 MHz carrier wave with a 200 kHz wave which in the water through sideband interaction (suppressed carrier) will lead to a difference-frequency wave of 400 kHz. Primary source level: 205 dBe re 1 μPa·m. The thermal inhomogeneities are produced by a movable array of electrical resistance wires positioned vertical to the parametric array axis.

DISCUSSION. Preliminary results seem to indicate that amplitude fluctuations tend to increase with increasing range, but no significant differences seem to exist between the linearly and nonlinearly generated waves apart from those differences also found in the isovelocity or temperature stratified cases discussed in [2-6].

REFERENCES
I. PHYSICAL ACOUSTICS
TEMPERATURE DEPENDENCE OF THE VIBRATIONAL COLLISION NUMBER FOR $O_2-H_2$, $O_2-He$
AND $O_2-CO_2$ SYSTEMS BETWEEN $300^\circ K-675^\circ K$*

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The vibrational collision number for three binary systems ($O_2-H_2$, $O_2-He$ and
$O_2-CO_2$) has been measured between $300^\circ K-675^\circ K$ using an acoustically resonant
technique. It has been found that, for instance, for $O_2-He$ mixture, the
experimental value agrees within 15% of J. G. Parker's theoretical prediction if the relative collision rate
and the relative vibrational collision efficiency are taken to be 1.58 and $3.53 \times 10^2$, respectively, assuming a
separation of repulsive centers in the oxygen molecule itself 0.605A and the
force constant for an $O_2-O_2$ interaction 4.10A$^{-1}$. When the data are extrapo-
lated to the pure oxygen case, the experimental vibrational collision number
is found to be in excellent agreement with the theory. Extrapolation of the
shock tube data$^2$ to this temperature region gives values which disagree very
strongly with both Knötzels$^3$ and Shields and Lee's$^4$ data and that obtained
here.

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SOUND PROPAGATION IN DUCT BENDS

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INTRODUCTION: This work concerns the acoustic characteristics of duct bends in the absence of flow. An elbow which joins straight sections of rigid duct was studied and the range of frequencies examined was such that the (2,0) mode could propagate. The pressure field was obtained from a numerical solution of the Helmholtz equation and experiments were performed to validate the model.

RESULTS AND DISCUSSION: Results for a 90° mitred bend are shown. For values of the wave number below the cut-on value of the (1,0) mode \( (k a = \pi) \), the results of the present work agree closely with the experimental results of Lippert [1]. Above this, however, the literature is almost entirely devoid of data. Said [2], using a mode matching technique produced results which differ significantly from the present work. The figure shows experimental and numerical results for the energy components of the reflected wave when either a plane wave or a first cross mode was incident on the bend. The results agree well. The figure shows that, except close to cut-on frequencies, the (1,0) mode transmits more energy around the bend. The numerical analysis also showed that most of the transmitted energy is in the form of the higher order modes. This has significance when acoustic absorption near the corner is being considered. The numerical results indicate that whenever the duct width approaches an integer multiple of half wavelengths, the incident plane wave is totally reflected. This occurs at each modal cut-on frequency and was verified up to the (3,0) mode. With a turning vane in the bend total cancellation of the transmitted sound took place below the cut-on frequency of the (1,0) mode and well below the design frequency at which the mean path length difference was equal to half the wavelength, where interference was expected to cause cancellation. Such effects have been found before [3,4] but the present solution showed additionally that small values of the reflection coefficient existed at the design frequency. Previous work did not examine the properties at these higher wave numbers. Clearly, in some instances, turning vanes can be of advantage acoustically and aerodynamically.

ACKNOWLEDGEMENT: The authors gratefully acknowledge the technical assistance given by Mr. I.G. Pearson.

REFERENCES:

Reflection 90° mitred bend;
---num.no vane;...num.vane;
\( \bullet \)Exp(o,o);\( \Delta \)Exp(1,0)(no vane)
ELASTIC WAVES IN CYLINDRICAL LAYERS
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THE AXISYMMETRIC SOLUTION IN A HOMOGENEOUS CYLINDRICAL LAYER

In a homogeneous elastic region bounded by concentric cylindrical surfaces, the equations of elasticity for axisymmetric wave motion may be solved as outlined below. Let the displacements and components of traction on a cylindrical face be of the form

\[ [u_x, \tau_{xx}, \tau_{xz}, u_z]^T = X(r) \exp i (kx-wt). \]  

Then, the use of elastic potentials, each of which satisfies Helmholtz' equation, allows us to express \( X(r) \) as \( A(r)C \) where \( A(r) \) is a known 4x4 matrix function of \( r \) and \( C \) is a column vector of 4 unknown constants. The relation between \( X \) at one radius and \( X \) at another may be expressed as

\[ X(r) = A(r)A^{-1}(r_o)X(r_o) \equiv Q(r,r_o)X(r_o). \]  

The transfer matrix, \( Q(r,r_o) \), which is dependent on the density and the shear and compressional speeds, has been found explicitly by the matrix multiplication in (2), and obviously has the property that \( Q^{-1}(r,r_o) = Q(r_o,r) \).

THE CASE OF SEVERAL CONCENTRIC LAYERS

Consider a set of \( n \) homogeneous cylindrical layers occupying the \( n \) regions between \( r=r_o, r_1, \ldots, r_n \). For well bonded layers, i.e., \( X \) continuous across the interfaces, the transfer matrix for the entire set of layers is given by

\[ Q_1(r_0, r_1)Q_2(r_1, r_2)\ldots Q_n(r_{n-1}, r_n). \]  

If fluid infiltrates one of these interfaces, say \( r=r_i \), it may happen that \( u_x, \tau_{xx}, \tau_{xz} \) are continuous but that \( u_z \) is discontinuous with

\[ \hat{u}_z(r_i^+) - \hat{u}_z(r_i^-) = \xi \tau_{xz}(r_i) \]  

where \( \xi \) is a constant of proportionality. This is a special case of a general linear slip interface. In this case an additional interface matrix appears between \( Q_i(r_{i-1}, r_i) \) and \( Q_{i+1}(r_i, r_{i+1}) \) in (3) of the form

\[ I + (i\xi/\omega)e_k e_k^T \]  

where \( e_k \) is the unit column vector with components \( \xi_k \), \( i=1, \ldots, 4 \). Thus a transfer function may be specified for any set of cylindrical elastic layers with or without shear between any pairs of layers. Examples of modal propagation in the axial direction are presented for several situations.
ON THE TRANSFER FUNCTIONS OF THE HYDRODYNAMIC TANKS AND TUNNELS WITH ACOUSTIC CHAMBERS

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Noise generated by the ship screw propeller is usually measured on the models in towing tanks and cavitation tunnels. Transfer functions needed for calibration of measurement results for low-frequency dipole source field (corresponding to rotational noise) and high-frequency monopole source field (cavitation noise) were investigated.

In the concrete walls tank (6.2x12.5x27.6 m) two types of shallow immersed sources were considered, monopole and dipole oriented along the tank. Omnidirectional receiver was located in the vicinity of the source, inside the lobe of dipole directivity pattern. In Fig.1 the ratio of the received pressure amplitude and the reference one is shown as the function of sound frequency. The reference is the pressure that would be generated by the free source at unit distance in the direction of the directivity pattern maximum. Monopole field (curve a - result of measurement) in the low-frequency range was found to be much more attenuated than the dipole field (b and c - calculated). Simple model of the dipole field incorporating direct wave and the one reflected from the water surface, was used (b). This model is applicable to the case of described geometry (see elsewhere above and Fig.2). It was checked by the model with many total reflections from the surface, bottom and side walls (c in Fig.1).

The cavitation tunnel of the cross-section 0.5x0.5 m was provided with two water-filled metal chambers (0.5x0.5x1 m). These were separated from the tunnel by the acoustically transparent windows. Reverberant field of the monopole source located in the tunnel was found in the chambers; Fig.3(a) describes the pressure amplitude referenced to the free field at unit distance vs. path length of the received wave. Simple model explaining this dependence was constructed (b). When the chambers walls were covered by the sound absorbing rubber wedges, the reflections were suppressed (c). However free field conditions were obtained only after the elliptic mirror with source and receiver in the focal points was applied inside the chambers (Fig.4).
SYSTEMATIC OF APPROXIMATION OF STEPPED SOLID WAVEGUIDES TO CONTINUOUS SYSTEMS

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INTRODUCTION

Under a practical point of view to construct solid sonic concentrators by means of very simple ways can represent a contribution to this technology. In this sense, this paper presents two means of achieving stepped acoustic concentrators equivalent to "continuous" systems.

ANALYSIS

The concentrator with continuous profile \( f(x) \), with some restrictions, can be approximated by a series of cylindrical waveguides of varying lengths and radius, \( l_i \) and \( r_i \). The series of \( l_i \) and \( r_i \) form a double infinite ensemble of possible solutions and to focus on the right \( l_i, r_i \) series, some considerations have to be given; for instance, the resonance frequency and magnification need to be as much closer as possible to those of the continuous line; on the other hand the computation of the \( l_i, r_i \) series, must be an easy one if the method have to show interest.

This paper suggests two ways, experimentally tested, for getting those values: method 1 suppose that the length \( l_i (i=1,2,3...,n) \) is given by \( l_i = l_k / n \) (\( l_k \) is total length of the continuous line; \( n \) is number of partial cylindrical lines), the radius \( r_i \) is given by \( r_i = f(x) \cdot r_j / f(l_j) \), where \( r_j \) is the radius of the source transducer. The 2nd method implies the relationship, \( Z_i = \frac{Z_{i-1}}{r_i} \cdot Z_{i+1} \), where \( r_i = \frac{r_j}{Z_{i-1} \cdot Z_{i+1}} \) and from the radius, the length of the partial line is readily obtained, \( l_i = f^{-1}(x_i) \cdot f^{-1}(x_{i-1}) \). Of course many other methods could be considered, but those above mentioned present great simplicity and their calculus converges rapidly.

RESULTS

Fig.1 shows some of the results. A catenoidal concentrator was compared with five different stepped waveguides (4, 6, 8, 10 and 12 total number of partial cylindrical steps) (ordinate axis); the horizontal axis represents the ratio between the magnification of each concentrator and that of the "continuous" line. Also the results using both methods, are shown. These results implied an input reflection coefficient \( R_0 \) very low; \( R_0 \) also can be obtained very easily in the form of a polinomial of complex variable as given by Holte and Lambert (1).

(1) J.A.S.A. 33, 3 pp 289-301. 1961
On considère une plaque mince infinie, placée dans le plan $z = 0$ ; l'espace est occupé par un gaz parfait dans lequel se trouve une onde sonore sphérique placée en un point $S(0, 0, s)$ et émettant une onde monocromatique ($e^{-iwt}$). On cherche à déterminer la pression $p$ et l'amplitude de vibration de la plaque $u$. Il est classique d'obtenir les transformées de Fourier bidimensionnelles $\beta$ et $\alpha$ de ces deux fonctions. Pour effectuer la transformation inverse, on décompose $\beta$ et $\alpha$ en une somme de produits de fonctions dont on connaît les originales ; on aboutit alors à une représentation exacte de la solution. En particulier, la pression $p$ prend la forme suivante :

$$p(M) = \frac{e^{ikr(S,M)}}{4\pi r(S,M)} - \frac{e^{ikr(S,M)}}{4\pi r(S,M)} + \sum_{\mu(p), e^{ikr(M,p)}} \frac{\mu(p) e^{ikr(M,p)}}{4\pi r(M,p)} dP + 2 \Re \int_{\mathbb{R}} \frac{\mu(p) e^{ikr(M,p)}}{4\pi r(M,p)} dP$$

expression dans laquelle les densités de couche $\mu(p)$ et $u(p)$ sont explicitées en fonction des paramètres caractéristiques de la plaque (on a une expression analogue pour $M$ dans le $1/2$-espace $z < 0$).

Cette expression peut être développée en une série convergente de la forme suivante :

$$p(M) = \frac{e^{ikr(S,M)}}{4\pi r(S,M)} - a(\theta) \frac{e^{ikr(S,M)}}{4\pi r(S,M)} - \sum_{n=1}^{\infty} D^n a(\theta) \frac{e^{ikr(S,M)}}{4\pi r^n(S,M)}$$

$$+ \sum_{4} \text{termes "complémentaires"}$$

Ici $a(\theta)$ est le coefficient de réflexion des ondes planes de la plaque ; $D^n a(\theta)$ est une dérivation d'ordre $2n$ de $a(\theta)$ ; cette première série est la série asymptotique dont les premiers termes suffisent à décrire le champ lointain. La série des termes "complémentaires" est utile pour calculer la pression acoustique à distance moyenne. En outre il est possible de calculer les différentes puissances mises en jeu.
SOUND DIFFRACTION BY A MANY-SIDED BARRIER

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The sound field around a many-sided barrier can be obtained for the incidence of a spherical wave with application of Keller's geometrical theory of diffraction[1]. This is given as the sum of the geometrical optics waves and the singly and multiply diffracted waves produced by the straight edges. Then the waves diffracted two or more times by a same edge are much smaller than the waves diffracted by the edge only once and are neglected.

Singly diffracted waves are expressed directly by Kouyoumjian and Pathak's asymptotic solution[2] for a single wedge. Level difference between the solution and the rigorous solution is less than 0.5dB for a wedge of arbitrary exterior angle only if \( r_s \sin \beta, r \sin \beta > \lambda/4 \) [3]. Here, \( r_s \) and \( r \) are respectively the distances from the edge of the wedge to the source and the observing point, and \( \beta \) is the angle between the incident ray and the edge.

Multiply diffracted waves are derived by Keller's theory and Kouyoumjian and Pathak's asymptotic solution[4]. Experimental results prove to be in sound agreement with the calculated results, as shown Fig.1. Consequently, by applying the above method we can very accurately predict and control the propagation of noise around thick barriers, buildings, slits, etc.

Fig.1 Equi-SPL contours, when 500Hz. (— calculated, ----- measured)

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DETERMINATION OF TRAVELLING WAVE COMPONENTS IN ACOUSTIC FIELD
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INTRODUCTION

Cases where opposite travelling waves take place are of particular interest with respect to the examination of acoustical field properties in reflective surroundings. It is often desirable to determine separate characteristics of each of the two components. The method described enables measurements of the waveforms and energies of such wave components to be taken, and can be accomplished by use of conventional equipment.

SEPARATION PROCEDURE

It can be shown that the acoustical pressures \( p_+ = p_+(r_0, t) \) and \( p_- = p_-(r_0, t) \), pertaining to the components travelling in + and - directions, are related to the total acoustical pressure \( p = p(r_0, t) \) through the operator \( L_{+,-} = 1/2[p_1c/\delta r \, dt] \). Here \( r \) denotes the direction of wave motion, \( r_0 \) is the coordinate of the observed point, and \( c \) is the speed of sound. A finite difference technique, applied to \( L_{+,-} \), yields an approximation:

\[
P_{+, -} = \frac{(p_1 + p_2)}{4} + \frac{c}{2\Delta} \cdot \int (n_1 - n_2) \, dt \quad \text{for } \Lambda <
\]

where \( p_{1,2} = p(r_0 + \Lambda/2, t) \), which makes it possible to apply to measurements in practice the result stated above. Thus the time history of either of the components can be determined using two equal pressure sensitive microphones, located perpendicularly to the local wavefront at a small distance \( \Lambda \) apart.

POWER CALCULATIONS

The power spectral density (PSD) functions of \( p_+ \) and \( p_- \) can be obtained on the basis of (1) in terms of the direct and cross spectral densities of \( p_1 \) and \( p_2 \):

\[
S_{+,-} = |z_{+-}|^2 (S_{11} + S_{22}) + 2Re(z_{+-}^2 S_{12}) \quad \text{Re - real part}
\]

via a complex factor \( z_{+-} = 1/4 + ic/(4\Gamma \Delta) \), \( i = \sqrt{-1} \), \( \Gamma \) - frequency). The total power \( \delta^2 \) the components is then obtained either by PSD frequency integration or by the time-averaging procedure:

\[
p_{+,-}^2 \, df = \frac{(p_1 + p_2)^2}{16} + \frac{(c/2\Delta)^2}{2} \int (n_1 - n_2) \, dt \quad \text{for } \Delta <
\]

\[
+ \frac{c}{2\Delta} \frac{p_2/p_1 \, dt}{p_2/p_1 \, dt}
\]

(3)
Analysis of echo signal reflected from continuously varying boundary layer
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In the conventional analysis of ultrasonic pulse echo signals reflected from an object, it is assumed in most cases that the acoustic impedance of the medium varies stepwisely at the boundary layers. Recently some researchers have reported that in some cases the echo cannot be obtained from soft tissues where the boundary layers can be observed visually. Although many explanations will be listed for this phenomenon, one of the most probable cause may be due to the reflection from the continuously varying boundary layer, that is, if the profile of acoustic impedance changes slowly at the boundary, the amplitude of echo signal becomes small and finally we cannot obtain any echo from the boundary if the profile changes sufficiently slowly. In this paper the influence of the profile of boundary layer on the pulse echo signal are examined both analytically and experimentally.

Arrangements for the experiment are shown in Fig.1, where the boundary layer between water and glycerin is formed carefully. Then the profile of acoustic impedance at the boundary changes with lapse of the time due to the diffusion. Consequently by recording the temporal variation of amplitude of echo signal, the influence of the boundary profile on the echo signal can be estimated. In this experiment the ultrasonic frequency of a transducer is 2.25 MHz and in order to avoid the distortion of boundary which may be caused by the incidence of ultrasonic pulse, only one pulse is used for each measurements of the amplitude of echo. One of the experimental results is shown in Fig.2, where the amplitude of echo signal reflected from water-glycerin interface is shown as a function of lapse of time since the diffusion starts. In this figure the amplitude of echo is normalized by the amplitude measured 8 min. after the diffusion starts. Since water must be piled on glycerin calmly, it takes rather long time to complete the boundary.

Construction of a computer model to simulate this phenomenon is also under way.

Fig.1 Experimental arrangements, where t.d. shows transient recorder.

Fig.2 Relation between amplitude of echo reflected from water-glycerin boundary and lapse of time since diffusion starts.
We consider the problem of minimizing the reflectance of planar inhomogeneous layers for normally incident radiation at given layers thicknesses. Six independently adjustable parameters are tuned for optimal results. Approximate theoretical expressions for effective material parameters of inhomogeneous materials are used to describe the coupling of matrix material properties and geometrical effects of random inhomogeneities. Numerical solutions of the relevant two point boundary value problems will be presented. The general solution for inhomogeneous materials includes continuously varying material properties normal to the surface of the layers. Limiting cases, based on homogeneous and inhomogeneous materials yield multilayered systems. In the limit of zero absorption we show that certain optimal multilayered systems consist of sequences of unit cells, each containing a small number of sublayers.
LA THEORIE GEOMETRIQUE DE LA DIFFRACTION APPLIQUEE A L'ETUDE D'ECRANS INDUSTRIELS

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La théorie géométrique de la diffraction, TGD, introduite par KELLER (1), est très utilisée pour traiter la propagation des ondes électromagnétiques. S'appliquant aux domaines où la longueur d'onde est voisine des dimensions des obstacles et de formulation relativement simple, il est intéressant de l'utiliser pour l'étude des écrans industriels.

Cette théorie asymptotique est basée sur le concept des rayons sonores. Les rayons pris en compte peuvent correspondre à une propagation directe, à des phénomènes de réflexion ou à des rayons diffusés dans un cylindre et du champ incident, ce qui donne des coefficients de réflexion et de réfraction des obstacles et des positions relatives source-écran-récepteur. Les rayons diffusés font avec l'arête de l'écran un angle égal à celui du rayon incident avec celle-ci.

Les expressions de KELLER et KOUTOURMIAN (2) pour un écran plan demi-infini permettent le calcul du niveau sonore derrière un écran rectangulaire placé dans un local industriel (plafond bas) (3). Les trajets source-récepteur possibles sont pris en compte en construisant les images du récepteur et de la source par rapport au sol et au plafond. L'énergie associée à chaque rayon est fonction des coefficients de réflexion des parois et du coefficient de diffraction. Des résultats de calcul et de mesure (niveau de pression sonore en fonction de la distance à l'écran) sur une armoire (1/10 à 1/20) sont reportés figure 1.

Un autre exemple d'application de la TGD est le calcul de l'effet de petits écrans tels que ceux placés devant un opérateur. Dans l'exemple reporté figure 2 nous avons évalué l'atténuation du champ direct par un écran trapézoïdal masquant une source ponctuelle en fonction de la fréquence. Les champs sonores provenant des différents trajets possibles ont été combinés en amplitude et en phase.

Pour ces deux types d'application de la TGD, la concordance entre les résultats expérimentaux et numériques est bonne. Pour des situations plus complexes (plusieurs écrans, locaux de forme non simple...), la recherche des trajets possibles peut s'avérer fastidieuse. Nous envisageons d'utiliser une technique numérique consistant à tirer des rayons dans des directions arbitraires et à suivre leur propagation jusqu'à leur collision avec une cellule de réception (4).

(3) LECOCQ, Thèse de doctorat 3ème cycle - Université du Maine 1977
(4) LEBLOND, LECOCQ, INTER-NOISE 78 et Notes Scientifiques et Techniques INRS.
Les mécanismes de diffusion, de réflexion et de transmission des ondes de Rayleigh par les discontinuités sont d'une grande importance pour la caractérisation des défauts superficiels.

Nous présentons nos résultats expérimentaux concernant l'interaction entre les ondes de Rayleigh se propageant sur une surface métallique polie et des défauts de forme géométrique simple. Ces mesures sont basées sur la relation entre le spectre des impulsions large-bande rétrodiffusées et la géométrie de la discontinuité.

Dans le cas de fissures superficielles semi-circulaires dont la longueur est supérieure à la longueur d'onde de Rayleigh \((1/\lambda_R > 1,5)\), l'intensité rétrodiffusée est considérée comme la transformée de Fourier de la distribution des amplitudes complexes dans la fissure.

Dans le cas de fissures rectangulaires de profondeur \(p\) telle que \(p \leq \lambda_R\), l'intensité réfléchie résulte des interférences entre les signaux provenant des bords supérieur et inférieur de la fissure. Les spectres caractéristiques des échos observés présentent des maxima et des minima dont la position en fréquence nous permet d'évaluer, en l'absence d'interférences parasites, les paramètres géométriques caractéristiques de ces défauts avec une précision de l'ordre de 10 %. L'influence de l'angle du dièdre, formé par les plans limitant la discontinuité, sur l'amplitude et la phase des ondes réfléchies et transmises est également discutée.
1. INTRODUCTION

The growing noise-abatement problems in great urban and suburban areas has arisen great interest in screens and wide barriers as a means of protection. Apart from the simple case of the thin reflecting screen there is very little theoretical information in the literature about more complicated cases. The results obtained within the GTD framework are valid in principle in the high frequency limit; therefore it should be interesting to compare this approach with an "exact" numerical solution of the boundary value problem. Moreover, there are very few quantitative results concerning combined effects of ground and impedance layers on the barrier and so on. We like to present here an approach combining those parameters.

2. A FULLY NUMERICAL APPROACH

Consider the following boundary value problem for the pressure field:

\[(\Delta + k^2) \varphi (r) = f (r)\]

+ boundary cond. on ground ; + boundary cond. on screen ; + Sommerfeld condition.

If \(G_0\) is the Green's function of the problem in the absence of the barrier, it may be shown that \(\varphi\) is given in terms of an integral equation on the barrier:

\[\varphi (r) = \int f (r) G_0 (r, r') dr' - \int \frac{\partial G_0 (r, r')}{\partial n} (r, r') + \gamma (r') G (r, r') \] \(\varphi (r')\)

Here \(\gamma\) is simply \(-\frac{ik}{Z}\) where \(Z\) is the point impedance on the screen.

Once the Green's singularities extracted, the equation is suitable for a boundary elements treatment: the field \(\varphi\) is expanded on a simple basis of functions supported by small intervals; one should integrate once more over the barrier to get rid of eventual angular points, and then solve a linear system of equations for \(\varphi\).

This yields a fully convergent algorithm we tested the solutions obtained with some well known field problems, and others less simple computed with GTD formulas and it proved very efficient: for a thin reflecting screen on a reflecting ground, or a trapezoidal, the accuracy was always less than one dB for a very reasonable computing time.

3. CONCLUSIONS

The boundary elements method may be used extensively for computing diffraction patterns of different barrier-shapes with various acoustical properties.

When all the distances considered are greater than five wavelengths the results are in full agreement with GTD calculations. Beyond that we expect that the GTD should yield wrong interpretations.
Experimental Study on Phase Velocity of Ultrasonic Wave Decaying in Space and in Time

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INTRODUCTION

It has been pointed out that phase velocity of sound propagating and attenuating in space is intrinsically larger than that of sound decaying in time. Among ultrasonic techniques currently used, pulse method or interferometer observes the former, while Brillouin scattering, the latter. The difference between the two phase velocities is, however, usually too small to be detected experimentally.

In this work, we used a modified system of high-resolution Bragg reflection to measure both the spacial and the temporal waves, and succeeded in observing the difference.

EXPERIMENT AND RESULT

The laser beam scattered by sound wave in the liquid is detected with the optical heterodyne system shown in Fig.1. If the receiving frequency of the spectrum analyzer is held and the rotating table is swept, then sound wave with well-defined frequency is observed. The obtained k-spectrum due to spacial decay gives \( v_B \), the phase velocity of spacial wave. On the other hand, if the table is fixed and the frequency is swept, \( \omega \)-spectrum of the sound with well defined wavenumber is obtained, yielding another phase velocity, \( v_T \).

Measurement was made at 20°C in liquid furan, which has a strong dispersion centered at 100 MHz. Peak angle of the curve in Fig.2(a) gives the central wavenumber which yields \( v_B = 1205 \text{ m/s} \) at \( f = 100.926 \text{ MHz} \). The curve of (b) shows the central frequency is by 527 kHz lower than \( f \) giving \( v_T = 1198 \text{ m/s} \). The result is in good agreement with the theoretical predictions.

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Laboratory measurements of shear wave attenuation as a function of frequency were made using recently developed ceramic bimorph bender transducers to excite transverse particle motion in a medium grain water-saturated sand. The measurements were made at 13 frequencies from 450 to 7000 Hz. Multi-cycle sine-wave pulses were used to insure steady state vibration at the measurement frequency. Attenuation was determined from the slope of a linear least-squares fit to the maximum received level versus transducer separation data. This not only affords an estimate of the attenuation but allows confidence intervals to be placed around that estimate.

The attenuation values increased with frequency but did not exhibit a simple first-power dependence on frequency. The results were compared with predictions based on a two-component model developed by Stoll and were consistent in both amplitude and frequency dependence. This work was supported by the Office of Naval Research.
The Reflection of Acoustical Transients from Fibreglass Batts

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Experiments were conducted on the reflection of an acoustical transient from a surface of particle board (19.2 mm thick) and a fibreglass batt (50.8 mm thick) for various source and receiver geometrics above the surfaces. The fibreglass was backed by the particle board. The experiments were simulated on a mini-computer using a corrected plane wave reflection theory. The results obtained indicated that the simple theory was adequate for predicting the transient sound field for most source and receiver geometrics.

A spark generator provided the required input for the experiments. A B & K 1/8 inch type 4138 microphone was used as the receiver. Both source and receiver heights were varied independently from a minimum of 0.02 m up to a maximum of 1 metre from the surfaces, in increments of 0.2 m.

The simulation involved the construction of a reflected pulse after the appropriate time delay and a reduction of its pressure amplitude according to the inverse square law. The reflected wave was appropriately modified if the surface was absorbing. The modification included a manipulation of the Fourier Transform of the wave. The Fourier Transform was numerically evaluated over a frequency band from 200 Hz to 18 kHz in increments of 200 Hz. Each component of frequency was subsequently multiplied by the complex reflection coefficient to yield the transformed reflected wave from the surface. Air absorption effects were corrected for using the method described in ANSI SI 26-1978. The reflection coefficient was obtained by measuring the normal specific acoustic impedance of the fibreglass in an impedance tube. The time representation of the reflected wave was derived by the inverse Fourier Transform.

The results were as follows. For the particle board - the shape of the reflected wave was identical to that of the direct wave. A reduction in amplitude however was observed and this corresponded to that resulting from geometrical spreading - a fact confirmed by the simulation. For near grazing incidence the amplitude of the pulse was doubled. The fibreglass surface - the reflected wave was of reversed phase with the incident wave. Its amplitude was substantially smaller than that from the hard surface. However the times of arrival of the reflections were identical to that obtained from the hard surface reflections. At near grazing incidence, a grossly reduced wave was observed.

The simulation provided accurate predictions of the reflections for source and receiver heights of 0.2 m and greater for both surfaces.
Reflection of narrow beam sound wave from a rigid cylindrical tube.

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Introduction

Formula of target-strength is ordinary defined in uniform sound field and
can not be applicable if a beam width is narrower than dimensions of
targets. Then approximate equation of reflection of narrow beam sound
wave from a rigid cylindrical tube is derived and compared with experi-
mental result.

Theory

Take the orthogonal coordinate system as shown
in Fig. 1. When scanning a disk type trans-
ducer in parallel with X axis within X, Z
plane including a cylindrical axis, narrow
beam target strength \( TSN(x_0) \), as a function of
transducer position \( x_0 \), is deduced by summation
of back-scatter waves from each part of cylinder
as the following formula.

\[
TSN(x_0) = \left( \frac{P_r \psi^2(\gamma)}{PL} \right)^2 = \frac{k \alpha \cos \theta}{4\pi} \int \psi^2(x-x_0) e^{-ikx\sin \theta} \, dx \]

\( k = \frac{2\pi}{\lambda}, \beta = \frac{\lambda}{\cos \theta}, P_r; \) sound pressure in the
direction of main beam at distance to the target.
\( P_l; \) reflected sound pressure at unit distance,
\( \psi(\gamma); \) directivity function of
a transducer.

Experimental Result

For the case of 6dB down beam
width by disk type transducer
10cm in diameter, at 50cm
distance, is approximately
half of iron cylindrical tube
length in water at cylindrical
tube, theoretical and experi-
mental narrow beam target-
strength \( TSN(x_0) \) as a parameter
of slant angle \( \theta \) is shown in
Fig. 2. Experimental values
agree well with theoretical
results within the experimental
errors.

Fig. 1 Arrangement for
the analysis.

Fig. 2 Measured and theoretical narrow
beam target strength of the rigid
cylindrical tube as a function
of transducer position \( x_0 \).
An elastic cylinder, circular in section and infinite in length, containing a concentric filler, is considered in infinite acoustic fluid. Sound pulse with plane front and infinite duration impinge upon the cylinder. A two-dimensional transient problem of the analysis of the acoustic field produced by an incident pulse is investigated. For solving the problem Fourier transformation with respect to time and the method of separation of variables is used. Numerical results are computed in case of a steel cylinder immersed in water and filled with fused silica. For an observation point on the ray $\theta = 0$ and far distances from the cylinder $r = 10^4$ acoustical pressure may be described in such a way. A single incident pulse produces a series of multiple echo returns. The first pulse is caused by specular reflection and refraction, and the subsequent pulses by reradiation. For the chosen observation point, even for such moderate wavelengths as $\lambda \sim (1+1/4)r$, it appears that the principle of isolated element is valid: for short time, until the appearance of refracted pulses, the amplitude of pressure in a scattered field may be found by the multiplication of pressure calculated in case of a rigid immovable cylinder on the reflection coefficient evaluated for the case of plane boundary division of liquid and elastic media at normal incidence. It is found that the contribution of refracted pulses is rather small, the same as that of the shear waves. The second and subsequent pulses are chiefly caused by the radiation of Rayleigh-type waves that are excited and radiated at the critical angle. By using time repeated pulses, one can calculate both the speed of Rayleigh-type surface wave and the critical angle. It is found that both of them considerably depend upon the elastic properties of the filler. The calculated pulse sequence is shown in the figure.
ON THE APPROXIMATIONS OF KIRCHHOFF'S BOUNDARY CONDITIONS

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The approximations of Kirchhoff's boundary conditions, that are the basis of Kirchhoff's diffraction theory, are frequently available to evaluate a diffracted or a reflected sound field.

\[
\begin{align*}
\text{on } A & \quad \phi = \phi^i, \quad \frac{d\phi}{dn} = \frac{2\phi^i}{\lambda} \\
\text{on } B & \quad \phi = 0, \quad \frac{d\phi}{dn} = 0
\end{align*}
\]  

(\(\phi^i\) is an incident wave, \(\phi^c = e^{ikx}\), \(k = \frac{\omega}{c}\).

\(\phi\) represents the velocity potential and \(\phi\) and \(\frac{d\phi}{dn}\) are proportional to the sound pressure and the particle velocity, respectively.

Strictly speaking, the particle velocity near the edge of the screen is disturbed and the velocity potential is not null behind the screen. The particle velocity near the edge of the plate and also the velocity potential behind the screen are exactly calculated and the difference of the above approximations is evaluated.

Now, the sound field around a rigid plate is given by the following formula (See Fig.2),

\[
\phi(x) = \phi_d(x) + \frac{1}{4\pi} \int_{F} \phi \frac{e^{ikr}}{2\pi r} \, ds
\]  

where \(\phi = \phi_d - \phi_c\), \(\phi_d = e^{i\phi_d}\), \(P_r\) denotes a point source, \(\phi\) the surface area of a plate and \(\phi\) is a unit normal to \(F\). \(\phi_c\) and \(\phi\) that are unknown functions, represent the velocity potentials of each side of the surface \(F\). Eq.(2) is differentiated with \(\phi\), converging \(\phi\) on a surface and then the following integral equation with \(\phi\) is obtained.

\[
\frac{1}{4\pi} \int_{F} \frac{2\phi}{2\pi} \frac{e^{ikr}}{r} \, ds = 0 \quad \text{Pe} \, F
\]  

where \(\frac{2\phi}{2\pi}(\frac{e^{i\phi}}{r}) = \frac{2\phi}{2\pi}(1 - ikr)\), because the plane is plane. Eq(2) gives the following representation, i.e.

\[
\begin{align*}
\text{on } A & \quad \phi = \phi^i, \quad \frac{d\phi}{dn} = \frac{2\phi^i}{\lambda} \\
\text{on } B & \quad \phi = \phi^i - \frac{\pi}{2}, \quad \frac{d\phi}{dn} = 0
\end{align*}
\]  

An example of the particle velocity on \(A\) near the edge in case of a circular disc is shown in Fig.3.
L'observation et l'étude des champs acoustiques rayonnés par des objets quelconques (sources artificielles, machines et installations industrielles, ...) posent le problème de leur représentation. Celle-ci doit être une représentation d'onde (spatio-temporelle) la mieux adaptée au problème considéré. Théoriquement, elle est solution de l'équation aux valeurs propres de l'opérateur de rayonnement et propagation. En fait, l'existence des conditions aux limites et de rayonnement doit être complétée par la structure forcée des champs mécaniques moteurs internes, qui conditionnent partiellellement l'état vibratoire de l'enveloppe rayonnante du dispositif étudié. Ces remarques conduisent à représenter les ondes acoustiques observées à distance sur une famille de surfaces d'ondes (caractéristiques), dont on isole les composantes sur l'une de ces surfaces. Pratiquement, cette surface sera "confinée" et permettra la réalisation d'une "antenne". Il apparaît alors que les seules mesures physiques que l'on sait réaliser sont de type énergétique et imposent ce caractère de représentation. Celle-ci caractérisera le "flux" énergétique traversant l'antenne, associé aux ondes directes et inverses composantes. Le confinement spatial et temporel conduit à définir par analogie avec la mécanique quantique des "paquets d'onde" localisant l'énergie dans les domaines conjoints temps et fréquence, espace et fréquence spatiale.

A titre d'exemple, les auteurs indiquent comment une telle représentation est possible tant en champ proche que lointain. Deux exemples tirés, l'un de l'acoustique des salles, et l'autre de l'imagerie spatio-fréquentielle à distance proche et lointaine de bruiters industriels ou de véhicules, illustrent cette approche.
INTRODUCTION
Le principal objet du travail est de décrire les champs proches créés par les haut-parleurs électrodynamiques (à diaphragme conique) dans des conditions usuelles de fonctionnement, c'est à dire en présence d'un écran de dimensions finies, l'onde arrière étant absorbée ou non par une enceinte. L'étude théorique du problème repose sur la résolution de l'équation intégrale de Helmholtz-Huygens pour le rayonnement et sur un modèle, décrivant le comportement du diaphragme, issu de la théorie des coques. Nous avons ainsi accès aux courbes en fréquence, pour une excitation électrique donnée.

LE RAYONNEMENT DU HAUT-PARLEUR
L'expression de la pression sonore est la somme de deux termes, le premier correspondant au champ sonore créé par le haut-parleur en l'absence d'écran et le second traduisant la contribution apportée par l'écran. L'introduction d'une impédance ponctuelle de rayonnement approchée dans l'équation intégrale de Helmholtz-Huygens, permet d'exprimer le premier terme en fonction de la vitesse ponctuelle du diaphragme. Le second terme est obtenu en fonction de la pression sur chacune des faces de l'écran; dans le cadre des approximations faites, cette pression précise est donnée par celle qui s'exercerait sur un écran infini: La pression sur l'arrière de l'écran est donnée par une intégrale de Rayleigh et, pour exprimer la pression sur l'avant, du côté de la concavité du cône, nous avons retenu la formulation de la diffraction de Rayleigh-Sommerfeld.

Nous avons également calculé le rayonnement du haut-parleur, dans les diverses conditions vues précédemment; mais cette fois-ci lorsque l'onde arrière est absorbée par une enceinte close.

Nous avons vérifié expérimentalement les résultats théoriques trouvés en introduisant dans les calculs, pour chaque fréquence, la répartition de vitesses sur le diaphragme obtenue en visualisant les vibrations de celui-ci au moyen d'un banc d'holographie optique.

LA REPRÉSENTATION DU HAUT-PARLEUR. LA COURBE DE RÉPONSE
La recherche d'un modèle pour le diaphragme part de l'équation différentiel d'ordre 4 du mouvement des coques coniques, admettant l'hypothèse d'une symétrie de révolution et tenant compte de la force exercée par la charge acoustique. La solution générale est une combinaison linéaire de quatre fonctions de Bessel d'ordre zéro. On écrit que la bobine mobile est encastrée au voisinage du sommet du cône, et que la suspension périphérique est caractérisée par son impédance mécanique. A l'issue des calculs (admettant les développements des fonctions de Bessel autour de l'origine), on montre que l'admittance motionnelle est l'expression exacte de l'impédance électrique d'un circuit de type admittance, quadripôle en saturé par l'impédance de rayonnement et faisant intervenir les impédances mécaniques de la suspension périphérique et de l'ensemble bobine-mobile-spider, le raidisseur de flexion et la masse du diaphragme. Ainsi, nous avons pu calculer, pour une impédance de rayonnement et une excitation électrique données, la vitesse de la bobine mobile et ses variations avec la fréquence. Ce résultat, associé à ceux relatifs au rayonnement a permis le calcul des courbes de réponse du haut-parleur. L'accord avec les résultats expérimentaux justifie a posteriori les approximations retenues.
TRANSIENT RADIATION FIELD FROM AN ELASTIC CIRCULAR PLATE EXCITED BY SOUND PULSE.

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INTRODUCTION
In this paper, an approximate solution of the on-axis transient radiation field from an elastic clamped circular plate in a baffle is developed analytically in the far-field when the triangular plane sound pulse (Fig.1) impinges normally on the plate, and then process of the transient sound radiation is discussed in terms of the waveform radiated.

THEORETICAL BASES
The transient solution is obtained by inverse Fourier transform of the frequency function of this pressure field, which has been given by the usual normal mode method[1]. On the assumption of \( \rho c \) for the radiation impedance per unit area of the circular plate (called \( \rho c \)-approximation, where \( \rho c \) is the characteristic acoustic impedance of the air), the waveform is given as follows:

\[
\phi(t) = \frac{\rho c^2}{mz} P \sum_{n=1}^{\infty} \left[ S_n x y_n(t) \right],
\]

where

\[
y_n(t) = \begin{cases} f_n(t), & T \geq t \geq 0, \\ f_n(t) - 2 f_n(t-T), & 2T \geq t \geq T, \\ f_n(t) - 2 f_n(t-T) + f_n(t-2T), & t \geq 2T, \end{cases}
\]

and \( f_n(t) \) is given as follows (\( A = \rho c/m \omega_n \) : the \( n \)-th resonant angular frequency of the plate):

\[
f_n(t) = \begin{cases} e^{-At \sin \sqrt{\omega_n^2 - A^2}} t/\sqrt{\omega_n^2 - A^2}, & \omega_n > A, \\ e^{-At \sinh \sqrt{A^2 - \omega_n^2}} t/\sqrt{A^2 - \omega_n^2}, & \omega_n < A, \\ te^{-At}, & \omega_n = A, \end{cases}
\]

where \( A \) and \( m \) are the radius and the surface density of the plate, respectively, \( z \) the distance from the plate to observed point, \( S_n \) the function of the eigen value of the \( n \)-th mode of the plate. The quantity \( A \) measures the air loading on the plate caused by the sound radiation.

RESULTS AND CONCLUSIONS
The waveforms are examined, with the material constants of the plate varied. It is noted that the first shock of the waveform (\( 2T \geq t \geq 0 \)) is mainly represented by the piston-like vibration of the plate, while the successive part following the first shock is caused by the flexural vibration, and that as the rigidity of the plate decreases, the whole waveform becomes to be approximated by the piston motion of the plate, while as it increases, the oscillatory waveform is obtained. In spite of the simple expression for the radiation impedance of the plate, this \( \rho c \)-approximation gives fairly good results for the waveform.

REFERENCE
Synthesis of cylindrical sound radiators on ground of given far-field

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One of the most important practical radiation problem is to realization of cylindrical radiator /sound column/ with given directivity pattern. We can consider this as solved problem if succeed to determine that surface velocity which generates the given directivity pattern in the far field. The directivity pattern in the far field, in the horizontal plane we can give with help of a finite series as follows:

$$F(\alpha) = \sum_{n=0}^{N} A_n \cdot \cos(n\alpha)$$  \hspace{1cm} (1.)

where: $\alpha$ is the angle measured in the horizontal plane; $A_n$ are given coefficients.

Moreover, let us assume that the particle velocity on the radiator's surface has the following distribution:

$$\phi(\varphi) = \sum_{n=0}^{N} B_n \cdot \cos(n\varphi)$$  \hspace{1cm} (2.)

Then if we multiply (2) by the sound field of the line source set on the surface of a rigid cylinder, and integrate over $0 \leq \varphi \leq 2\pi$, we get through comparison of the series the next form:

$$B_n = i^n \cdot A_n \cdot \hat{\mu}^{(2)}_n(ka)$$  \hspace{1cm} (3.)

Where $i = \sqrt{-1}$; $\hat{\mu}^{(2)}_n(ka)$ is the first derivative of the second kind Hankel-function, $k$ is the wavenumber, and $2a$ is the diameter of the cylinder.

It will be given some measuring result of sound columns which were constructed on ground of this theory.
ON SOURCE RADIATION
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The power output from given sources is usually ascertained via an energy flux integral over the normal directions to a remote (far field) surface; an alternative procedure, which utilizes an integral that specifies the direct rate of working by the source on the resultant field, is described and illustrated for both point and continuous source distributions. A comparison between the respective procedures is made in the analysis of sound radiated from a periodic dipole source whose axis performs a periodic plane angular movement about a fixed direction. Thus, adopting a conventional approach, Sretenskii (1956) characterizes the rotating dipole in terms of an infinite number of stationary ones along a pair of orthogonal directions in the plane and, through the far field representation of the latter, arrives at a series development for the instantaneous radiated power, whereas the local manner of power calculation dispenses with the equivalent infinite aggregate of sources and yields a compact analytical result.
ON THE DETERMINATION OF PARAMETERS OF SHELLS BY ECHO-SIGNALS

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INTRODUCTION

The diffraction of acoustic waves from a spherical or cylindrical shell immersed in a liquid is considered. The distant determination of the parameters of the shell by means of echo-signals is investigated theoretically. The governing system of equations for solving the inverse problem is analysed in the case of a cylindrical shell.

RESULTS

The echo-signal from a thin shell consists mainly of a reflected pulse and of some series of radiated pulses generated by the elastic waves propagating in the shell and in the surrounding liquid. The amplitudes, arrival times and form differences of the echo-pulses in comparison with the incident pulse are taken as the governing parameters of echo-signals. On the other hand, the parameters of a shell are: the radius, the thickness and the constants of the material. The correlations between the parameters of echo-signals and the parameters of the shell, which will later be used in solving the inverse problem, are found theoretically. The Timoshenko-type shell theory is used for a mathematical model of the shell and the surrounding liquid is described by means of linear wave theory. The integration is based on Fourier and Watson transforms.

The correlations between the shell and the echo-signal's parameters are very complicated, therefore on the basis of calculated correlations found by special programmes for many shell parameters' combinations the more simple empiric formulae are obtained by means of polynomial or exponential approximation. The set of these formulae gives the system of equations for the determination of the shell parameters. The analysis shows this system being stable and the errors not exceeding 8% for the density and 3% for the other parameters. However, this solution of this system is not singular, but in the case of well-chosen initial approximations the solution is converging to the right one. The problem how to choose a good initial approximation is shortly discussed.
A THEORY OF RESPONSE FUNCTION RECIPROCITY

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In a previous paper (ASME Publication 79-WA/Saf-2) experimental evidence for the reciprocity of reverberation times was presented and discussed. This paper presents a theory to account for the observations.

If in an enclosed, partially enclosed or unenclosed space, a sound source at point A be discontinued, the reverberation time determined at point B is equal to that which would have been determined at A had B been the location of the source.

The following conditions govern the validity of this proof -
a) sound travels from point A to point B and from point B to point A along corresponding discrete paths,
b) it does so in equal times,
and, c) the attenuation coefficients (α) are equal in both directions.

At time t, the sound power (w_{Bi}) arriving at B is the sum of an infinite series of weighted, earlier, sound powers emanating from A(w_A), thus,

$$w_{Bi} = \sum_{j=1}^{\infty} \alpha_j \cdot w_A(i+j-1)$$  \hspace{1cm} (1)

where some of the attenuation coefficients are normally zero.

Let the interval between terms of the series be a common factor of the times for all transmission paths for which α_j is not zero. The w_{Bi}s may be represented as a sequence of sound powers with the same interval;

$$w_{B1} = \alpha_1 w_A1 + \alpha_0 w_A0 + \ldots$$
$$w_{B2} = \alpha_1 w_A2 + \alpha_0 w_A1 + \ldots$$
$$\vdots$$
$$w_{Bi} = \alpha_1 w_Ai + \alpha_0 w_A(i-1) + \ldots$$

\hspace{1cm} (2)

In reverberation time measurement, the w_A's (on the R.H.S. of Eq.(2)) are zero to some term and have the value 1 (say) for the rest of the series, to infinity ideally. In another method, only one w_A is non zero. In all cases, however, the pattern is repeated in the next row of Eq.(2), but with a shift to the right and with the new attenuation coefficients.

When the source is at B and the receptor at A,

$$w_{Ai} = \sum_{j=1}^{\infty} \alpha_j w_B(i+j-1),$$  \hspace{1cm} (3)

and a sequence in w_A corresponding with Eq.(2) ensues.

These two sequences are proportional, and since the intervals are equal, the reverberation times are equal. This result is easily extended to other response functions.
"R-MATRIX THEORY OF ELASTIC SCATTERING FROM FLUID SPHERES IN SOLIDS."

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We consider the cases of acoustic and elastic wave-scattering from fluid (possibly viscous) spheres contained within either solids (i.e., underwater sound attenuators) or dissimilar fluids (i.e., bubbles in liquids), and analyze the resulting scattering matrix $S_n$, and the poles appearing in the associated scattering amplitudes. The one-term Taylor series we used in the past (Refs. 1, 2) to obtain the "one-level approximation" to $S_n$, which was to useful in our earlier development of the Resonance Theory of Viscoelastic Wave-Scattering from cavities (Ref. 1), can now be replaced by the exact Mittag-Leffler expansion of the modal mechanical impedance of the system. Since all the singularities of this impedance are simple poles, it admits a meromorphic function representation in terms of a Mittag-Leffler series expansion. The reduction of the corresponding S-matrix to that obtained in the one-level approximation (Refs. 1, 2) provides a shift between the poles of $F_n$ and those of $S_n$, and also introduces a width. Absorption causes further shifts and the unitarity property of $S_n$ no longer holds. The new representation is not only exact, but reduces properly to the right asymptotic limits and our earlier "backgrounds" and "resonances" can be re-interpreted in the light of those new (and simpler to obtain) poles. The plot below shows some of these poles in three dimensions plotted versus $n$ and $k_d$. The families of "resonance spikes", their size and spacings, the ridges they form, their cleavage and inclination, all quantitatively help to explain the resonance scattering phenomenon taking place around the cavity in the perforated solid. (H. Ü. was additionally supported by Code 421 of ONR.)

References:
COUPLING OF A HARTMANN-SPRENGER TUBE WITH A HORN

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The Hartmann-Sprenger tube (H.S. tube) is a repetitive shock-tube in which strong flow oscillations are generated at nearly the acoustic frequency of the tube. The tube is excited by a high speed gas jet. When a needle is mounted on the axis of the jet nozzle, an excellent emission stability is achieved. For this so-called "needle" generator, it may be shown that the amplitude of the pressure pulses is approximately equal to $2\gamma M_j p_a$, where $\gamma$ represents the specific heat ratio of the gas, $M_j$ the jet Mach number and $p_a$ the ambient pressure. For $M_j = 2$, this lead to a pressure amplitude of 6 bars and makes of the H.S. tube a powerful sound generator. It has therefore been considered to use this device as an acoustical warning system for trains or ships.

For a given application, the frequency, the driving pressure and the air consumption are generally given. Whereas the frequency determines the length $L$ of the tube, the driving pressure and the consumption determine the nozzle diameter of the jet and, consequently, the H.S. tube diameter $D$. It follows that the ratio $L/D$ is imposed and is generally much greater than unity. For these ratios, the acoustic efficiency of the H.S. tube alone is quite low (1). To increase the efficiency at large $L/D$ ratios, it is necessary to couple the device to a horn.

![Diagram](image)

N: needle

L

The paper shows the results of an experimental and theoretical investigation on the acoustic coupling of a H.S. tube to a horn. One of the most interesting features observed is a frequency jump as the tube length is varied continuously. Measurements have been carried out in an anechoic chamber. Sound intensity levels up to 130 dB have been recorded at a distance of 3 m from the source and acoustic efficiencies up to 7% have been obtained.

1) BROCHER, E. Contribution à l'étude des générateurs acoustiques à jet d'air, ACUSTICA, 32 (1975), 227.
ON THE PIEZOELECTRIC TRANSDUCERS FOR GENERATING HIGH-FREQUENCY ULTRASONIC POWER

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High-frequency intense ultrasound above 100 kHz exhibits outstanding effects in activating chemical reactions, internal layer heating of viscoelastic materials, atomizing various liquids, etc. However there exists very few examples of industrial applications since ultrasonic transducers suitable for generating high-frequency ultrasonic power are not available. One of the problems is that transducer diameter is large compared to the wavelength and as a result coupled modes of radial and thickness vibrations are strongly excited. Large cross-section is required independent of frequency to generate large amount of ultrasonic power, whereas wavelength decreases with increasing frequency. To develop optimal transducers, vibration characteristics of piezoelectric transducers are investigated in relation to their configurations.

Figure 1 shows the configurations of piezoelectric-ceramic transducers tested in the experiment. Vibration characteristics are evaluated through vibration amplitude measurement by the use of laser-beam heterodyne method as well as admittance measurement. Figures 2(a) and (b) show the radial distributions of vibration amplitude of thickness direction measured at resonance frequencies of thickness and second radial modes, respectively. Vibration amplitude and admittance increase in order of A, B, C in Fig. 2(a), and in order of B, A, C in Fig. 2(b).

It was concluded that bevelling circular edges of disk-type transducers is effective for reducing radial vibrations and increasing thickness vibrations, and making a hole in the disk for increasing both of thickness and radial vibrations.

![Fig.1 Configuration of transducers](image1)

![Fig.2 Radial distribution of vibration amplitude of thickness direction measured by laser-beam heterodyne method](image2)
ON ACOUSTIC CAVITATION NOISE SPECTRA

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A digital data acquisition and processing system has been developed to study acoustic cavitation noise.

The cavitation noise is picked up by a microphone and directly digitized (after suitable analog filtering to avoid aliasing) at conversion rates up to 2 MHz. The data are stored in a shift register of 128 k bytes and analysed by using the fast fourier transform. The development in time of the noise spectra is obtained by transforming 4 k bytes and shifting the 4 k bytes over the 128 k bytes stored. A software package has been written to plot the spectra successively or to plot selected spectral lines as a function of time. Also computer films have been produced to give an even better impression of the time development of the spectra when the input to the transducer generating cavitation is altered. This alteration can also be programmed and is then executed under computer control.

Complex measurements are possible with this system. E.g., we have measured the time development of the noise when the sound pressure amplitude is raised inside a hollow piezoelectric cylinder totally submerged in water. Bursts of spectral lines were observed near the cavitation threshold with a strong spectral line at half the driving frequency and some noise around it. Also regions of spectral lines seem to oscillate in strength. At higher driving voltages beyond the cavitation threshold the \( \frac{n}{2} f_o \)-spectral lines \( (n = 3, 5, 7, \ldots, f_o = \text{driving frequency}) \) attain the same strength as the normal harmonics \( mf_o, m = 1, 2, \ldots \), and some indication of \( \frac{n}{4} f_o \)-spectral lines \( (n = 1, 3, 5, \ldots) \) are found.

The spectra can be modeled surprisingly well when adding up the sound pulses from a field of bubbles of different sizes.
A RANDOM WALK MODEL OF LIGHTNING MAPPED INTO THE SOUND OF THUNDER

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A rigorous nonlinear analysis of the lightning→thunder process is practicable only by modelling the lightning as a rectilinear vertical stroke. Known solutions for such a model show but a single thunderclap; this is applicable solely when the observer is very close to a cloud-to-ground stroke. In order to obtain the rumble and roll of thunder, the tortuosity of the lightning must be considered. A rigorous analysis is no longer feasible, and alternate approximate methods have to be sought.

In the analysis reported here a tortuous lightning stroke of specified shape is modelled as a piecewise linear distribution of impulsive point sources (or, point explosions); an N-shaped wave of pressure ('sonic boom') is assumed to evolve by nonlinear propagation from each point. These N-waves can be analytically integrated, following linear acoustics, to yield the pressure signature radiated from one line segment as a function of observer distance, the angle subtended to him, the length of the segment, the duration of the N-wave, and the retarded time (cf. Wright-Medendorp in context of sparks). The computer sums the signatures from all the lightning segments, keeping track of their respective arrival times, to produce the extended pressure-time history of the thunder signature. These signals are then processed through a real-time D/A converter-amplifier-loudspeaker system to produce audible synthetic thunder. A tape recording will be demonstrated if facilities permit.

Frequency considerations dictate that the lightning stroke must possess a fine structure (e.g., 3 m. segments) not resolvable from photographs (40 - 60 m.); also we need the full 3D shape of the channel. Thus we have resorted to a computer to generate lightning channels as a random walk or Monte Carlo process with step lengths averaging 3 m. Repeated trials with different deflection statistics, including schemes of vertical bias and "memory smoothing", were made; these led eventually to satisfactory simulation of real lightning channel configurations: their general appearance appeared very realistic, and deflection histograms for 60 m. segments closely matched those of real lightning. Having generated the 'lightning' channel, the computer algorithm described earlier is then applied to compute the corresponding 'thunder' that would be heard at a specified location on the ground.

The predicted thunder (i) is compatible in its fine scale features (e.g., ~3 m. segments) with small-scale experimental results for sparks (Wright-Medendorp); (2) is compatible with rigorous nonlinear analysis (Plooster) for the specialized case of a rectilinear lightning channel; (3), (4) is judged to be realistic in both audible sound and waveform appearance. ("...human beings can recognize patterns in ways no contemporary computer can begin to approach", F.H.C. Crick, Sci. Amer., Sept. 1979).

(For an earlier version of the lightning→thunder model see Ribner et al, Prog. in Aero. & Astro 46, 77-87 (1976)).
An acoustic signal that propagates through the atmosphere or the ocean will interact with the temperature (or velocity) microstructure resulting in a scattered field with a directional intensity spectrum that can be related to the statistics of the microstructure. In the reference paper a normal mode theory has been formulated for estimating the loss of spatial coherence (Fourier transform of directional intensity spectrum) of acoustic signals propagating through a randomly inhomogeneous waveguide. The theory is applicable into the multiple-scatter region. The horizontal resolution limitation of an acoustic signal can be related to a characteristic distance defined by the transverse decay of this coherence function.

In this paper we investigate the resolution limitation caused by scattering of an acoustic signal propagating through a waveguide with index of refraction fluctuations having a horizontal power spectrum that follows a simple power law or a linear sum of such power laws. This is a realistic power spectrum for many situations of interest. For example, experimental data indicates that the ocean temperature microstructure gives rise to a random medium that can be described by such a combination of power laws.

Reference

ELASTIC WAVE PROPAGATION AT LINEAR SLIP INTERFACES

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INTRODUCTION

A generalization of a bonded interface between two elastic media that retains all linearity properties is a linear partially bonded slip interface across which the traction, \( \mathbf{t} \), must be continuous but the displacement discontinuity, \( \Delta \mathbf{u} \), satisfying a slip condition of the form:

\[
\Delta \mathbf{u} = \eta \mathbf{t}
\]

\( \eta \) is the interface compliance matrix which must be symmetric and positive definite and of dimension length/stress. For isotropic interface behavior, \( \eta \) is diagonal with \( \eta_N \) the normal compliance relating the normal components of \( \Delta \mathbf{u} \) and \( \mathbf{t} \), and \( \eta_T \) the tangential compliance relating the tangential components. The compliances may be real or complex, frequency dependent corresponding to elastic or viscoelastic slip interfaces.

PLANE WAVE REFLECTION AND TRANSMISSION

At a plane interface, reflection and transmission coefficients for incident P, SV, or SH waves may be calculated as functions of the compliances. Anisotropic interface behavior, as when tangential slip can occur in one tangential direction and not in another, implies that any incident wave in medium 1 will in general excite P, SV and SH reflected waves and transmitted waves in medium 2. Isotropic interface behavior implies the usual separation into plane strain and antiplane strain cases. In this case the reflection and transmission coefficients for SH waves and for normally incident P and SV waves are of the form:

\[
R = \frac{-Z_1Z_2+i\omega nZ_1Z_2}{Z_1+Z_2-i\omega nZ_1Z_2}, \quad T = \frac{2Z_1}{Z_1+Z_2-i\omega nZ_1Z_2}
\]

where \( \eta \) is the appropriate compliance and \( Z_1, Z_2 \), the appropriate acoustic impedances. Note, for real \( \eta \), that \( R, T \) satisfy \( |R|^2+|T|^2=1 \). For a pure viscous interface, having the form \( i\xi/\omega, \xi \) real and positive, it may be seen that \( R \) and \( T \) are frequency independent but \( |R|^2+|T|^2<1 \), representing energy loss at the interface. The full plane strain case is more complicated but analogous results have been derived.

PHYSICAL INTERPRETATION AND APPLICATIONS

Wave behavior across a low impedance layer of thickness, \( h \), perfectly bonded between two semi-infinite elastic media approaches the behavior across a partially bonded interface as the ratio of \( h \)/wavelength becomes small if \( \eta_M = h/G \) and \( \eta_N = h(K+4/3 \ G) \) where \( K \) and \( G \) are bulk and shear moduli, respectively. This gives a physical interpretation to the compliances in terms of the physical interface properties. The linear slip interface may be used to model bonds in laminates, fluid saturated fault zones for seismic interpretation, and cracks due to partial melt. The influence of interface compliance on reflections from buried interfaces, on Love wave dispersion and on Stoneley wave propagation and attenuation is discussed.
FINITE ELEMENT ANALYSIS FOR PULSE PROPAGATION IN A FLUID SOLID SPHERE

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FINITE ELEMENT ANALYSIS FOR PULSE PROPAGATION IN A FLUID SOLID SPHERE

A general purpose finite element structural analysis computer program which solves the equation of elasticity $\nabla^2 \gamma + g = a \gamma + b \partial \gamma / \partial t$, where $\nabla^2$ is the Laplacian operator, dots denotes partial time differentiation, the functions $g$, $a$ and $b$, are position-dependent, and the unknown scalar function $\gamma$ depends on both position and time. This program was applied to a non-structural problem in acoustic propagation, solving the wave equation $\nabla^2 \gamma = \gamma / c^2$.

The model chosen is a spherical liquid inclusion in a spherical solid with a centered point source. This particular model illuminates certain problem areas of great interest. The first is the boundary conditions associated with the solid liquid interface. This problem was resolved by judicious choice of isotropic finite elements, carefully chosen shear and Young's moduli, and a refinement of the mesh size.

The second problem is that of propagation in the low frequency range. The time step integration, wave length and mesh size are all interdependent for a properly converging solution to occur.

An analytical solution is obtained to illustrate both the accuracy of the solution as well as exhibit the implementation of the procedure. This analysis is used to obtain displacement in our model due to an impulsive compressional pulse from a point source situated at its center.

The analytical solution is compared to the finite element solution for a centered source and a finite element solution is obtained for an off centered source.


ANSYS Theoretical Manual Swanson Analysis.
FINITE ELEMENT ANALYSIS OF ACOUSTIC DEVICES

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The response of acoustic devices has long been studied with the help of lumped equivalent circuits but this fruitful approach suffers from several limitations, particularly when the driving frequency is of the same order than resonance frequencies of the constitutive parts. To avoid such problems, the finite element method (1) is a very powerful tool.

We have developed a three dimensional finite element analysis for the determination of resonance frequencies and mode shapes, using an isoparametric hexahedron (2), (3) with twenty nodes which yields accurate predictions even for loose meshes. A first application has been made to an axisymmetric structure such as to test the accuracy with respect to experimental as well as to other theoretical results. Then, a cross shaped directional coupling device has been studied, for symmetrical and unsymmetrical geometries.

Comparison with experimental results has been performed by using tweeter correctly orientated around the geometry to excite the good mode and a crystal pick up lied on the surface to detect the resonance.

These experimental results have shown that the fifteen first resonance frequencies are predicted within a few per cent or error, a graphical representation of the mode shapes allowing their definite experimental identification. The method is now able to help the design of new devices.

REFERENCES

(2) idem, p. 178-210.
THE PROPAGATION OF SOUND IN A MOVING NON-UNIFORM MEDIUM

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It is assumed that the fluid carrying the sound signal between the source and the reception, is moving and non-uniform. The equation describing the propagation of the density perturbation wave is:

\[
\frac{d^2 \rho'}{dt^2} + 2 \frac{d \rho'}{dt} \text{ div } \vec{V}_0 + \rho_o \text{ div} \left( \frac{c^2}{\rho_o} \text{ grad } \rho' \right) + \rho' \left[ \frac{d}{dt} \left( \text{ div } \vec{V}_0 \right) + (\text{ div } \vec{V}_0)^2 + \rho_o \text{ div} \left( \frac{\text{grad } \rho_o}{2} \right) \right] = S
\]

where \( \vec{V}_0 \) is the unperturbed fluid velocity (pressure \( \rho_o \), density \( \rho_o \)), \( \frac{d}{dt} \) is the material derivative

\[
\frac{d}{dt} = \frac{\partial}{\partial t} + \vec{V} \cdot \nabla = \frac{\partial}{\partial t} + \sum_{i=1}^{3} u_{0i} \frac{\partial}{\partial x_i}
\]

and \( S \) an self-generated noise, which is assumed small, as is \( \rho' \) compared to \( \rho_o \). The sound wave velocity is , while the terms featuring \( \text{ div } \vec{V}_0 \) correspond to damping and dispersion, (although the effect of viscosity is neglected here). Different types of behaviours are considered, due to the motion and the heterogeneity of the medium, for several cases of non-uniform flow.

An important basic steady flow assumes \( \text{ div } \vec{V}_0 = \alpha \) a constant.

A separation of variables then yields

\[
\rho' = e^{-\alpha t} \text{ Re} \int_0^\infty e^{i \omega t} F_\omega(x_1,x_2,x_3) d\omega
\]

where \( F_\omega \) is ruled by the following equation:

\[
\text{div} \left( \frac{c^2}{\rho_o} \text{ grad } F_\omega \right) + \left[ \frac{\omega^2}{\rho_o} - \text{ div} \left( \frac{\text{grad } \rho_o}{2} \right) \right] F_\omega = 0
\]

\( \text{grad } \rho_o, \rho_o \) and \( c^2 \) being given functions of the space variables \( x_i \).
Simplified Solution of Burgers' Equation for Numerical Calculation

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INTRODUCTION

The aim of the paper is derivation of such form of a Burgers' equation solution which make it possible to compute numerically the input, especially the input spectrum in the case of the Gaussian, zero-mean stochastic input.

THEORETICAL BASIS

The starting point is simplified solution of Burgers' equation given in the form [1]:

$$u(t,x) = \psi + \sum_{n=1}^{\infty} (-1)^n \frac{x^n}{n!} \frac{d^{n-1}}{dt^{n-1}} \left[ F^n(\psi) \frac{d}{dt} \psi \right]$$

$$F(\psi) = \frac{1}{c_s + \beta \psi} \approx \frac{1}{c_s} - \frac{\beta}{c_s^2} \psi$$

where $\psi = \psi(x)$ - the input, $u(t,x)$ - the output, $c_s$ - the small signal velocity, $\beta$ - the nonlinear parameter $\beta = 0.5(\lambda + \gamma)$, $\gamma$ - the ratio of specific heats.

Using the "harmonic input" method [2] the Volterra transfer function of Burgers' equation solution was determined.

RESULTS AND CONCLUSION

The r-th order Volterra transfer function for mentioned above solution (1) can be written in the form:

$$H_r(\omega_1, ..., \omega_r, x) = \sum_{n=1}^{\infty} (-1)^n \frac{1}{n!} \left( -\frac{\beta}{c_s \omega_1, ..., \omega_r} \right)^n \left[ j(\omega_1, ..., \omega_r, x) \right]$$

Thus the relation between the input and the output can be written as:

$$U(\omega, x) = \sum_{\nu=1}^{\infty} \left[ H_r(\omega_1, ..., \omega_r, x) \phi(\omega_1) \ldots \phi(\omega_r) \delta(\omega - \omega_1, ..., \omega_r) \right] d\omega$$

$$U(\omega, x) = \mathcal{F} \mathcal{T} \left[ u(t,x) \right] ; \quad \phi(\omega) = \mathcal{F} \mathcal{T} \left[ \psi(t) \right]$$

The results obtained are convenient for the numerical calculations. The calculations are not time-consuming if the limited accuracy is sufficient.

LITERATURE

A NEW METHOD FOR THE GENERATION OF THE BIFURCATION EQUATION

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An extensive treatment has been given to the mathematical problem that describes linear systems with layered inhomogeneities in their physical properties (1-8). Such a description encompasses many outstanding problems in a variety of applications to propagation and vibration. Extension to non-linear systems is now made.

Though the applications are diverse, a unified, systematic and recursive approach has been developed for linear systems (1-8). The analytical approach is compatible with machine operations for easy implementation on a real-time computing system. The integral equation formulation is re-structured so as to allow an iterative solution of the problem. The iterations are guaranteed to converge to a unique solution. The auxiliary constants have a physical meaning and the poles correspond to the natural frequencies of the system.

At times, more realistic modelling procedures are called for. Usually, this means using non-linear models rather than linear ones. This paper extends the above linear approach to solving or gaining insight into solutions of non-linear problems in sound and vibration. The treated non-linear problems are describable by a Hammerstein integral equation. Regular successive approximation methods can lead to a divergent series solution. Here, a new functional series representation of the solution is produced to a re-structured form of the Hammerstein equation. The bifurcation equation is generated and questions of uniqueness, of type and number of solutions are addressed. One application is made to propagation in a non-linear and inhomogeneous medium. Another application is made to forced oscillations of finite amplitude in a periodic system.

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DISCONTINUITY IN THE STOCHASTIC PLANE WAVE CASE

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INTRODUCTION

A statistical description of the shock formation region in the intensive stochastic plane wave case is given. It is assumed that the wave is propagated in a lossless medium, and the excitation signal \( \psi(t) = u(0,t) \) is described by the stationary Gaussian process with the zero mean and with the autocorrelation \( B(\tau) \). Statistical aspects of the intensive stochastic acoustic wave were studied in [1,2]. Nevertheless, the said problem has not been presented in literature yet.

ANALYSIS

Starting from the formula for the intersection point \((\bar{x}, \bar{\varepsilon})\), (cf. [3]), of two characteristics corresponding to any \( t_1, t_2 \) of the time axis, for any fixed sample function of the \( \psi(t) \) process, the probability density for \( \bar{x} \) has been derived. The probability density is given by:

\[
P_{\bar{x}}(x) = 1(x) \left[ \frac{c_0^2 f(\tau)}{2\pi \sigma \beta x^2} \exp \left( - \frac{c_0^4 f^2(\tau)}{4\sigma^2 \beta^2 x^2} \right) + \frac{c_0^2}{2\beta x^2} \delta \left( \frac{c_0^2}{x^2} \right) \right],
\]

in which \( f(\tau) = \tau / \sqrt{1 - R(\tau)} \), \( \sigma^2 = B(0) \), \( R(\tau) = \sigma^2 B(\tau) \), \( \beta = (\gamma + 1) / 2 \), and \( \tau = t_2 - t_1 \) is a parameter of \( P_{\bar{x}} \).

CONCLUSIONS

1. The expected value of \( \bar{x} \) equals infinity, then it is useless statistical parameter of the random variable \( \bar{x} \).
2. The distance \( x \) such that the probability of the event that \( x < x \) is 0.1, is given by: \( x = 0.43 c_0^2 (\beta \sigma)^{-1} f(\tau) \) and it reaches its minimum at \( \tau = 0 \); \( \min x = 0.61 c_0^2 (\beta \sigma)^{-1} (R(0))^{-0.5} \).

If \( R(\tau) = \sin \omega \tau / (\omega \tau) \), then \( \min x = 3 \cdot 10^2 (k \omega)^{-1} [m] \), \( k = \sigma / c_0 \) (for the sine wave, i.e. when \( u(0,t) = k_0 \sin \omega t \), then the shock formation distance is given by: \( x = 2.83 \cdot 10^{-2} (k \omega)^{-1} [m] \).

LITERATURE

J. AEROACoustics, Atmospheric Sound
Prise en compte de l'effet de sol dans la prévision des niveaux de bruit.

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KIA OR6

La majorité des auteurs s'accordent sur la nécessité d'envisager trois modèles de sols: a) surface à réaction localisée - b) milieu poreux d'épaisseur infinie - c) couche de milieu poreux d'épaisseur finie. Les deux derniers modèles correspondent à une surface à réaction étendue, et se ramènent asymptotiquement au premier.

Dans le présent travail, on étudie la réflexion d'une onde sphérique issue d'une source \( S (x = 0, y = 0, z = s) \) par un plan \( (z = o) \) correspondant successivement aux trois modèles proposés. On montre que, dans tous les cas, la pression acoustique s'exprime comme la somme du champ incident, du champ réfléchi provenant de la source image \( S'(z = o) \) et d'intégrales représentant des potentiels de couche dont les densités dépendent des paramètres caractéristiques du modèle de sol envisagé.

On montre ensuite que la pression en un point \( M \) peut être représentée par une série convergente dont les premiers termes fournissent une approximation "grande distance":

\[
p(M) \approx \frac{e^{ikr(S,M)}}{4 \pi r(S,M)} - a(\theta) \frac{e^{ikR(S',M)}}{4 \pi R(S',M)} - D(a(\theta)) \frac{e^{ikR(S',M)}}{4 \pi R(S',M)}
\]

+ termes à décroissance exponentielle

où \( a(\theta) \) est le coefficient de réflexion en ondes planes sous incidence \( \theta \); \( D \) est un opérateur de dérivation, d'ordre 2 en \( \theta \). Une étude numérique montre que les termes à décroissance exponentielle ont une contribution appréciable lorsque \( R \) n'est pas très grand. Une comparaison entre expérience et calcul numérique met en évidence la précision des prévisions de niveau sonore que l'on peut obtenir à l'aide de cette expression approchée. Enfin, on décrit une méthode expérimentale permettant de mesurer, in situ, les caractéristiques acoustiques d'un sol correspondant à un modèle donné et de préciser l'erreur moyenne à laquelle on doit s'attendre dans un calcul de prévisions.
ACOUSTO-OPTIC SYNAPSIS IN INTERSTELLAR COUPLING
ANGLO AUSTRALIAN OBSERVATORY

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In situ measurements of plasma by recent Voyager and Mariner space flights have shown that space is strongly inhomogeneous. These observations may be correlated with the recently discovered anomalies in the measurement of stellar/stellar coupling via the synaptic acoust-optic coupling constant $\Gamma$. The anomalies have previously been attributed to acousto-optic effects of the interstellar medium. An alternative theory is presented invoking recently refined values of the Holz parameter than can accommodate these anomalies and at the same time predict absolute values for the acoustic-optic constant that are several orders of magnitude different from measured values. The theory makes use of recent results\(^1\) on the scattering cross-section effect on acousto-optic transformation in plasma in the Crab nebula.

   Physical Processes in the Crab Nebula.
RISETIMES OF SONIC BOOM SCHOCK WAVES

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The present study was initiated on the basis that shock-wave risetimes in sonic booms have been reported to be 100 to 1000-fold longer than the risetimes predicted by the weak-shock theory. The discrepancy has for a number of years challenged researchers to investigate its causes. The explanation is to be found in the following effects.

I. Real Gas Effects

II. Wave interaction, such as
   a) Wave - Turbulence
   b) Wave - Wave
   c) Wave - Ground surface
   d) Wave - Thermal Gradient

III. Monitoring
   a) Transducer and associated electronics
   b) Method for determining the risetime

Our theoretical and experimental analysis show that real-gas effects can not cause a thickening of the weak shock more than about a factor 2 for peak pressures greater than 100 Pa.

The wave interaction effects are demonstrated by experiments at UTIAS and ISL-FRANCE. We found these effects strong enough to explain the discrepancy between measured and predicted risetimes.

From collected sonic-boom data from NASA, France and RAE we conclude that in some cases the risetime of the steep part of the sonic boom has been limited by the response time of the measuring system.

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AN INVESTIGATION OF THE FRESNEL-REGION FIELD OF ECHOSONDE RADIATORS

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This paper investigates the Fresnel-zone Field which is near the Bore-sight of an acoustic echo-sounding (Echosonde) antenna that has been successfully used to measure the temperature fluctuations and turbulent velocity fields of the lower atmosphere. The paper starts with a brief discussion on the establishment of the Fresnel-zone range condition given by \( R \geq 0.6204 \times \left( \frac{D}{\lambda_a} \right)^{2} \), for an acoustic wave (where \( D_u \) is the upper transmitting aperture of the three-dimensional analytic model of the acoustic antenna, and where \( \lambda_a \) is the acoustic wavelength).

When the range condition is thus determined, it is invoked on the exponential quantity described by \( \exp(-jk\cdot r')/r' \), that appears in the integrand of the general integral expression prescribing the field radiated by the antenna (where \( r' \) is the distance from the source point to the field point, and where \( k \) is the acoustic wavenumber). Consequently, one obtains the Fresnel-zone field in the boresight neighbourhood of the antenna. We give an analytical example of practical interest in which the Fresnel-zone field is evaluated in closed form for an idealized parabolic source intensity of common occurrence. The closed-form expression thus obtained, is given in terms of highly convergent Lommel \( V_p \) functions, which are expedient for practical computations. A special case deduced from the results is the Fresnel-zone Field produced by an idealized uniform source distribution. The results given are general because any source function \( A_0(\zeta) \) can naturally be represented by an even polynomial of the argument. Such a polynomial representation can then be converted to a distribution of the form \( A_0(\zeta) = (1 - \zeta^2)^m \) from which we deduce the parabolic source excitation considered here when the parameter \( m \) assumes the value of unity. We finally interpret the parabolically tapered source distribution physically, in terms of the design parameters of the acoustic-radar antenna.
TOWARD A MORE RIGOROUS SYNTHESIS OF ECHOSONDE PATTERNS

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In a recent paper (S. A. Adekola and J. W. Wescott: ACUSTICA, Vol. 42, No.4, pp. 249-259, 1979) a pair of Hankel Transform integral equations was deduced for pattern synthesis of an acoustic echo sounding antenna invoking the Dirichlet condition. The present paper establishes that, without invoking the Dirichlet condition, this pair of integral equations can be used for a more rigorous echosonde synthesis of specified echosonde patterns leading, in some cases of the assumed patterns, to closed form solutions of the source distributions. It is observed that the synthesis of exact beam patterns produces source distributions that are described by integrals which are carried out over infinite domains. Because source functions are finite in real life, a physical requirement to be satisfied is that the integration which gives the source distribution, be carried out over a finite region.

In practice, the source distribution is confined only to the finite echosonde aperture (i.e. $0 \leq \rho_s < \rho_s(\text{max})$, where $\rho_s(\text{max})$ and $\rho_s$ are the aperture radius and the variation across the echosonde transmitting antenna aperture respectively) although in theoretical analysis the specified farfield pattern is recoverable by integrating the source distribution to infinity provided the radius is much larger than the wavelength. If however the integration is now truncated on a finite $\rho_s(\text{max})$ we obtain an approximate pattern. Analytical examples of some rigorous pattern synthesis of practical utility presented for clarity, show how the approximate patterns approach the exact patterns when the antenna size becomes large compared with the acoustic wavelength, or when $\rho_s(\text{max})$ becomes infinite. These include: exponential, Gaussian and complete sine-cosine patterns, while the approximate pattern evaluations provided, are given in closed forms.
A previous study of the author (S. A. Adekola, J. Acoust. Soc. Am. Vol. 64, Suppl. No. 1, pp. S124–S125, 1978) developed a compact theory for investigating Echosonde arrays which have become increasingly of great importance in acoustic remote sensing, particularly when a single element may not be able to provide adequate directivity and control. In the present paper a workable closed-form expression for the geometrical directivity $D$, of a Broadside Echosonde-array is derived. The expression reveals that, while the directivity at $d = \lambda_a / 2$, (where $d$ and $\lambda_a$ are the arbitrary inter-element spacing and acoustic wavelength respectively), is less than that for $d = 3\lambda_a / 4$, (i.e. $D_{d=\lambda_a/2} < D_{d=3\lambda_a/4}$), the directivity at $d = \lambda_a$ is identically equal to that at $d = \lambda_a / 2$. In addition, it is found that the same directivity is numerically equal to $n$, the total number of elements in the array (i.e. $D_{d=\lambda_a/2} = D_{d=\lambda_a} = n$). It is also confirmed in this paper that, on the reduction of the array to a single element, that is when $d$ tends to zero, the directivity is numerically equal to unity.

The practical utility of these results is that while it is desirable to make inter-element spacing greater than a half-wavelength, the spacing should always be less than a wavelength, if grating lobes are to be removed. Furthermore, another interesting feature of the present investigation is the development of a criterion for determining the optimum spacing for a Broadside array of Echosondes. Based on the results of many computations, the optimum spacings for a four-, eight-, and twenty-element array have been determined to be about 0.8\lambda_a, 0.91\lambda_a, and 0.95\lambda_a respectively as examples, while for a larger number of elements, the optimum spacing for grating lobe attenuation is about 0.97\lambda_a. Directivity increases with inter-element spacing until an optimum is reached. Thereafter, directivity reduces with further increase of inter-element spacing. This is because when the optimum spacing is exceeded, grating lobes are introduced into the array factor which reduce the directivity.

When a design is required such that the beam-maximum is changed, as in scanning for example, the optimum spacings should be less than the numerical values given above, and should be close to a half-wavelength. Some numerical examples of computed beam-patterns of Broadside Echosonde-arrays are also presented.
It is shown in this paper that it is possible to use Orthogonal Zernike circle-polynomials to study the characteristics of an acoustic-antenna that is conventionally employed for remotely sensing the lower atmosphere. The radiation integral that describes the field radiated by the shielded antenna is transformed to an integration over a unit circle, so that the Zernike circle-polynomials can be directly applied to express the field in a closed form, while any arbitrary aperture-excitation of the antenna can also be expressed by similar circle-polynomials. It is noted that this representation is of considerable practical significance, in that a utilization of the error-minimizing property of these orthogonal polynomials enables one to achieve accuracy as well as economy of time in the computerisation of the radiated field. Indeed, the analysis enables us to study the Zernike Polynomials to much higher degrees of accuracy than those available in the existing literature. Graphical illustrations are also provided for the sake of further clarity.

Examples presented include: simulations of patterns diffracted by two practical antennas that have been successfully used for acoustic remote sensing, both in the marine environment and over dry land. Modified Zernike aperture distributions used for these two antennas provide relative ground-level sidelobe attenuations within -66.13dB and -26.62dB for 1-1.5kHz frequency band, while sidelobe reductions between -54.39dB and -57.23dB are obtained for 2-3kHz acoustic frequency band.
ACOUSTIC RESONANCE IN FLOW DUCTS CONTAINING PLATES: WHERE IS THE SOUND SOURCE?

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INTRODUCTION

Acoustic resonances excited by periodic flows around a plate in a duct were investigated by Parker [1], Cumpstpy and Whitehead [2] and others. When the Parker β-mode was excited Cumpstpy and Whitehead concluded that excitation of the resonance came mainly from pressure fluctuations in regions just upstream of the plate's trailing edge. These pressure fluctuations were induced by vortex shedding from the trailing edge.

Kristiansen [3] showed that for a plate with a square leading edge and thickness equal to half the duct width, Parker's β-mode resonance was excited at frequencies equal to the frequency of the vortex shedding from the leading edge. Consequently, Kristiansen believed the sound producing region to be located near the leading edge of the plate.

Welsh and Gibson [4] found that for a plate with a square leading edge, a chord to thickness ratio of five and a thickness equal to 2.5% of the duct width, a Parker β-mode resonance was initially excited at the frequency of vortex shedding from the plate's trailing edge. As the sound level increased during establishment of the resonance, the vortex shedding frequency suddenly doubled and vortices were shed from the leading edge [5].

RESULTS AND DISCUSSION

For the plate and duct system investigated in [4], it is hypothesised that the initial sound producing flow region was associated with pressure fluctuations due to vortex shedding near the trailing edge, in a similar manner to that described in [2]. When the vortex shedding frequency increased suddenly and vortices were shed from the leading edge, applying the theory of [2] implies that the sound producing region shifts to just downstream of the plate's leading edge. A similar change to the location of the sound producing flow region is hypothesised for the plate shown in Fig. 1.

It has been shown theoretically by Crighton [6] and more recently by Howe [7], that a vortex filament passing around the edge of a semi-infinite half plane is a source of noise. Clearly, this possibility exists for the plate shown in Fig. 1.

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INTRODUCTION: The generation and propagation of sound from a compact transient acoustic source in a duct has been examined in detail. The disturbance pressure field, both upstream and downstream of a mass, force or energy source, has been examined for subsonic, sonic and supersonic plug flows. Interaction of each type of source with the mean flow is different and has different consequences on the upstream and downstream propagation of the ensuing sound pressure disturbance.

ANALYSIS: Starting with the inviscid equations of motion and energy for a perfect, non-conducting gas, it may be shown that, for a plug flow of velocity $U$ in the $x_1$ direction, the pressure $p'$ generated by a compact acoustic source is given by the convected wave equation

\[
\frac{\partial^2}{\partial t^2} - c_o^2 \frac{\partial^2}{\partial x_1^2} \frac{p'}{(\gamma - 1)} \frac{\partial}{\partial t} \left( (E' + \frac{1}{2} m' U^2 - UF') \delta(x_{1j} x_j) \right)
- c_o^2 \frac{\partial}{\partial x_1} \left( \delta (x_{1j} x_j) \right)
\]

where $m'$, $F'$ and $E'$ are the mass flux, force and energy flux components of the source at $x_{1j}$, and $D/Dt$ is the mean flow component of the total differential. The source terms on the right hand side of (1) differ from others in the literature, e.g., [1,2], because, in the derivation, the perturbation energy-equation was invoked to avoid any assumption that the disturbed flow is either isentropic or irrotational. Equation (1) has been solved for the case of a transient source in a rectangular duct of infinite extent.

RESULTS AND DISCUSSION: The key results of this work are embodied in the right hand side of (1). Through interaction with the mean flow, both a mass source and a force source contribute to the perturbation energy fluctuations, and all three sources affect even the plane wave mode of sound propagation. Thus the ratio of amplitudes of upstream and downstream travelling sound pressure signals from a transient sinusoidal source depend very much on the character of that source, even at very low frequencies. Figure 1 shows the ratio of amplitudes for plane wave sound for each type of source investigated in a subsonic flow, together with the experimental results of [3] for a loudspeaker source.

REFERENCES
Determination, by correlation techniques, of turbulence, plane wave and higher mode contributions to the wall pressure field in disturbed pipe flow.

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Severe flow disturbances, such as that due to a mitred bend, in piping systems containing fully-developed turbulent flow result in intense internal wall pressure fluctuations. Well downstream of any separated flow region in the vicinity of the disturbance, undisturbed pipe flow is re-established, but the wall pressure field there consists not only of the turbulence but also of a superimposed propagating acoustic field, which comprises plane waves and higher order modes. For the prediction of pipe wall response to this pressure field it is essential to know the relative contributions of plane waves, and higher order modes to the pressure, in frequency bands. This resolution can be achieved by means of longitudinal space-time correlations in filter bands of appropriate width. The technique previously used by several researchers [1,2] has been extended [3] to allow the plane wave component of the pressure field in a band containing higher order modes to be obtained. The contribution of higher order modes in any filter band can then be obtained by subtracting the known turbulence component and the measured plane wave component from the total mean square pressure in the band.

Correlation measurements have been made of the disturbance due to a 90° mitred bend, for several flow Mach numbers (0.2 < M < 0.5) and four transducer spatial separations in octave and 1/3-octave bands. These show that the bend is a strong generator of plane waves at frequencies below the cut-off frequency of the first higher order mode, and that, as frequency increases, the proportion of this component in the wall pressure decreases (Fig.1). Pressures due to the first higher order mode, the (1,0) mode, can be estimated in filter bands containing no other higher order modes. Bands at higher frequencies will contain more than one higher order mode and with the present technique, these cannot be distinguished from each other because of their dispersive nature. In these cases, only the pressures associated with all higher order modes in a filter band can be obtained. For the particular case of the 90° mitred bend, higher order modes dominate the wall pressure field at all frequencies above the cut-off frequency of the (1,0) mode.

REFERENCES


Fig.1 Relative contributions of plane waves, higher order modes, and turbulence to the wall pressure field downstream of a 90° mitred bend in a region where mean velocity profiles of undisturbed flow have been re-established (M ≈ 0.4).
VELOCITY DEPENDENCE OF THE SPECTRA OF HYDRODYNAMIC AND ACOUSTIC WALL PRESSURE FLUCTUATIONS IN TURBULENT PIPE FLOW DOWNSTREAM OF DISTURBANCES.

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Many investigations have shown that the mean square value $\bar{p}^2$ and power spectral density $\Phi$ of wall pressure fluctuations in undisturbed turbulent pipe flow depend broadly on $U^4$ and $U^3$ respectively (where $U$ is the mean centre-line velocity). When the flow is disturbed by pipe fittings this is generally no longer the case, even well downstream of the disturbance where non-propagating components of the disturbance have died out, and the wall pressure field comprises a propagating acoustic field superimposed on the pressure field characteristic of undisturbed turbulent pipe flow. Flow disturbances due to 45° and 90° mitred bends, 90° radiused bends with ratios of bend radius to pipe radius of 3.0 and 6.4, a fully-open gate valve and a fully-open butterfly valve have been investigated at flow Mach numbers $M$ in the range 0.2 < $M$ < 0.5. Mean 1/3-octave band values of non-dimensional power spectral density $\Phi = \rho_f U^2/2$, $\rho_f$ fluid density, 'a' internal pipe radius) are shown as a function of Strouhal number $\Omega = \omega a/U_0$ in Fig.1. Data for undisturbed turbulent pipe flow are also shown.

At a given Strouhal number, the increment in $\Phi$ due to the internal acoustic field scales as $U^3$ at frequencies below the cut-off frequency of the first higher order acoustic mode and as $U^5$ at frequencies above it. This is illustrated for a 90° mitred bend in Fig.2 (where $\Phi_0$ is the spectrum for undisturbed flow).

The scaling of the resultant $\Phi$ is therefore as $U^3$ at frequencies below the first higher order mode cut-off and with a power of $U$ between 3 and 5 (depending on the severity of the disturbance) at frequencies above it. The 45° mitred bend, both radiused bends, and the gate valve generate only weak internal sound fields (Fig.1). In these cases $\Phi$ at all Strouhal numbers and $p^2$ scale essentially as $U^3$ and $U^4$ respectively. The 90° mitred bend and butterfly valve generate intense internal acoustic fields: scaling of overall $\Phi$ is as $U^3$ at low $\Omega$ but as a higher power of $U$ at high $\Omega$. Even so, the scaling of the overall $\bar{p}^2$ does not depart markedly from $U^4$. 
Large Scale Model Measurements of Air Frame Noise Using Cross-Correlation Techniques in the 40-80-Foot Wind Tunnel

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Cross correlation techniques are used to measure the sound radiated by wing/flap airfoil configurations in the NASA-Ames 40- by 80- Foot Wind Tunnel using a 5.8 m high model, with three deployed flaps. The sound from flap corners exceeds other airframe noise by 10dB and more; the noise from the leading, outboard corner of the leading flap seems to be strongest. The sound is estimated using two formulas based on standard aeroacoustic theory and one method using the near-far-field cross-correlation size; this last is essentially independent of such theory—all three are in fair to good agreement with one another. The classic dipole angular distribution pattern for one dipole is compared with measurements; it is found that there is qualitative but not quantitative agreement. The dependence of intensity on \( u_0 \) is roughly though not exactly, \( u_0^6 \), where \( u_0 \) is the free stream speed. The turbulence length scales on the flap surface, as determined by the characteristic time of the measured correlation function and the free stream speed, are from a few to many cm, of the order of the flap thicknesses. Time delays from the correlation between the far field signal and the surface source are determined from the correlation functions and are in good agreement with the flow-refraction-corrected results.
Aerodynamic Sound Generated by a Slotted Trailing Edge
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This paper discusses the theory of the generation of sound by turbulent flow over a trailing edge flap of an airfoil or guide vane. A narrow slot separates the flap from the airfoil, and the configuration is modelled analytically by means of a semi-infinite rigid plate which contains a slot at an arbitrary, finite distance from the edge. The aerodynamic sound problem is formulated in terms of a singular integral equation which can be solved in closed form when the width of the slot is small compared with both the acoustic wavelength and the chord of the flap. The theory is applied to examine (i) the influence of the slot on the noise generated as a result of boundary layer separation close to the trailing edge of the flap, and (ii) the generation of sound by boundary layer turbulence in the immediate vicinity of the slot. At low, subsonic mean flow Mach numbers the presence of the slot is shown to reduce the level of the radiated noise in case (i) provided that $kd \leq 10$, where $k$ is the characteristic acoustic wavenumber and $d$ is the chord of the flap. Boundary layer noise is predicted to be significantly enhanced by the slot in the range $0.1 \leq kd \leq 10$. 
Porous diffusers have been developed in the past few years, and are noted for small volume, simple structure and high noise reduction. In this work, the aerodynamics of flow through porous materials is studied in detail. Let a block of porous material be set at one end of a tube and air exhausts through it. If there is no heat exchange between air flow and its surroundings, equation of conservation of energy is satisfied. Equation of momentum for air passing through porous material becomes

\[ \frac{dP}{L} + \rho u \frac{dP}{u} + \frac{dP}{L} = 0 \]

Flux of flow is constant, \( J = \gamma \frac{P}{M^2/u} = \text{const.} \). Based on above equations, the following relations are derived:

\[ J = \frac{\gamma R}{f(M_0) - f(M_1)} \quad (1) \quad P_{s1} = J \frac{C_s}{M_1} \left( 1 + \frac{\gamma - 1}{2} \frac{M_1^2}{M_1} \right)^{\frac{\gamma + 1}{2(\gamma - 1)}} \quad (2) \]

\[ f(M) = \int_{M_0}^{1} (1 - M^2) dM/M^2 \left( 1 + \frac{\gamma - 1}{2} M_1^2 \right)^{\frac{1}{2}} \quad (3) \]

where \( C_s \) the stagnation sound velocity, \( \gamma \) the ratio of specific heats, \( R \) the flow resistance of the block, and \( P \) the stagnation pressure. Subscripts 1 or 2 denote the quantity measured upstream and downstream, resp., of the porous block. In the case of sonic efflux, \( M_2 = 1 \). In the case of subsonic efflux, \( J \) can be determined by \( M_2 \):

\[ J = \gamma P_0 M_1 \left[ 1 + \left( \gamma - 1 \right) \frac{M_1^2}{2} \right]^{\frac{1}{2}} / C_s \quad (4) \]

For given values of \( R, P_0 \) and \( M_2 \), we can calculate \( M_1 \) by solving equation (1) and (4). Substituting \( M_1 \) and \( M_2 \) in equations (1) and (2), relation between \( J \) and \( P_{s1} \) then is found. In the region of sonic flow, the curve of \( J - P_{s1} \) approaches a straight line as \( P_0 \) is large. Extending of the straight line bisects \( P_0 \) axis at positive \( P_{s1} \), unless \( R = 0 \). Equation of the straight line can be written as

\[ J = \alpha (P_{s1} - \Delta P_{s1}) \quad (5) \]

Efflux from a sonic nozzle is

\[ J = \beta P_{s1} \left( \frac{J + 1}{2(\gamma - 1)} \right)^{\frac{\gamma + 1}{2(\gamma - 1)}} \quad (6) \]

where

\[ \beta = \gamma \left( \frac{C_s}{T_0 \gamma} \right) \left( \frac{J + 1}{2(\gamma - 1)} \right) \]

Comparing equations (5) and (6), we suggest that the sonic efflux from a block of porous material can be simply characterized by an effective stagnation pressure drop \( \Delta P_{s1} \) and an effective area \( A = \alpha / \beta \), \( \Delta P_{s1} \) and \( A \) are determined by experiments.
NOISE GENERATION OF A SUPersonic COAXIAL JET WITH INVERSE TEMPERATURE PROFILE

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EXPERIMENTS

The spatial distribution of apparent sound source intensity within a flow consisting of a cold supersonic jet surrounded by a hot annular jet was determined by means of an acoustic mirror telescope (1). The jets emanated from a cutting torch nozzle, the annular jet was heated by a propane or acetylene flame (2,3).

RESULTS

The maximum sound radiation of the supersonic core jet alone occurs about 15 nozzle diameters downstream of the nozzle, in good agreement with results obtained earlier for jets of considerably greater diameter. The subsonic hot jet (heating flame) generates sound only in the immediate vicinity of the nozzle. If both jets are combined, two regions of strong sound generation are observed, see Fig. 1. The first one close to the nozzle can be identified as the combustion noise of the heating jet. The second one is located very far downstream of the nozzle. This can be explained by investigations of the flow field (2,3) showing that the heating jet prevents the core jet from disintegrating over a long distance by suppression of the mixing process with the ambient atmosphere.

Fig. 1: Relative distributions of sound source intensity in a coaxial jet for the 1/3 octave band with Strouhal number 0.06

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The noise radiated from inverted velocity profile cold jets has been investigated experimentally over a wide range of simulated flight conditions. Such jets consist of a low-speed gas flow in the center and a higher speed annular flow. Because of the promising potential for the use of variable-stream-control engines in advanced supersonic transports, which employ inverted velocity profile jets, experiments have been performed which simulated such jets in flight. The diameters of the center and the annular nozzles were 1.27 cm and 2.03 cm respectively. The coaxial nozzles were located in an anechoic chamber. They were surrounded by another annular nozzle of 17.8 cm diameter. Flight simulation velocities as high as 150 m/s could be obtained in the outer annular flow. Analysis of the far-field noise data showed that as the flight simulation velocity was increased for a given inverted-velocity profile jet, relatively little noise reduction was observed in the forward quadrant as compared to noise measured close to the jet axis. Typical results of the far-field noise data at $\theta_c=60^\circ$, and $\theta_f=140^\circ$ are shown in Figure 1 where $\theta_c$ is the angle with respect to the intake. The radiated noise was normalized to an arc of radius 1000D. Close to the jet axis, the spectrum of the radiated noise indicates the noise reduction to be of a relatively lower frequency component of the pressure signal. A surprising observation was that at various flight simulation velocities, the absolute noise reduction was virtually constant both with and without the presence of a given center flow velocity in the angular range $60^\circ<\theta<150^\circ$. This noise reduction with center flow was observed over the entire frequency range at $\theta=60^\circ$; whereas a significant reduction of lower frequency components was observed at $\theta=140^\circ$. It is inferred from these results that the modification in the jet structure which occurs with the presence of the center flow, is not altered by the flight simulation flow. This paper represents one phase of research performed by the Jet Propulsion Laboratory, California Institute of Technology sponsored by the National Aeronautics & Space Administration Contract NAS7-100.
The use of air jets in industry is often associated with cleaning, cooling or movement of major produced items. Thus a substantial component of the noise produced is due to the acoustic dipole effect originating in the unsteady forces produced by the jet. This paper considers the relationships which are appropriate to the prediction of the dipole noise source component under these conditions. The analysis considers the use of available data for aerodynamic turbulence in jets for the establishment of the variation of far field noise with nozzle velocity, spacing between nozzle and the solid object causing the dipole effect, and other geometric parameters. The particular case of small diameter cylindrical objects is considered, and relationships for the sound produced are derived.

Results of a series of experimental noise measurements are compared with the analytical predictions based on jet turbulence characteristics. It is found that the rate of change associated with the dipole source mechanism is better represented by the body size than by the turbulent microscale, as the latter is much the smaller quantity for the experimental conditions investigated. The directivity of the radiated sound field strongly suggests that the force fluctuations transverse to the direction of the jet flow greatly exceed those in a streamwise sense. The sound is produced most strongly in the one third octave band centered at a Strouhal Number of 0.3 based on local flow velocity and cylinder diameter. When the cylinder is close to the nozzle an increase of sound intensity with the square of the spacing between nozzle and cylinder is predicted, whilst when the cylinder is located in excess of 20 diameters from the nozzle a decrease of intensity with the fourth power of spacing is predicted.
K. MUSIC AND MUSICAL INSTRUMENTS
HARMONIC GENERATION IN ORGAN FLUE PIPES

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The mechanisms involved in sound generation in organ pipes, and related instruments such as the flute and the recorder, have been the subject of scientific study for more than a century, but it is only during the past few years that any detailed understanding has been achieved. The present paper summarizes some recent work performed in our laboratory which clarifies some important details of the processes of excitation and harmonic generation.

Conditions for sustained oscillation require that the feedback loop, containing the pipe resonator and the exciting jet, should close, and this is accomplished when the jet length encompasses about half a wavelength of the transverse disturbance propagating along it. In an organ pipe of relatively wide scale the jetdeflection at the pipe lip is nearly sinusoidal at the fundamental frequency, but the form of the jet velocity profile leads to a high degree of nonlinearity in the resulting acoustic driving force. Consideration of the dependence of the form of this nonlinearity upon various pipe-voicing parameters leads to immediate semi-quantitative understanding of harmonic generation. Of particular interest is the dependence of the relative intensities of odd and even harmonics on the offset of the upper lip of the pipe relative to the symmetry plane of the jet. Measurements on an adjustable organ pipe confirm these theoretical predictions.
Musical Tones Generated in Pipes by Low Velocity Wind

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The Singing Ship memorial was erected at Emu Park near Rockhampton in Central Queensland in 1970 to commemorate the discovery of that part of Eastern Australia in 1770 by Captain James Cook. The memorial represents a ship in sail or an anchor depending on the direction from which it is viewed. Thirty six PVC pipes with various lengths and diameters are incorporated in the 13m high reinforced concrete structure and these 'sing' in the wind. Most of the pipes are arranged in the 'mast' of the 'ship'.

Edge tones are generated as wind blows across openings in the sides of the pipes and these cause the air columns in the pipes to oscillate. The size and position of each opening was determined experimentally using a small wind tunnel so that the pipes produce pleasing sounds in a gentle breeze and never produce objectionable noise in strong winds. To allow for the variation in wind direction, bundles of six similar pipes with opening slits directed outwards were constructed and fitting into the 'mast' in such a way as to allow the bundles of pipes to rotate slightly.

Since the memorial is located only 120m from the nearest residence, it has been necessary to restrict the rotational motion of the bundles of pipes to reduce the loudness of the sounds and avoid noise complaints.

The sound from the 'ship' varies with the strength and direction of the wind and is never excessively loud. The memorial has become a local attraction and is popular with tourists.
WAVE PROPAGATION ON TURBULENT AIR JETS

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The air jet in a normal organ flue pipe is fully and homogeneously turbulent, a situation aided by "nicking" of the languid. We have therefore extended our previous studies of wave propagation on laminar jets to the case of the fully turbulent jet, which does not appear to have been extensively investigated in this context previously.

Gratifyingly the propagation behaviour, which is rather complex for the case of a real laminar jet in a fluid with finite viscosity [Fletcher and Thwaites, Acustica 42, 323 (1979)] becomes rather simple and systematic for a fully turbulent jet. We report here the results of measurements of the real and imaginary parts of the wave propagation coefficient (or the phase velocity and the amplification coefficient) for a range of frequencies and jet parameters of interest in musical instruments such as flutes and organ pipes.

The general behaviour is very much like that for the high-frequency limit for ideal inviscid jets of similar velocity profile. The phase velocity is nearly constant at about half the jet centre-plane flow velocity, while the amplification coefficient goes through a maximum at a frequency related to the local width of the jet.

S’appuyant sur les idées mentionnées ci-dessus, on a préparé dernièrement en Pologne une méthode pratique pour étudier des orgues. Elle tend vers un inventaire acoustique de nombreux instruments, surtout anciens, se trouvant dans des églises polonaises, et qui jusqu’à présent n’ont été étudiés que sporadiquement [4]. Elle permet les comparaisons entre les résultats objectifs, acquis sous forme de distributions de la densité spectrale intégrée de longue durée, les résultats des mesures de caractéristiques acoustiques de l’intérieur, et les résultats subjectifs d’évaluation de la qualité d’un instrument.

On décrit cette méthode en détaillant la technique de mesure utilisée et en donnant les résultats des études exécutées sur quelques orgues choisis en Pologne du Nord. Comme illustrations on présente les extraits des enregistrements de ces orgues, fait au cours des études.

Cette méthode a pour but de rendre possible les comparaisons de la qualité entre divers instruments d’orgue, même étudiés par différents chercheurs, d’où l’intérêt d’une large discussion basée sur les résultats préliminaires, avant que la méthode ne soit définitivement introduite dans la pratique.

Les études décrites sont menées par le Laboratoire d’Électroacoustique de l’Université Technique de Gdańsk, avec l’appui du Comité d’Acoustique de l’Académie Polonaise des Sciences à Varsovie.

CALCUL DE L'IMPEDEANCE D'ENTREE DES INSTRUMENTS A VENT.

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INTRODUCTION

L'équation des pavillons dite de Webster, a des solutions connues dans certains cas (conique, exponentiel, ...). Mais ces solutions ne sont plus valables quand on introduit l'amortissement qui dépend du rayon, dû aux pertes visco-thermiques aux parois. Pourtant, le calcul des fréquences de résonance en ondes stationnaires (sans rayonnement ni pertes visco-thermiques) a été largement pratiqué depuis Bernoulli, car il reste très convénable. Nous avons tenté d'élargir le calcul classique aux calculs de grandeurs caractérisant les ondes quasi-stationnaires (essentiellement les extrênsus du module de l'impédance Z), en remarquant que la partie réelle de Z est presque partout très inférieure à sa partie imaginaire.

L'EQUATION DES PAVILLONS AVEC AMORTEMENT

Les amortissements visqueux et thermique aux parois peuvent s'écrire pour un tuyau cylindrique sous la forme respectivement d'une impédance en série \( Z_v \) et d'une admittance en série \( Y_t \). On en déduit l'équation de Webster pour l'admittance réduite \( \frac{Y'}{g} = \frac{Z_v}{Y^2 - Y_t} \) où \( g = 2r^2/r \), \( r \) étant le rayon. Pour \( Z_v \) et \( Y_t \), on est amené à prendre les expressions suivantes : \( Z_v = k (P_v + j) \) et \( Y_t = k (P_t + j) \), où \( P_v \) et \( P_t \) sont petits, et proportionnels à \( 1/r \) \( \sqrt{T} \).

FONDEMENTS DE LA METHODE APPROCHEE ET RESULTATS

On écrit l'équation sous la forme : \( a' + ga = 2 \kappa a - k P_v b^2 - k P_t \)
\( b' + gb = - k (b^2 + 1) \)

où \( Y = a + jb \) si \( b \) n'est pas trop grand, et \( Y = (a + jb) (1 + a^2/b^2)^{-1} \) si \( b \) est grand. La deuxième équation est l'équation des ondes stationnaires, la première est linéaire et s'intègre dès qu'on connait la solution stationnaire. Les extrems de \( Z \) sont alors donnés par : \( |Z|_{\text{max}} = \lim_{b \to 0} 1/a \) et \( |Z|_{\text{min}} = \lim_{b \to \infty} a/b^2 \)

Le calcul a été fait pour les pavillons conique et exponentiel, et comparé à un calcul numérique exact, basé sur le découpage du pavillon en petits troncs-de-cônes où l'amortissement est supposé constant. L'erreur est en général de 1 à 3%, et atteint quelquefois 10%. Mais il faut noter qu'on est alors bien loin des ondes stationnaires, puisque \( |Z| \) est de l'ordre de 2, et donc le coefficient de réflexion de 1/3 ! Les formules obtenues sont satisfaisantes, et permettent de mieux comprendre le fonctionnement des instruments à vent.

L'exemple du cône \( g = 2/x \) est intéressant, car il fait apparaitre le facteur \( (1 + (kx)^{-2}) \) pour les maximums et non pour les minimums. Celui-ci contribue à baisser beaucoup les pics d'impédance du cône par rapport au cylindre, et explique la mauvaise réaction du tuyau pour les instruments coniques à anexe.
α PARAMETERS IN THE TRANSIENT OF THE BAMBOO TONE, SHAKUHACHI, BY LP METHOD AND ITS SYNTHESIS

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1. INTRODUCTION
This is a continuation of a paper presented at ASA-ASJ joint meeting, Hawaii 1978*. A major concern of this paper is to analyze the behavior of α parameters and its synthesis by means of α and k parameters in a simplified way. As in the previous paper, the model used here consists of an all-pole filter that is excited by a quasi-periodic train of impulses superposed on a random noise source.

2. PROCEDURE AND MATERIALS
Having 2 players, taped pentatonic scale which was used for LP analysis, was performed fast and slow in one breath. The spectrum with 10ms or 20ms window length is plotted every 1.6ms. The synthesis process is based on the taped* and analyzed* pentatonic scale. α parameters of 10 frames are used with the envelope of attack, steady-state, decay to build 5 tones. Using this material, k parameters are introduced and tone quality is improved. A tone burst is represented by 4 frames. k parameters are linearly interpolated at every 3 pitch periods and are converted to α parameters for the synthesis.

3. RESULTS
In case of the upward scale, it could be said that the transient occurs while the finger is leaving the aperture, but for descending, the transient is hardly recognizable, due to the behavior of the instrument. Regardless of speed, its duration lasts about 26ms, except for A to C which is longer and is fluctuated most. The smoothest is from G to A. The low ordered harmonics move in the direction of the nearest harmonic.

One cannot create α parameters. reducing the data, my further aim is to experiment in the manipulation of α parameters to obtain a lively quality of sound.

I thank Mr. T.Takasugi and all staff members of The Radio Research Laboratories for their support.

4. REFERENCE
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Fig.1 Transient from D to F
10 ms window length

Fig.2 Spectrum envelope

Original tones Synthesized tones
REAL-TIME MUSIC SYNTHESIS

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The synthesizer for an advanced electronic musical instrument is described. This instrument, which is intended for real-time music-making by ensembles or for polyphonic performance by a single player, incorporates micro-processors in an essential way, and should largely eliminate knob twiddling and cord patching. Its key features in use are flexibility and ease of control.

The new instrument, like the earlier Timbron family of instruments from which it was developed, is designed so that the players can instantaneously change one or more aspects of the timbre of the sound by operating an actuator. Thus the timbre is under the direct, immediate and infinitely flexible control of the players. A variety of actuators can be used. In addition, the players may interact in ways which are not possible if individual synthesizers are used.

The system has three main parts: the signal acquisition system, in which the signals from the players are preprocessed, the main microprocessor, which transforms these signals under program control into control variables for the synthesizer, and the synthesizer itself.

Flexibility is achieved because the transformation of the control variables is under program control. The main microprocessor deals only with relatively slowly changing control signals, since the essentially repetitive and high-speed synthesis operation is done by the specialized separate hardware of the synthesizer.

The new all-digital synthesizer has 64 channels, each channel consisting essentially of a waveform generator, an amplitude envelope generator and a frequency profile generator. These channels may be combined in groups to form more complex voices (e.g. as in additive synthesis), or their outputs may be used to modulate the parameters of other channels (e.g. as in frequency modulation synthesis). The 64 channels are achieved by multiplexing a single system. The sampling rate is 31.25 kHz, so that only simple filtering is needed at the output.
INTRODUCTION

Artificial piano sound is produced by computer simulation of sound production mechanisms in natural pianos. This method is very useful for studying the relation between the physical elements and the piano tones.

SOUND PRODUCTION MECHANISMS AND SIMULATION

The piano consists of the following mechanisms: (a) transmission of energy from the hammer to the strings, (b) propagation of energy on the strings, (c) transmission of energy from the strings to the soundboard at the bridge, and (d) acoustic radiation by the vibration of the soundboard. These mechanisms are expressed in terms of an equivalent electric circuit model (Fig. 1), which is simulated by computer programs. A nearly half-sinusoidal pulse which is produced by the hammer's stroke, propagates on the special transmission line, and is reflected by the impedance of the bridge and the fixed end. The transmission and reflection lead to the transformation and decay of the waveform. The driving velocity of the soundboard is made by supplying the summing forces of the strings at the bridge to the driving point impedance. This velocity produces the sound pressure whose waveform is determined according to transmission characteristics of the soundboard. Artificial piano sound is produced from calculated waveforms by a D-A converter.

A TYPICAL RESULT

When each fundamental frequency of the group of strings is slightly detuned, beats are produced in each partial which is inharmonic. The changes are slow in the lower partials, but the higher they become, their changes are very rapid. The decay rate of the sound is sharp at the beginning, but it becomes more and more slowly. These are some causes that gives a delicate timbre to the piano.
Continued research on the effect in the finished violin (and other members of the violin family) of tuning several of the lowest bending modes in free plates, which can be adjusted to have a high Q, to certain frequency relationships is indicating the importance of mode #2, the x or cross mode, that occurs about an octave below the tap tone, or mode #5. Findings show that smooth, overall ease of playing results in the finished instrument when mode #2 is at the same frequency in both top and back of a pair of free plates before assembly. If this parameter can be kept constant, then mode #5 in the pair of plates can be so arranged in frequency as to control tone quality to a certain extent. Thus if mode #5 in the top plate is within a tone of the frequency of mode #5 in the back plate a good sounding instrument results. If mode #5 in the top is lower than mode #5 in the back a darker tone quality results; if higher in the top than in the back plate a more brilliant tone results. Brilliant tone quality also results when mode #5 is at the same frequency in both top and back free plates of a pair, but in this case mode #2 must also be kept at matching frequency or a harsh, gritty, hard-to-play instrument results. Implications for tuning mode #5 in the free plates to certain frequencies in relation to the inside higher air modes, as described by E. V. Jansson, will also be discussed.
L'INFLUENCE DU VIEILLISSEMENT DU BOIS SUR LES CAPACITES DE RESONANCE DES INSTRUMENTS A CORDES

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On sait que les propriétés de résonance des instruments à cordes dépendent de la qualité du bois et de son âge. Nous avons considéré dans le présent article comment l'âge du bois influe sur les capacités de résonance du corps d'un instrument à cordes et comment la qualité du ton des cordes particulières change.

Le vieillissement du bois a été provoqué d'une manière artificielle par la radiation des rayons radio-actifs dans quelques étapes. Après chaque radiation les caractéristiques d'amplitude-fréquences de la réponse de la table de résonance ont été enregistrées par la méthode électroacoustique; on a suivi les variations du nombre d'oscillations propres et de leurs fréquences. Par la méthode d'une analyse harmonique nous avons enregistré les spectres toniques des cordes et nous avons observé les changements respectifs provoqués par les changements du nombre d'oscillations propres de la table de résonance et de leurs fréquences.

L'article contient une analyse des résultats experimentaux obtenus.
PRINCIPAL DIRECTIONS OF ELASTICITY OF A BRIDGE ON BELLY AND
HOLOGRAPHIC MAPS OF THE SOUND BOX EXCITED IN THOSE DIRECTIONS

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The points on a bridge where strings rest have their own principal directions of elasticity X and Y as shown in Fig. 1, and the difference of the stiffness in those directions at each point are expressed in the fact that the natural frequencies of the strings in those directions differ slightly as shown in the same figure.

The bowing direction is near but not exactly the same with X-direction of each string.

The above results are reduced by observing the motion of the string which is excited in various directions with electro-magnetic force. (Fig. 2) The lengths of heavy lines on the principal directions in Fig. 1 express the amplitudes at their resonant frequencies in those directions, which means that shorter the length, heavier the damping.

Holographic maps of the sound box were taken by Ar-laser (4880 Å) when a string is excited along each of two special directions of the bridge revealed by the measurement above. Fig. 3 illustrates the vibrational patterns when E-string is excited in those directions. In case of X-excitation, the belly between f-holes rocks and the foot of the sound post at the bottom plate does not move, while by Y-excitation the belly as a whole moves up and down together with the bottom plate.
KALEIDOPHONIC FIGURES OF PLUCKED STRINGS; WHAT THEY TELL ABOUT THE INSTRUMENT

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Hold your eye near an end of a string, and look at some bright spot on it, and then pluck the string. You will see its movement, the shape of which is rapidly changing and soon dies away to a point. We call the locus of such a movement "Kaleidophonic Figure of a Plucked String". The motion is so fast that we can see only its envelope. Such an example is shown in Fig. 1 which is taken by a microscopic camera.

The kaleidophonic figure of a plucked string is to be understood as a modified type of Lissajous' Figure in which, due to the damping of the string, the envelope lines of the original movement make a series of pseudo-parallelograms instead of a rectangular shape. Fig. 2 is a Lissajous' Figure drawn by computer taking damping into account.

If the corner points within each of four quadrants are connected successively toward the center point, there appear four curves all join at the center. These curves are located in the "object and image" positions of two orthogonally oriented mirrors, the cross line of which lies on the center point. Then, we can conclude that the directions of the mirrors X and Y (Figs. 3 and 4) coincide with the principal axes of elasticity of the bridge. The shape of the curves is expressed by the formula

$$\frac{Y}{b} = \left(\frac{X}{a}\right) \frac{\alpha_x}{\alpha_y}$$

where $\alpha_x$ and $\alpha_y$ are the damping factors of X- and Y- directions respectively, and a and b are parameters of the initial plucking direction.

Thus, if we get a kaleidophonic figure of a plucked string, we can determine the principal directions of the bridge and also the ratio of the damping factors inherent to those directions. Two examples are shown in Figs. 3 and 4.

Fig. 1

Fig. 2 $\frac{\alpha_y}{\alpha_x} = 1.5$

Fig. 3 $\frac{\alpha_x}{\alpha_y} = 1$

Fig. 4 $\frac{\alpha_x}{\alpha_y} = 2$
NORMAL MODES OF CHURCH BELLS

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Frequencies and nodal patterns of a good quality modern English church
bell have been measured as accurately as possible for the first
hundred or so partials, together with modal shapes for selected partials.
The results, coupled with finite element calculations, have enabled us
to propose a much improved scheme for mode classification as well as
shedding light on the mechanisms responsible for the production of the
various types of partial identified. This knowledge is necessary in
order to understand why bells have developed their present shape and in
order to make suggestions for improvements in bell design on other than
an ad hoc basis.
ACOUSTICS OF BELLS, GONGS, AND CYMBALS

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Although bells, gongs, and cymbals are basically circular plates, they vibrate and radiate sound in quite different ways. In handbells only the lowest two modes of vibration are normally tuned by bellcrafters, the first overtone generally being the third harmonic of the fundamental. Because of their shape, however, they radiate sound at twice the modal frequencies, giving the sound spectrum a second and a sixth harmonic as well. Carillon bells are considerably thicker, and the profile is carefully shaped so that at least five modes are tuned, often as many as eight.¹

Gongs and cymbals are examples of plates which vibrate in a large number of modes. Gongs convey a fairly strong sense of pitch, whereas cymbals have an indefinite pitch.² Vibrations of cymbals and gongs have been studied in some detail by the use of Chladni patterns, holographic interferometry, probing the near-field sound, and attaching vibration sensors. Some Chinese gongs show an interesting pitch change, the frequency of the principal modes either rising or falling after being struck.³

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The natural frequencies of vibration of an annular membrane depend on the zeros of a cross-product of Bessel functions. The modes turn out to have frequencies which are very nearly exact integer multiples of the fundamental, for a ratio of external to internal diameter up to about 1.25. The harmonicity is very much better than for a full circular membrane of a conventional drum. No tuning cavity, as on a kettledrum, is required. The theoretical harmonicity is at least as good as reported results for stringed instruments such as piano and violin.

Circularly symmetric modes are particularly good. Harmonicity of angle-dependent modes is improved by suppressing lower radial modes. The analysis thus suggests characteristics for construction and playing technique of a musical annular drum of relatively simple construction with excellent harmonic tone quality.
ADAPTATION DE L'APPAREILLAGUE AUDIO A LA NOTION DE "GAMME JUSTE"

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ADAPTATION DE L'APPAREILLAGUE AUDIO
A LA NOTION DE «GAMME JUSTE»

L'appareillage de lecture utilisé en audio fréquence présente des défauts techniques qui entraînent une modification de la fréquence des notes musicales, donc une altération de la justesse supposée rigoureuse. On admet une fluctuation relative de vitesse de 3/00 ; la variation apparente de fréquence est du même ordre.

Dans les conditions pratiques d'exploitation, ce défaut est imperceptible, mais qu'en est-il pour du matériel moins parfait ? Imaginons que l'on tolère une fluctuation de 1 %, d' où une incertitude de 2 à 5 hertz environ pour la hauteur des notes. Ces écarts sont très sensibles pour le musicien, mais qu'en est-il en laboratoire ?

Les discussions sur les gammes de Zarlin, de Pythagore, et sur la gamme Tempérée sont théoriquement justifiées par le souci de respecter les lois mélodiques et harmoniques de la composition, mais on peut se demander comment elles sont respectées en pratique. L'écart n'est que de quelques hertz, donc correspondant à la tolérance la plus sévère pour les fluctuations.

Nous avons réalisé un essai d'un type nouveau, à partir de sons de violon.

PRINCIPE

Il s'agit d'apprécier subjectivement les gammes de Pythagore, de Zarlin et la gamme Tempérée (que nous désignerons dans la suite par les initiales P, Z, T). Ces gammes étaient donc enregistrées préalablement, puis écoutées par de nombreux sujets qui devaient en faire une évaluation chiffrée (de 0 à 10).

Les fréquences des notes étaient calculées d'avance, et le violoniste réglait la hauteur de ses notes sur indications de l'opérateur qui lisait un compteur électronique de fréquence.

Les auditeurs, en groupe, remplissaient un questionnaire comportant trois parties : des présentations isolées, des comparaisons binaires, des présentations ternaires.

RÉSULTATS

Il existe une différence sensible entre la gamme montante et la gamme descendante pour la seule gamme Tempérée. Les réponses sont différentes pour les gammes montantes et descendantes, et même contradictoires sauf pour la comparaison de la gamme de Pythagore et de la gamme Tempérée. Les mélodies montantes et descendantes n'ont donc pas le même sens pour l'auditeur. En moyenne, la gamme de Zarlin est préférée à la gamme Tempérée, celle-ci l'est à la gamme de Pythagore, et la gamme de Zarlin est nettement mieux appréciée que celle de Pythagore. Il s'agissait de discerner la gamme qui différait des deux autres annoncées comme identiques : les auditeurs distinguent nettement les gammes. (Cotes moyennes : Z 6,4 — T 6,2 — P 5,8).

CONCLUSION

On ne peut conclure à la préférence pour l'une de ces trois gammes. Il est difficile de dire que l'une est plus juste que l'autre, mais on a mis en évidence la différence très nette qui existe entre la perception de chacune d'elles. Les variations de fréquence mises en jeu ne dépassent pas les tolérances des magnétophones courants sur les fluctuations de vitesse ; elles influent dans le cas présent sur la perception globale* de la mélodie.

* Référence : Cours de Stéréophonie, par R. CONDAMINES - Masson - Paris 1978
ARCHIVING REPRODUCING PIANO ROLLS

JAMES A. MADDEN ASSOCIATES PTY. LTD.

COOPER STEVEN EDWIN

27 Whistler Street, Manly, NSW 2095 Australia

The Phillips pulse-width system has been developed to record the data from a reproducing piano roll onto magnetic tape at an average of 5000 bits/sec.

The purpose of the system is to archive one of the most important reproducing piano roll collections in the world, as the rolls are deteriorating with age, and to enable other collectors to have access to this collection.

The Phillips pulse-width system records Ampico rolls onto 1/4" mastering tape, and these master tapes are then used to duplicate copies for distribution onto a C60 cassette.

Research is in progress on archiving Duo-Art and Welte reproducing piano rolls, on magnetic tape in a similar format.

The outstanding features of the highly developed system are:

1. Superior performance of the piano by the tape system than by the direct use of the reproducing piano roll, with particular emphasis of the superior soft playing.

2. The data storage format developed allows the use of low speed tape without the need for special high speed equipment.

3. The low cost of the playback system.

4. Small size of the playback system allows the installation of the system into an existing reproducing piano without interfering with the workings of the piano.

5. The installed Phillips pulse-width system does not interfere with the normal operation of the reproducing piano, allowing the piano to operate by cassette or by a reproducing piano roll.
Musical sounds form a special class that differ markedly in structure from either pure tones or noise, the two extremes that are usually considered by acousticians. Methods of analysis that have been devised for pure tones or noise are often unsuited to the analysis of a musical sound which varies markedly with time. It is firstly necessary to describe the macrostructure — usually in terms of the time dependence of the loudness and frequency of each significant partial tone. These dependences may be simplified with the aid of line segments or an approximate envelope function. The microstructure of a note is also important and includes variations in the time envelope, variations in the spectral envelope, temporary appearance of inharmonic partials and the presence of noise components.

The relative importance of the starting transient and the steady state parts of a musical note have been studied by a number of workers. Although it is generally agreed that the starting transient plays an important role in the recognition of a musical sound, the relationships between starting transient and steady state have not been clearly defined. An important role of the starting transient is to draw attention to the sound. In this regard it is useful to plot the derivative of the loudness level, \( \Delta L_t \), versus time. Fig. 1 shows the partial tones having the greatest rate of change of loudness in each 10 ms time band for three organ pipes: (a) Gedackt, (b) Principal and (c) Vox Humana, each of pitch C4 (262 Hz). From the diagram it is possible to determine which partial tones for these particular pipes are dominant in gaining the attention of the ear. For the gedackt, partials 1 and 5 dominate initially; for the principal, partials 2 and 11-13; for the vox humana, partials 1, 3 and 11-13. The partial number is marked against each graph segment.
The oral-cavity function, in the performance of woodwind instruments, has been the centre of conjecture among instrumentalists for many years. It can be shown that any unusual manipulation of the tongue or throat will change the timbre, pitch or even produce multiple sonorities (multiphonics).

Under 'normal' playing conditions two similar spectra will result when simultaneous fourier analyses are taken from within the players mouth and near the instrument's bell. Any unusual alteration to the oral cavity shape will result in considerable changes to both spectra. Thus, it can be concluded that the vibrating reed inside the mouth produces similar characteristic modes or frequencies within the oral cavity as is produced in the air column of the instrument, and both modes are controlled by the oral cavity shape.

Further to this, an x-ray investigation of the throat while playing exhibits uniform 'vowel' shapes for every pitch played. A formant of the accompanying 'vowel' shape usually coincides with the fundamental frequency being produced. Therefore, the oral cavity fluctuations must arise from the need to manipulate the vocal tract 'vowel' formants to match the instruments fundamental frequencies, in the same way as a singer enhances the amplitude of a note by matching it with the vowel formant frequencies.
TRANSFORMING EMOTIONALLY EXPRESSION TOUCH TO SIMILARLY EXPRESSION SOUND

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If a person expresses a particular emotion through the form of an act of touch, such as the expressive pressure of a finger on a finger rest, in a standardized sitting position, characteristic transient forms are obtained constituting biologic dynamic signatures of emotional expression. These forms of 1-5 seconds duration are obtained as vectors of pressure having vertical and horizontal components. A pressure transducer capable of measuring vertical and horizontal components of pressure independently (sentograph) is used for this measurement (see Emotions – Their Parameters and Measurement ed. L. Levi, pp. 561-602, Raven Press, N.Y., 1975).

In order to determine to what extent the specific dynamic expressive forms found for touch expression are also involved by the central nervous system in the generation of expressive sound, to see whether touch and sound have expressive forms sharing common origins in brain function, we may ask the question of how the touch forms need to be transformed dynamically in order to produce sounds of similar expression.

Sinusoidal tones were frequency and amplitude modulated according to the touch expressive forms (essential forms) and the transfer functions were determined by means of an analog computer for the emotions of Anger, Hate, Grief, Love, Sex, Joy and Reverence. It was found that for the slower expressions only the sign and depth of frequency modulation needed to be specifically chosen for each emotion. Faster expressions also required a dynamic correction for the inertia of the arm. When appropriately scaled, the dynamic touch forms were also correct ones for producing like emotional expressive sounds: only when the dynamic forms were thus preserved could the quality of the expression be realized. The parameters for the transformation will be presented and the frequency and amplitude modulation envelopes given for each emotion.

Although the tones were single sinusoids modulated in this way, it seems that much of the emotional range of expression can be encompassed. The sounds will be demonstrated.
Subtleties of the structure of acoustic expressions of human emotion free from linguistic influence have not been extensively studied scientifically. In this study expressive sounds, as frequency and amplitude modulated sinusoids, derived from touch expressions of emotion were tested in a number of ways.

The findings to be described were prompted by a previous study where specific recognition touch expressions of emotion was tested on 232 subjects, who identified them from a film of a hand executing the expressions. Expressions of Anger, Hate, Love, Grief, Joy, Sex and Reverence were correctly recognized by a large majority of subjects both male and female, given the choice of these seven categories. Confidence ratings were systematically higher when correct choices were made. The touch expressions thus appeared to validly express the emotions concerned.

1. These expressive forms were transduced into sound in the manner presented in the previous paper, and were then tested on 100 subjects. Three expressions of each of the seven emotions tested were presented to each subject. The sequence was repeated and subjects were asked to identify the seven different expressive sounds and give a confidence rating for each choice.

2. In a separate study subjects were asked to choose the "best" expression of a particular emotion from a set of nine expressions. This set comprised the expression transformed from the corresponding touch expression as well as eight modifications of this expression created by deliberately varying parameters, such as depth of frequency modulation. Parameter changes were kept slight so as to make the discrimination rather difficult. This process was repeated for each emotion tested. Emotions in both studies were ordered according to a latin square design to avoid biasing results with the order of presentation.

Further, to test the interaction between the sound and touch forms they were superimposed on film together. Subjects were tested similarly with the combined touch and sound expressions.

The results of these studies are significant in showing that the expression of touch and sound have common formal elements in brain function.

*Film produced with the help of Australian Film & Television School, Sydney
RECOGNITION BY ABORIGINALS OF EXPRESSIVE SOUNDS DERIVED FROM EXPRESSIVE TOUCH OF URBAN SUBJECTS
New South Wales State Conservatorium of Music, Music Research Center, Sentic Laboratories
DOBKIN Mark, O'BRIEN Jane, SCHIEMER Greg, TERRAZAS David, CLYNES Manfred
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Expressive forms in touch and sound measured and produced according to the methods described in the previous two papers were presented to northern Aboriginal subjects. These sound forms may function implicitly in musical and vocal communication of various cultures. Although the specific touch expressions had been previously measured in Bali, Japan and Mexico, as well as in the U.S. (Clynes: Ann. N.Y. Acad. Sci., Vol. 220, Art. 3, pp. 55-131, 1973), sound forms generated from these were tested only in Australian urban populations. Would these particular sound and touch forms be recognized by a population of different cultural background? This question was tested by obtaining results from a group of Aboriginals in the Northern Territory. It was expected that if these forms are universal human characteristics they should find their echo at least as readily among Aboriginals as among the groups with whom they were tested originally. The results concerning discrimination between expressive sounds expressing Anger, Hate, Grief, Love, Sex, Joy and Reverence will be presented as well as the results of how Aboriginals selected specific expressions of a particular emotion from a group of similar expressive sounds that had been deliberately distorted in various slight degrees, by changing the parameters. The findings relate to the thesis that these expressive forms are of biologic origin rather than culturally developed, and thus common to mankind.
Beat and rhythm in music incite the feet to dance. Their powerful influence on the form of movement has not been adequately studied scientifically. A theory of beat and rhythm which elucidates the various kinds of energy and qualities of experience engendered has been lacking, particularly in Western Music.

In this paper we shall present movement forms that result from various repeated combinations of single and paired tone bursts of different durations and amplitudes, constituting elemental beat forms or "driving functions". Resulting movement forms are measured as finger pressure transients which are produced rhythmically while listening to the patterns tested. The repeated movements are averaged on a CAT computer and read out on an X-Y plotter for each pattern tested. The patterns thus obtained show a wide variety of forms and forcefulness and are stable for a particular sound pattern. These output measurements delineate how characteristic muscular action is fitted to specific sound "initiators". For example, the lowest point of a down beat synchronizes quite differently with the musical time, depending on the type of beat and rhythm. Differences between Beethoven and Mozart in this respect will be shown, as well as input-output phenomena specific to rock music and acid rock.

The beat of a piece of music may be represented by a particular driving function. Rhythmic movement forms obtained by this method from rock and classical music may be compared with the driving function patterns tested, and it can be shown how the music implies the elemental beat forms or driving functions which produce the movement forms. The durations involved are far more subtle than are capable of being notated through musical notation. The results, combined with previous findings on the property of the central nervous system of time form printing (Clynes: Society for Neuroscience Annual Meeting, Abstract, 1977, and The Communication of Emotion: Theory of Sentics, in Theories of Emotions ed. R. Plutchik, Academic Press 1980) may be regarded as elements for a theory of musical rhythm with neurophysiological foundation.
A BOWED STRING SEEN THROUGH ANAMORPHIC OPTICS OF
A HIGH ASPECT RATIO (with slides and movie)

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Helmholtz has first opened the gate of study on the behavioeur of a bowed string, with his ingenious device — vibration microscope. His experimental observations were fully enriched by his theoretical considerations, very strong enough to establish a number of important conclusions, which had been ever unexpected and have been accepted by the following scientists more than a century since then.

Our report concerns the visualization of this Helmholtzian wave, which is achieved by use of (1) an anamorphic optics of a high aspect ratio (1:20)* and (2) stroboscopic flashes simultaneously (Fig. 1). Fig. 2 will impress readers the ability and effect of the device (1) by presenting both a drill and its anamorphic image side by side. An anamorphic camera properly focussed to a bowed string continuously illuminated records its sweep area with the clear envelope of the bend. Fig. 3 shows four frames from the movie film taken with (1) and (2). You will see a Helmholtzian wave clearly in its sweep area. Fig. 4 shows the existence of two Helmholtzian waves simultaneously on a string** when we see three straight lines having two bends and also the swept area divided into a number of different densities depending on the speed of the string. Fig. 5 is the detail illustration of Fig. 4.

Fig. 1

Fig. 2

Fig. 3

Fig. 4

Fig. 5

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* Report of the 8th ICA
London, 1974, Vol. 1, 346
** Report of the 9th ICA
Madrid, 1977, Vol. 2, 800 and 801
A HIGH ANAMORPHIC VISUAL MONITOR OF STRING VIBRATION DURING PLAYING A VIOLONCELLO

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INTRODUCTION

A stroke of bowing makes on the string a Helmholtzian wave or a number of Helmholtzian waves* simultaneously depending on the ways of playing. But it has been impossible to know directly the types or modes of vibration excited on the string except by hearing the sound. This paper deals with the way of visualizing the envelopes of the Helmholtzian waves on strings being played with a servo-controlled anamorphic optics and a video camera.

VISUAL MONITOR SYSTEM (Fig.1)

A video camera with a high anamorphic optics** is focussed on the area covering all of four strings of a violoncello with the aid of a mirror which is servo-controlled by the movement of the tail-pin of the instrument. On the video monitor, we can see at every moment the vibration modes of the four strings, two examples of which are shown in Illust.1 and 2. In Illust.1, both neighbouring strings are vibrating each with a single Helmholtzian wave but with different amplitudes. While in Illust.2, we can clearly see on the string at the extreme right two Helmholtzian waves which have the phase difference of 120° and the amplitude ratio of 4/3. We hope this system will serve in the both fields of education and self-training.

** One of the papers of our group submitted to this congress titled "A Bowed String Seen through Anamorphic Optics of a High Aspect Ratio"
THE SOUND DAMPING IN A MUSICAL STRINGED INSTRUMENT

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The damping of the vibrations of the chords of a stringed instrument has both internal and external origins. Internally viscoelasticity acts along the chord and at its extremities, whereas external friction is due to the action of the surrounding air and also to the different bodies in contact, whose motion furthermore may play the role of an excitation.

A mathematical model constructed following these lines shows several effects on the generated sound, especially on its spectrum.

The differential equation for the transverse displacement \( y \) of the chord is written in distributional notations, where delta functions are freely used. If \( x \) is the abscissa (derivative \( y' \)) and \( t \) the time (derivative \( y \)), this equation is in its linear form:

\[-(Ty')' - (Dy')' + ky + hy + my = E(x,t)\]

where the coefficients may depend on \( x \) and \( t \). (\( T \) and \( D \), viscoelastic tensions, \( k \) and \( h \) viscoelastic stiffness coefficients, \( m \) distributed mass, \( E \) excitation.

The location of the excitation is important, and it is shown how the theory of characteristic phase lags can help in explaining the differences in timbres which are observed following this location, for instance in the cases of violins and cellos.

As the characteristic phase lag is that existing between a synchronous harmonic mode arising at a given frequency and the corresponding appropriate harmonic excitation, it is obvious that it plays a central role in the type of the actual excitation and the timbre of the resulting sound, and the characteristic phase lags depend predominantly on the type of damping in the instrument. So far, only the timbre of the sound emitted by the chord has been considered. There is a further modification due to the geometry and the dynamic action of the walls of the sound-box which should be considered, but this is beyond the present treatment, in which only the dynamic behaviour of the chord is involved.

It appears however that the definition of a convenient mathematical model of the vibrating chord requires the identification of several damping coefficients besides the elastic ones, and if a linear type of equation is assumed sufficient, the evaluation of internal as well as external damping coefficients, the latter involving not only the end conditions, but also the mode of excitation.

For a freely vibrating uniform chord, of length \( L \), it is shown that the higher harmonics of order \( n \) are critically or supercritically damped out as soon as

\[ h + D(n\pi/L)^2 \geq 2m^{1/2} \left[ k + T(n\pi/L)^2 \right]^{1/2} \]
CLUSTERING OF VIOLIN SOUNDS BASED ON A DISTANCE MEASURE ON FREQUENCY

SPECTRA

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INTRODUCTION

Generally speaking, it is difficult even for specialists to distinguish the difference between the sounds of violins made by Antonio Stradivari(1644(?)-1737) and Joseph Guarneri(1698-1744)?1. Here presented is an automatic clustering of these violins by sound using the k-nearest neighbor (k-NN) method2) in a distance space defined on their frequency spectra.

CLUSTERING PROCEDURE

Sample sounds are taken from a special disk. These are sustained open G tones played by a professional violinist with 6 violins made by Stradivari and 5 by Guarneri in the same recording condition. As a spectrum of each sound, the following three kinds of spectra are employed: (i) averaged spectrum for one second duration through moving Hamming window, (ii) spectral envelope obtained by homomorphic filtering, (iii) spectral envelope obtained by 16-pole linear prediction analysis. The distance between the spectra of sample i and j is defined as follows: 

$$d_{ij} = \sqrt{\frac{1}{M} \sum_{k} \left( F_i(k) - F_j(k) \right)^2 W(F_{\text{max}}(k))}$$

where 

$$F_i(k) = 10 \log|P_i(k)| - 10 \log\max_{i} P_i(k)$$

$$W(x) = \begin{cases} 1+ x/50 & -50 \leq x < 0 \\ 0 & x < -50 \end{cases}$$

$$P_i(k):$$ power spectrum of sample i at discrete frequency f=40000k/N [Hz], N=1024: analysis frame size.

Based on this distance measure, the k-NN method is applied to automatic clustering. Let \( \Omega_k(i) = \{ j_1, \ldots, j_k \} \) denote the k nearest neighbors of sample i. If the condition \( i \in \Omega_k(j) \cap \Omega_k(i) \) is satisfied, the two samples i and j are assumed to belong to the same cluster.

RESULTS

The results of the clustering are shown in Table 1, where quite satisfactory grouping is obtained with only one exception(sample 6). Though the physical feature of each group still remains unclear, once the clustering is obtained, the identification procedure becomes simple with this method.

<table>
<thead>
<tr>
<th>spectral analysis method</th>
<th>(i)</th>
<th>(ii)</th>
<th>(iii)</th>
</tr>
</thead>
<tbody>
<tr>
<td>grouping</td>
<td>(1,2,3,4,5) (6)</td>
<td>(1,2,3,4,5)</td>
<td>(1,2,3,4,5)</td>
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<td></td>
<td>(7,8,9,10,11)</td>
<td>(6,7,8,9,10,11)</td>
<td>(6,7,8,9,10,11)</td>
</tr>
</tbody>
</table>

Samples 1-6 : Stradivari 7-11: Guarneri

REFERENCES

On présente une contribution aux études des propriétés vibrationnelles des instruments à cordes à table de résonance par trois méthodes:

1. électroacoustique, en déterminant les oscillations propres de la table pour différentes fréquences,
2. la méthode optique d'interférométrie holographique enregistrant les figures d'interférométrie de la table vibrante. On obtient ainsi le nombre d'oscillations propres, leur forme et leur position sur la table, et
3. par la méthode de l'analyse harmonique du son on a étudié les changements dans le spectre tonique par suite des changements des oscillations propres de la table et des effets de résonance quand les paramètres physiques (masse et rigidité) de la table changent.

L'article contient une analyse comparative des résultats obtenus par l'investigation simultanée du corps de la guitare par ces trois méthodes.
L. TRANSDUCTION: ACOUSTICAL DEVICES

FOR THE GENERATION AND

REPRODUCTION OF SOUND
DIFFRACTION CONSTANTS FOR PRESSURE GRADIENT TRANSDUCERS

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Conventionally the diffraction constant $D$ of a receiving transducer (microphone or hydrophone) is defined as the ratio of the average sound pressure acting on the blocked diaphragm to the free field sound pressure. This definition is inadequate for transducers that have complex vibration patterns, and, in the absence of ad hoc modifications, it will fail to give the correct results for pressure gradient transducers. A new definition, which overcomes these objections, is therefore introduced. It is:

$$D = \frac{1}{A} \int_{A} P_{n}^{*} V_{n} \, dA$$  \hspace{1cm} (1)$$

where $A$ is the surface area of the transducer, $P_{b}$ is the blocked pressure on the surface, $V_{n}$ is the normal velocity distribution of the electromechanically coupled mode, and $V_{0}$ is the reference amplitude chosen for that mode. The asterisk signifies the complex conjugate of the complex amplitudes $V_{0}$ and $V_{n}$.

As an example of a pressure gradient transducer, consider a flexurally vibrating bilaminar piezoelectric disk mounted inside of a heavy metal inertial ring. Both sides of the disk are exposed to the sound field, and the disk responds to the pressure difference between front and back. The amplitudes $V_{0}$ and $V_{n}$ may be considered to be real for this vibrator. Lamb's diffraction theory is suitable for finding the pressure distribution $P_{b}$ over the disk-ring composite body (modeled as one large disk). The velocity distribution $V_{n}$ over the vibrator is obtained from the deflection curve of an edge-supported disk; over the ring $V_{n}$ is zero. Performing the integration indicated in Eq. (1) extracts from the pressure distribution $P_{b}$ the component that drives the disk in its flexural mode. This component is small compared with the average pressure. The reference velocity $V_{b}$ is taken to be the spatial root mean square of $V_{n}$. Everything is now in hand to complete the calculation of Eq. (1). In a typical case the outside radius $b$ of the ring is 1.5 times the outside radius of the disk; the calculation then yields the result $D = .50 \, \text{kb}$, with $k$ being the wave number.

REFERENCES

Characteristics of the Specular Reflection of a Rectangular Transducer
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Analysis of the directional characteristics of a receiver to a source usually leads to numerical calculation of a complicated integral except in the case of far field. We have succeeded in expressing the receiver to source characteristics for a rectangular transducer in a explicit form which includes Fresnel integrals and several interesting results have been deduced from this formula.

A model for the analysis is shown in Fig.1 where the rectangular transducer of 2a*2b radiates sound waves and receives the sound reflected from a plane located at distance d from the transducer and inclined by an angle θ. Then the received signal is proportional to

$$P = \frac{j \omega Ve^{jut}}{2\pi} \int_{S_1} \int_{S_2} \frac{e^{jkR}}{R} ds_1 ds_2$$  \hspace{1cm} (1)

and it can be rewritten as Eq. (2) after some lengthy calculations

$$P(\theta) = \frac{j}{Z} \frac{e^{-2j\pi F Z/Z^2} + e^{-2j\pi F Z/Z^2} + 2\cos(2\pi F Z/Z^2) F Z/Z^2}{4F Z/Z^2 - \exp(-j2\pi Z/Z^2) - 1}$$  \hspace{1cm} (2)

where $Z = \text{Kasin}\theta$, $a = \sqrt{a^2 \cos^2 \theta / \lambda d}$, $b = \sqrt{b^2 / \lambda d \cos^2 \theta}$, and $F(x)$ shows Fresnel integral. Eq. (2) is a fundamental relation and by introducing appropriate approximation formulas for $F(x)$, more concise formula can be deduced from Eq. (2). Fig.3 and Fig.4 show the far field and near field characteristics of rectangular transducer calculated by using Eq. (2). In the far field Eq(2) can be approximated as Eq. (3)

$$P(\theta)/P(0) = (\sin(Z/Z))^2 + j \pi Z/Z^2$$  \hspace{1cm} (3)

As seen from Eq. (3), the directional pattern coincides with the well-known formula $\sin^2(2Z/Z)^2$ in the case of $\alpha = 0$. In case of $\alpha \neq 0$, the minimum level of side lobes is limited by the second term of Eq. (3), that is, $|\pi Z/Z^2|$. This minimum levels are also plotted in Fig. (2).

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**Fig.1.** A model for the analysis of the directional pattern in far field. ($\alpha < 1$).

**Fig.2.** Directional pattern analysis of the directional pattern.

**Fig.3.** Directional pattern analysis of the directional pattern in near field. ($\alpha > 1$).
Non linear distortion in the transients of loudspeakers
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Many studies consider the transients of loudspeakers due mainly to pulses of different kind that can be supplied either periodically, either singularly, and analyzed respectively as a periodic phenomenon or otherwise require a more elaborate processing.

If the non linear distortion is negligible, the frequency response gives a complete information permitting to determine analytically or by a computer the transient response for any kind of pulse excitation; these calculations are noticeably simplified if amplitude and phase of the frequency response are given separately. One could attempt to specify some properties of the frequency response indicative of the transient behaviour.

To investigate this behaviour, a particular kind of excitation is given by bursts, f.i. of sinusoidal signals: in this case, the theory demonstrates that for a single pole circuit, when the frequency supplied does not coincide exactly with the resonance, the transient is related also to the proper frequency of the pole.

In case of a loudspeaker, account should be taken of the nodes of its equivalent circuit, or better of the fact that this is but an approximation, the loudspeaker being a continuous system: the phenomena are much more complicated than in the previous case.

When non linear distortion is present, the rise of components different from that of the signal supplied may generate beats, amplitude and frequency modulations, combination tones.

Preliminary measurements on loudspeakers in an anechoic room permit a careful investigation of the effects of non linear distortion, if any, combined with the transients. By a convenient non linear electric circuit one can put into evidence the rise of combination tones of the resonance frequencies of the circuit with the signal supplied.

One reports the results of some investigations and measurements: the study of the combined effects of transients with non linear distortion may bring a helpful contribution to the question of evaluating the performance of loudspeakers.
ON THE STUDY OF MINIATURE ELECTRET CONDENSER MICROPHONE FOR HEARING-AID

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INTRODUCTION

Today, condenser microphone for hearing-aid is expecting in order to improve signal to ambient noise ratio. So, this paper describes construction and characteristics of electret directional condenser microphone of the first order pressure gradient type.

CONSTRUCTION & EXTERNAL DIMENSIONS

![Diagram](image)

Fig. 1 Illustration of miniature electret condenser microphone's construction

EXPERIMENTAL RESULTS

![Graph](image)

Fig. 2 Frequency characteristics of miniature electret condenser microphone

CONCLUSIONS

On the basis of these results it become to possible to design and produce electret directional condenser microphone of these external dimensions.
Properties of Prepolarized Condenser Microphones

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For some years, prepolarized condenser microphones (electret microphones) have been offered for sound measurements. After their appearance on the market, papers and reports have been published which compare their merits with those of the well-established condenser microphones using metal diaphragms and external polarization. NBS in Washington has contributed with a comprehensive report (Technical Note 931), describing the behaviour of different microphones; among them there were electret and condenser microphones. The aim of that report was not to compare polarization principles but to compare existing microphones in 1976. Therefore some of the differences reported are not related to the polarization principles but to differences in the mechanical design and the materials used.

This paper deals with differences related to the polarization principle. Two microphones of identical mechanical design and materials will be compared. Both microphones use metal diaphragms for stability reasons. The electret of the prepolarized microphone is placed on the backplate surface (and not on the diaphragm) to eliminate influences of the mechanical properties of the electrically charged polymer which change considerably with temperature and with time under loading.

Electrets of very high stability can be made if proper materials, treatments and charging procedures are chosen. A typical charge decay is 0,1 - 0,3 dB per year in an environment of 50°C and 90% RH. Extrapolation from results measured at higher temperatures indicate a decay rate of less than 1 dB in thousands of years at room temperature in a dry air environment. Thus decay of the charge does not limit the use of electrets in microphones usually used for sound measurements.

Any electret microphone is a condenser microphone with an additional element, the electret. Electret microphones are therefore more expensive to produce than condenser microphones of corresponding quality. In spite of the higher cost, they may be preferred as they offer advantages for certain applications.

The advantages are mainly related to the simplification in the design of subsequent instrumentation in the measuring chain. For transportable - especially pocket-sized - instruments it is important that space and power consumption are minimized. In some cases the reliability of the electronic circuits may be increased in humid environments because of the elimination of the relatively high voltage needed for polarization. Additionally, the extra costs incurred for the cartridge may be made up for in the simpler design of the preamplifier and power supply.

Climatic tests have shown very similar behaviour of the microphones, independent of polarization principle. It seems that prepolarized microphones will gain wider acceptance in future for normal sound measurements. However, for the time being, it does not appear reasonable to develop prepolarized measuring microphones for other purposes, especially not as laboratory standard microphones; for this purpose the design should be as simple as possible.

THE DESIGN CONCEPT OF A NEW SMALL-SIZED RECEIVER FOR HEARING AID

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(1) Introduction
The small-sized receiver to be used in miniaturized hearing aid, requires such essential factors as to be high sensitivity, in spite of its small size, to withstand in bad conditions and to be easily assembled. The new small-sized receiver is designed with ideas concerning improvements in those points.

(2) Design Concept
(a) In order to get large sound pressure, the effective area of diaphragm is made as large as possible. To decrease the stiffness of diaphragm, a balancing type magnetic circuit should be used. The diaphragm and the driving system of receiver must be made in one body.
(b) The construction and the material for it should be considered able to stand long. Titanium foil which does not rust is used for diaphragm, and a rare earth magnet with large coercive force, is used as the magnet.
(c) To make producing easier, the receiver should be designed making all parts in one piece.

(3) Construction And Its Results
Applying the above-mentioned concept, the new small-sized receiver is designed, and its basic construction is shown in Fig.1. The driving forces rise at the place in the gap between the upper-yoke and the diaphragm. The diaphragm with the armature stuck on it, is sustained in proper place in the gap keeping its balance. The frequency response is shown in Fig.2.

(4) Conclusion
The small-sized receiver, designed according to the design concept, seems to show its good features in its response, durability and manufacturing.

![Diagram](image)

Fig.1. Illustration of the basic construction

![Graph](image)

Fig.2. The frequency response
ANALYSIS OF PIEZOPOLYMER MICROPHONES WITH SELF-SUPPORTED MEMBRANES

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INTRODUCTION

Since the discovery of the piezoelectric effect in Polyvinylidenefluoride /1/ this material has found use in a variety of applications such as audio transducers. Piezopolymer microphones and headphones principally consist of a curved piezoelectric polymer film which is mounted in a frame. The simplest systems use freely suspended diaphragms /2/. In piezopolymer microphones sound waves cause combined bending and extensional vibrations of the thin polymer membrane and thus generate an electrical voltage across the metalized foil.

COMPUTATIONS OF SOUND PRESSURE AND VIBRATION SENSITIVITY

First the deformations of the spherically and cylindrically formed piezopolymer membranes were computed for sound and vibration excitation. The curved polymer films are treated as thin shells. In the mechanical differential equations the damping effects of the polymer material as well as the losses due to radiation were considered. The influence of the coupling volume in the back chamber was also included under the assumption that the pressure changes are adiabatic in the cavity. After solving the boundary value problem one obtains normal and tangential diaphragm displacements as well as the mechanical stresses inside the foil as a function of location and time. Then the amplitude and phase of the generated output voltage is determined by means of the general relations between the mechanical stresses and the electrical field strength applicable to poled Polyvinylidenefluoride.

RESULTS AND CONCLUSION

The theoretical as well as the experimental studies showed that there is a significant dependence of the sensitivity and the resonant frequency on the diaphragm geometry. For both cylindrically and spherically formed diaphragms one obtains higher sensitivities for a larger radius of curvature of the membranes while the resonant frequencies are lowered. The influence of the second parameter of geometry, the opening arc angle, is small. By means of these calculations we can optimize the microphone output for a given fundamental resonant frequency or for given overall dimensions.

REFERENCES

DETERMINATION DU SCHEMA ELECTROMECANIQUE D'UN VIBRATEUR TRILAME

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On détermine le schéma équivalent électromécanique de trilame (montage électrique parallèle) vibrant en flexion aux basses fréquences (<20KHz) en n'utilisant que des mesures d'impédance électrique du trilame dans l'air ou supportant des charges acoustiques connues. Les éléments du schéma sont supposés indépendants de la fréquence et des charges acoustiques.

METHODE EXPERIMENTALE

On relève les diagrammes d'admittance de Kennelly pratiquement circulaires du trilame maintenu à température constante (20°C ± 0,1°C) en mesurant précisément les fréquences de résonance série F₀ (conductance maximale) et parallèle Fₚ (fréquence d'antirésonance où la susceptance est nulle) (1) dans les cas suivants :

- trilame dans l'air (sorties acoustiques AM et MB en court circuit). On en déduit le schéma électrique ramené au primaire du transformateur : branche motionnelle r, l, c, shuntée par la capacité Cₚ.
- trilame acoustiquement chargé sur une ou deux faces par des cavités fermées à fond rigide de section s = 4,52 cm², de profondeur d = 31 mm, remplies sous vide d'huile silicone 47 V100 (ρ = 968 kg/m³, ν = 1015 m/sec) et agissant sur MA ou MB comme des impédances acoustiques (2):

\[ Z₀ = jX₀ = -j \frac{\rho ν}{8} \cot g 2mF \frac{d}{v} \]

On en déduit quatre expressions de la valeur du rapport de transformation électromécanique N et donc les éléments R, L, C ramenés au secondaire.

RESULTATS

On a étudié un trilame à lame centrale en alliage d'aluminium AU4G de 0,8 mm d'épaisseur portant deux céramiques de diamètre 20 mm et d'épaisseur 0,5 mm, dans la gamme de fréquences de mesures : 9294 Hz < f < 13620 Hz.

En adoptant une profondeur de cavité d = 32,5 mm, valeur compatible avec les conditions expérimentales (structure de cavité, indétermination sur v ...) on obtient une valeur N = 354 Pa/v commune aux quatre déterminations théoriques. On en déduit, avec Cₚ = 10,9 nF :

\[ R = 1,94 \times 10^7 \text{ kg/m}^4 ; L = 1,59 \times 10^4 \text{ kg/m}^4 ; C = 1,07 \times 10^{-14} \text{ m}^5/N \]

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(2) MERCIER J., Acoustique (P.U.F.) T.1, p. 36
AN IMPROVED EQUIVALENT CIRCUIT OF PIEZOELECTRIC TRANSDUCERS
INCLUDING EFFECTS OF DIELECTRIC LOSS-ANGLE

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As well-known, the dielectric loss-angle, $\delta$, of piezoelectric material decreases the mechanical $Q$ of transducer under the electric-terminal condition of short-circuit and make it smaller than that under the open-circuit condition. This effect cannot be neglected even if $\delta$ is fairly small, for transducers of high electromechanical coupling.

The conventional equivalent circuit of piezoelectric transducers, as shown in Fig.1, cannot represent the effect of dielectric loss-angle on the mechanical $Q$ only by adding a shunt conductance to the clamped capacitance $C_d$, because not only $C_d$ but also $L$, $C$ and $R$ are required to take complex values when $\delta \neq 0$.

We have theoretically derived the equivalent circuits applicable to the cases of $\delta \neq 0$, which consist of only the elements of real constants independent of the frequency. Here we show only the simplified equivalent circuits for the case of large dielectric constant ($\varepsilon = \epsilon_0 > 1$) and small dielectric loss-angle ($\varepsilon = 5 < \pi / 2$). Fig.2 shows an improved four-terminal equivalent circuit, where $Z_B = \frac{r_B + j\omega m + \delta B}{j\omega}$ is the mechanical impedance under the open-circuit condition. The $Z_B$ is related with the mechanical impedance, $Z_A = \frac{r_A + j\omega m + \delta A}{j\omega}$, under the short-circuit condition as $r_B = r_A - (A^2 / \omega |C_d|) \delta$ and $\delta_B = \delta_A + |A|^2 / \omega |C_d|$. From Fig.2, we obtain an improved two-terminal equivalent circuit for the case of mechanically free ($F = 0$) as shown in Fig.3. Another possible equivalent representation is also shown in Fig.4. The series resistance $R_d$ and shunt conductance $C_d$ in these circuits that represent dielectric losses cause the mechanical $Q$ to decrease under the short-circuit condition.

![Fig.1. Conventional equivalent circuit.](image1)

![Fig.2. Improved four-terminal equivalent circuit.](image2)

![Fig.3. Improved two-terminal equivalent circuit.](image3)

![Fig.4. Another representation of two-terminal equivalent circuit.](image4)
PIEZOPOLYMER MICROPHONES WITH RIGIDLY SUPPORTED MEMBRANES
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INTRODUCTION
A number of electroacoustic transducers based on the piezoelectric effect in Polyvinylidenefluoride (PVDF) have been described in the literature. Many of these contain a curved and edgeclamped membrane which utilizes the transverse piezoelectric effect for the electromechanical conversion (1). The membrane curvature is obtained by stretching the PVDF film over an elastic backing or by pre-forming the film.

In the present paper, microphones with rigidly-supported membrane are described. Such transducers have advantages compared to the elastically-supported or self-supported structures.

DESCRIPTION OF THE MICROPHONES
The cross section of a typical piezopolymer microphone with rigidly-supported membrane is shown in Fig. 1. The metallic support plate has 0.5 mm high ring-shaped supports and holes which couple the thin air gap with an air volume behind the support plate. The membrane is a 25 micron thick pre-formed PVDF film. It assumes a curved shape between the lines of support and is thus capable of linear operation. Other designs use one or more point-shaped rather than line-shaped supports.

EXPERIMENTAL RESULTS AND DISCUSSION
The microphones have typically resonance frequencies of about 5 kHz and sensitivities of -55 to -60 dBV per N/m², similar to other piezopolymer microphones (2). The harmonic distortion shows a steady increase with sound pressure from less than 1 percent at 100 dB SPL to about 4 percent at 120 dB SPL. Near resonance, the distortion is about twice as high. The sensitivity decay at 70°C amounts to about 3 dB over a period of 30 days for nonaged microphones and is less for aged systems. Advantages of these microphones as opposed to elastically-supported or self-supported structures are the well-defined geometry of the membrane which is important for reproducible microphone sensitivity, the good mechanical and thermal stability, and the possibility to vary the membrane tension and to introduce damping by the membrane support.

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(2) R. Lerch, J. Acoust. Soc. Amer. 66, 952 (1979)
A NEW THREE-MASS RECIPROCITY CALIBRATION METHOD FOR ELECTRODYNAMIC TRANSUDCERS

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Introduction

A new reciprocity calibration of electrodynamic transducer is proposed. The three-mass reciprocity calibration method is developed. The process requires that the electrical impedance of the transducer coil be measured for three added masses separately mounted on the transducer mechanical terminal.

This three-mass reciprocity calibration only requires one transducer which be measured. This reciprocity calibration also gives phase angle of transducer.

Transducer electrical impedance is measured by ratio transformer bridge. The calibration frequency is 400 Hz and 1KHz. Some results of the experiment and verification are given. The sensitivities agree to within about ±0.5% of a transfer calibration from national medium frequency vibration standard and the three-mass reciprocity calibration.

Three-Mass Reciprocity Theory

The four-pole equation which mathematically describes the whole electrodynamic system of transducer and load are:

\[
\begin{align*}
E_4 &= a_{11} F_4 + (j\omega M_1 a_{11} + a_{12}) V_4 \\
I_4 &= a_{21} F_4 + (j\omega M_2 a_{21} + a_{22}) V_4 
\end{align*}
\]

Finally, Sensitivity of the transducer can be obtained as:

\[
S_4 = (w)^{\frac{1}{2}} \left( \frac{M_{ab}}{M_{bc}} \right)^{\frac{1}{2}} \left( \frac{Z_a}{Z_b} \right)^{\frac{1}{2}} \left( \frac{Z_{ab}}{Z_{bc}} \right)^{\frac{1}{2}}
\]

where

\[
\begin{align*}
M_{ab} &= M_a - M_b, & M_{ac} &= M_c - M_a, & M_{bc} &= M_c - M_b, \\
Z_a &= Z_{a} - Z_{a}, & Z_{bc} &= Z_{c} - Z_{c}, & Z_{bc} &= Z_{c} - Z_{b}.
\end{align*}
\]

and

\[
M_a, M_b, M_c \quad \text{--- three added mass load to the transducer mechanical terminal.}
\]

\[
Z_a, Z_b, Z_c \quad \text{--- drive-coil impedance with mass } M_a, M_b, M_c \text{ respectively.}
\]

\[
a_{11}, a_{12}, a_{21}, a_{22} \quad \text{--- four-pole network parameters.}
\]

\[
F, E, I, V, w \quad \text{--- force, voltage, drive current, velocity, frequency respectively.}
\]

Pole Zero Configuration Measurement for Loudspeakers

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The linear response of loudspeakers is expressed in several ways (the "frequency characteristic is most popular.) It is given in a pattern of curve(s.) Usually, man tries to find features in the pattern. The most appropriate set of parameters will be the pole zero configuration to an integrated understanding of the loudspeaker linear response.

THEORY

Loudspeakers are assumed to have the transfer function

\[
G(s) = \frac{a_m s^m + a_{m-1} s^{m-1} + \ldots + a_0}{b_n s^n + b_{n-1} s^{n-1} + \ldots + b_0}.
\]  

Measure \( G \) at \( m+n+1 \) points of frequency \( s \), and \( m+n+1 \) sets of \( (s,G) \) are known numbers. They make (1) into simultaneous linear equations, which are solved for \( a \)'s and \( b \)'s. Factorizations give the poles and zeros. Actually, a method of search for \( m \) and \( n \) accompanies the solving process.

![Graph showing frequency characteristics.](image)

Pole zero configuration of the loudspeaker having the left frequency characteristics.
A Finite Element Simulation for the Prediction of Loudspeaker Characteristics

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A coupled structural-acoustic system for sound radiation is considered. A vibrating shell of revolution backed by an enclosure on its rear side is to radiate sound waves into a semi-infinite acoustic field in front. A semi-infinite acoustic region has been dealt with as a special boundary for the finite element region and a computer program has been developed[1]. This program is now coupled to a program for the axisymmetric vibrations of the shell through the compatibility condition that volume displacement is continuous on the shell-acoustic medium boundary.

As a numerical example, some frequency characteristics of a direct radiator-type loudspeaker system model is considered. The driving-point impedance of the radiator, the effects of the suspension device, the sound radiation and the backing enclosure on the driving-point impedance, and the far field sound pressure frequency characteristics are calculated. The effects of a so-called vented enclosure and the driving voice coil-bobbin unit attached to the diaphragm are also discussed.

The calculated example to show the effect of the driving unit is shown in Fig. 1. The characteristic resembles a typical one of loudspeaker systems. The sound pressure characteristics are shown for two bobbin thickness, together with the case when the driving unit is absent. With the driving unit attached, the lowest resonant frequency and the sound pressure level as well decrease due to the increase of the effective mass of the vibrating system. The higher frequency response deteriorates and sound radiating shell above a certain frequency due to the coupling between the driving unit.


![Figure 1. Normalized far field sound pressure (air loading in front) --- Without a voice coil-bobbin unit; --- -- with a voice coil-bobbin unit attached \( h = h_0; \), \( h = h_0/6; \), \( m = 285 \text{ g}; \), \( M = 147 \text{ g} \).]
A Phase-Inverter Loudspeaker System with Motional Feedback both from Driver Loudspeaker and Passive Radiator

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INTRODUCTION AND THEORETICAL BASES

In this paper a design method for a generalized direct-radiator loudspeaker system or a phase-inverter loudspeaker system with motional feedback both from driver loudspeaker and passive radiator shown in Fig.1 is first discussed. The low frequency behaviour is controlled by a system parameters vector $X$ in the 20-dimensional euclidean space, whose elements $X_i$, $i=1,2,\ldots,20$ are composed of the parameters given in Fig.1

$$X^T = [X_1, X_2, \ldots, X_{20}] = [r_m, s_0, m_0, r_b, s_b, r_p, s_p, m_p, \beta_d, \beta_v, \beta_b, r_e, g, a_0, a_p, r_e, a_p, \nu_0, \nu_v, \nu_e]$$

Generally, $X$ has constraints which are given by a designer's intention, and is shown as follows:

$$X_{\text{min}} \leq X \leq X_{\text{max}}$$

(2)

In order to realize flat response and at the same time extension of the low frequency response within maximum allowable peak displacements of driver ($\delta_{\text{pmax}}$) and passive radiator ($\delta_{\text{rmax}}$) and within maximum allowable input power ($P_{\text{max}}$), an evaluation function or an objective function is defined, which is related to sound pressure response, displacements of driver-loudspeaker and passive radiator, and input power. By minimization (optimization) of the evaluation function, it is possible to obtain an optimal vector which realizes flat response and extension of the low frequency response.

RESULTS AND CONCLUSION

Computation results of sound pressure responses of this system (MBF 2) are shown in Fig.2 in comparison with those of a phase-inverter loudspeaker system with motional feedback only from driver-loudspeaker (MBF 1) where the constraints are

$$50 \leq r_m \leq 800, \quad 50 \leq r_p \leq 800, \quad 4.2 \times 10^5 \leq s_0 \leq 3 \times 10^6$$
$$7 \leq m_0 \leq 25, \quad 20 \leq r_b \leq 200, \quad 7 \leq m_p \leq 100$$
$$4 \times 10^4 \leq r_e \leq 1.6 \times 10^5, \quad 1.0 \times 10^7 \leq r_{\nu} \leq 1.0 \times 10^9, \quad \delta_{\text{b}} = 3.6 \times 10^5$$
$$0 \leq \gamma, \beta_d, \beta_v, \beta_b, \nu_0, \nu_v, \nu_e \leq 100, \quad 5 \times 10^6 \leq a_0, a_p, A_{\text{p}} \leq 1.5 \times 10^7$$

(3)

and where $\delta_{\text{pmax}}$ is 0.6 cm, $\delta_{\text{rmax}}$ is 1.0 cm and $P_{\text{max}}$ is 50 watt.

This shows that MBF 2 is superior to MBF 1 in extension of the sound pressure frequency response.
Fast calculation of the vibrations and sound radiation of non-rigid loudspeaker cones

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The theory of the sound radiation from a loudspeaker cone, in the frequency region where it behaves as a rigid piston, is well known. At higher frequencies the influence of the non-rigidity of the loudspeaker cone on the sound radiation is of vital importance. In this paper we will give a physical explanation of the vibrations in the loudspeaker cone and their influence of the sound radiation. On studying the vibrational behaviour of a cone or a thin shell we can distinguish "membrane" and "bending" vibrations. It appears that "bending" vibrations only have a minor influence on the sound radiation. This sound radiation is mainly determined by "membrane" vibrations.

The sound radiation of a non-rigid loudspeaker cone cannot be calculated analytically; therefore numerical methods have to be used. The numerical calculations however require a long calculation time on a large computer, so that they are expensive. Responsible for this long calculation time are the bending vibrations. The calculation time can therefore be reduced considerably if we ignore the bending vibrations in the numerical calculations. This model is called the "membrane model". The sound radiation calculated with this membrane model is practically identical to the sound radiation calculated with the complete model, as one would expect from the theory.

![Diagram](image)

Sound pressure level of a 8½ inch woofer calculated with the exact (with bending) and the membrane model (without bending).
A SIMPLE METHOD FOR DETERMINING THE TRANSFER-FUNCTION OF SYSTEMS WITH CONSIDERABLE TIME-DELAY (PERFORMANCE OF LOUD-SPEAKERS IN THE FAR FIELD)

DPG

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Two circuits of a pseudo-random noise generator (maximum-length sequence) are synchronized by the same clock-pulse. By interrupt of the clock input of one generator the two noise sequences are shifted against each other in time. The minimum time shift is 1/f, f = 2 MHz (clock frequency). The maximum time shift depends on the repetition time of the maximum-length sequence (about $2^{31}/2 \cdot 10^6 \approx 1000$ s). It is expensive to realize a delay of such magnitude by shift registers.

For measuring the transfer function (TF) of the system 'loudspeaker - microphone in the far field', for instance, one signal is applied to the loudspeaker, recorded by the microphone, and fed into one channel of a spectral analyser (FFT), which is capable of calculating the TF. The other channel is fed with the signal of the second noise generator. By monitoring magnitude and phase of the TF a minimum-phase condition is adjusted by interrupting the clock pulses. The output of the magnitude and phase is becoming 'noiseless' as the phase condition approaches a minimum because of the coherence of the time intervals in the running FFT.

The TFs of some loudspeakers are compared with subjective judgements of the performance in a preference data analysis.

---

Professional loudspeaker with equalized response in the spectral amplitude, phase shifts $2\pi$ from 100 Hz to 1 kHz

thin line: spectral amplitude; fat line: spectral phase

Loudspeaker, sounding more 'natural' in a preference test, though amplitude response is not flat; phase shifts only about $\pi$ from 100 Hz to 1 kHz
A New Method of Listening Tests on Loudspeakers

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INTRODUCTION

A method of listening tests on loudspeakers that listeners judge the fidelity of the sound reproduction on a true-to-nature scale was recommended by IEC [1]. However, EIAJ (Electronic Industries Association of Japan) provided for a new method of listening tests in which the listeners evaluate the sound quality of sound-reproducing systems on the multidimensional sensory scales consisting of crisp/clear, soft/calm, gay/brilliant, powerful/rich, light, (sound source position) definite/stable, and deep/filling scales [2]. Listening tests using the new method were performed for 6 stereophonic loudspeaker systems and 44 male subjects.

SUMMARY OF EXPERIMENTS

Position of loudspeakers and subjects:
Four stereophonic loudspeaker systems were arranged on a circle of radius 2.8 meters. The angle between two loudspeakers of a stereophonic system was 60 degrees at the center of the circle. Four subjects were arranged near the center of the circle.

Program and evaluation:
EIAJ specified the program which consists of 7 different pairs of music each 20 seconds long for 7 different sensory scales. A subject listened to a pair and judged non-comparatively the sound quality of one out of 6 systems mentioned above by choosing a code in the range of -2, -1, 0, 1, and 2.

Procedure:
For each of the 7 scales, a subject judged 4 systems in a randomized order three times. Each time 2 of the systems were exchanged. A evaluation for a system was calculated from two judgements for the system. Results were calculated by taking the average among 44 subjects.

CONCLUSION

The following results were obtained:
(1) The characteristics of each stereophonic loudspeaker system were expressible on 7 dimensional sensory scales although they were not drastically different from each other.
(2) There were no differences on (sound source position) definite/stable scale among 6 stereophonic loudspeaker systems.

REFERENCE

[1] Listening tests on loudspeakers, IEC-Publication 268-13
[2] Technical File of 4 channel Stereophonic Study Committee STC-004, Engineering Department, EIAJ
Flat Response Finite Plane Baffles

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Shape and size of finite plane baffles are designed whose far field axial response is flat above the cutoff frequency \( f_L \).

**THEORY**

Frequency response \( G(f) = \frac{1}{(1+f/jf_L)} \) is prescribed and converted into impulse response. Sound from the back source is composed of impulses with strength \( p(t)dt \). Each must come through an excess propagation distance \( R = ct \). It is assigned to the length of baffle edge radius vector \( R(\theta) \), where \( d\theta/2\pi \) gives the strength \( p(R)dR \) [1]. Integrating \( p(R)dR = d\theta/2\pi \) gives the intended baffle shape \( \theta = 2\pi(1-\exp(-R/R)) \), where \( f_L \) gives baffle size \( \bar{R} = c/2\pi f_L \). This baffle costs just twice the area of the circular baffle with the same \( f_L \).

**EXPERIMENT**

The response has been measured on both the front and back axes. The distinct cavity resonance behind the baffle has been scarcely found to influence the front axis frequency response; that non-uniformity of \( R \) has spread the influence uniformly over the front axis frequency response.

[1] Inoue T., Finite plane baffles of irregular shape. 9ICA, Q33, p. 841.
A METHOD FOR THE RECIPROCITY CALIBRATION OF PIEZOELECTRIC ACCELEROMETERS
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Introduction
A reciprocity calibration of piezoelectric accelerometer is reviewed. A new method is proposed. This reciprocity calibration only requires two piezoelectric accelerometers through two experiments. The process requires that the voltage ratios be measured for two experiments.

Piezoelectric accelerometer reciprocity formula of resistor shunt

The four-pole equation can be obtained as

\[
\begin{align*}
\Delta E &= \frac{\partial E}{\partial q} \Delta q + \frac{\partial E}{\partial x} \Delta x \\
\Delta F &= \frac{\partial F}{\partial q} \Delta q + \frac{\partial F}{\partial x} \Delta x 
\end{align*}
\]

If this four-pole system is linear, and two accelerometers are connected tightly to each other. A applied force equals the sum of the two forces is:

\[ F = F_1 + F_2 \]

where \( F_1 \) is a force on accelerometer '1' and \( F_2 \) is another force on accelerometer '2'.

Finally, Acceleration sensitivity of piezoelectric accelerometer can be written as follows:

\[
S = \left( \frac{E_1^{11}}{E_2^{11}} \frac{E_1^{11}}{E_2^{11}} \frac{R(m_1 + m_2)}{W} \right)^{1/4}
\]

where \( E_1^{11}/E_2^{11} \) -- Voltage ratio is accelerometer '1' output/accelerometer '2' output, when shaker excites these two accelerometers at the same time.

\( E_2^{11}/E_1^{11} \) -- Voltage ratio is accelerometer '2' output/Voltage drop across resistor R.

\( m_1 \) \( m_2 \) -- mass of accelerometer '1' and '2',

\( R \) -- shunt resistor.

Conclusion
The frequency range for this reciprocity calibration is 1KHz to 8KHz. Sensitivity error is about \( \pm 2.2\% \).

Reference
1. B & K Instruction and Application Type 4290.
ACCURACY OF MICROPHONES USED FOR AIRCRAFT FLY-OVER MEASUREMENTS

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The measurement of aircraft noise is described in the Annex 16 to the convention on International Civil Aviation.

The economical influence of the requirement imposed on aircraft manufacturers, airlines and aerodromes are such that small variation in measurement accuracy is of great importance. Annex 16 is therefore very strict in the requirement to microphone calibration, stability and recalibration.

Improved data collection accuracy has made it possible to compare different measurement situations as well as microphone configurations to within 0.1 dB.

The influence of the configuration and location of microphones will be discussed for different types of microphones in practical use for aircraft certification as well as for permanent airport monitoring.

The use of electrostatic actuator excitation of the microphone enables an overall check of a system in place, while insert calibration ensures a total electrical performance check of the system. The combination of these two techniques should satisfy the most demanding need for reliable data collection.
EXPERIMENTAL ASPECTS OF THE ANAMORPHICAL METHOD IN MICROPHONE CALIBRATION

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This work deals with the main experimental factors governing the accuracy of the anamorphical method in the secondary calibration of cylindrical pressure microphones.

The anamorphical relationships $A(\omega)$ between the instantaneous sound levels picked up by two microphones, placed in equivalent positions of the incident sound field, is obtained electronically by means of a specially designed device. Microphone sensitivities $S_1$, $S_2$ are related with $A(\omega)$ according to the expression $A(\omega)21 = S_2 - S_1 + \Delta P_{21}$, where $\Delta P_{21}$ is the sound level difference on microphones due to diffraction.

Experimental error sources in the measure of $A(\omega)$ are: unevenness of the incident sound field, unbalance between channels of the electronic device, diffraction and errors in $S_1$ (reference microphone). Even under free field conditions, it is impossible to produce an homogeneous incident sound field on the whole frequency range. Nevertheless such inhomogeneities can be compensated in a high degree averaging the $A(\omega)$ values obtained interchanging the positions of microphones. Fig. a) shows this fact for two 1/2" B&K condenser microphones in a free field.

It happens similarly for unbalance between channels of the electronic system. Fig. b) summarizes the results for the microphones of Fig. a).

The diffraction term $\Delta P_{21}$ can represent some additional error because it only cancels out for two identical microphones but increases as discrepancies in form, size and nature of both microphones do. Fig. c) shows $\Delta P$ for two 4134 1/2" B&K microphones and Fig. d) the corresponding $\Delta P$s for the precedent microphones compared with microphones of the type 4133.

As conclusion we can state that the anamorphical method used as a discrete point to point procedure, can lead to accuracies of the order of 0.1 dB in the calibration of laboratory condenser microphones.

ANALYSIS OF ACOUSTIC RESPONSE OF ELECTRET MICROPHONES
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INTRODUCTION
The electret microphone has found increasing use in measurement applications over the past several years. It operates on the same principle as a normal condenser microphone but does not require an external polarization voltage. The electric field is generated by an electret, for example Teflon (1).

Fig. 1 shows the cross section of a circular electret microphone. Placing the Teflon electret on the backplate allows one to choose a membrane material with best mechanical properties because electret and membrane functions are separated. We used Mylar membranes in our calculations.

RESULTS AND CONCLUSION
The acoustic response of electret microphones is affected by several mechanical and electrical parameters such as thickness of the polymer films, separation distance of the supporting rings, membrane material properties and tension and backchamber volume (2).

It is shown that a small separation distance between membrane and backplate causes high damping of membrane displacement. One the other hand a small air gap is desirable because a higher output voltage is generated. Geometrical arrangement and dimensions of the holes also influence the vibration damping. If only one ring of holes is used the damping minimum can be reached by placing the ring approximately in the middle between supporting rings and membrane center. If the spacing is small, microphones with large membrane radius need two or more rings of holes to show a flat frequency response. A larger radius causes greater displacement and a lower resonance frequency. A higher membrane tension causes a higher resonance frequency but a lower sensitivity.

Most of the aforementioned microphone parameters are coupled with each other. Therefore it is quite difficult to get a smooth and wide frequency range. In our analysis circular electret microphones with flat frequency response for audio, telephone and general purposes are discussed.

(2) A.J. Zuckerwar J. Acoust. Soc. Am. (1978), 64, 1278
Microphone Calibration Using Laser Doppler Measurement of Particle Velocity
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A number of techniques have been used, with varying success, for calibrating microphones and, although reciprocity is well established as the most popular, precise, method, it is still interesting to investigate others. One reason for this interest is the possibility of identifying systematic errors inherent in any particular method. In the National Measurement Laboratory, we have been investigating one such alternative technique which relies on determining the sound pressure in a field by knowing the acoustic impedance and measuring the particle velocity by laser Doppler methods. This is a variation on a previous method which used Rayleigh discs to measure the particle velocity.

Laser Doppler methods offer the special virtues of (a) an inherent absoluteness based on simple theory, and (b) the fact that a measurement can be made without introducing anything more than some small light-scattering particles into the field. A number of improvements have been made in our method since we first reported it in J. Acoust. Soc. Amer. in 1977 so that velocities of about 5 mm s\(^{-1}\) r.m.s., at frequencies around 500 Hz, can be measured with standard deviations of the order 0.02dB and an expected accuracy of about the same magnitude.

In order to apply the laser Doppler technique to the calibration of microphones, a sound field is established as a travelling wave in a 4 m long transparent pipe which is terminated with an almost perfectly absorbing plug. The microphone is mounted with its diaphragm flush with the wall of the pipe and the optical system is so arranged that particle velocity can be measured in the vicinity of the microphone. Measurements of particle velocity at positions along the length of the pipe and over its cross section show that the standing wave ratio is small, as are the losses in the pipe. Provision is made for monitoring temperature and humidity of the air in the pipe during the experiment so that one can then estimate the specific acoustic impedance from calculations of density and sound velocity.

Results obtained, thus far, indicate an agreement between this method and reciprocity calibrations of better than 0.2 dB with the discrepancy tending to be in the direction which attributes a lower sensitivity to the microphone when it is determined by the reciprocity method.
THE ACCURACY OF CALIBRATION OF STANDARD MICROPHONES AT HIGHER FREQUENCIES

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INTRODUCTION

The reciprocity method is generally accepted as the most accurate method for determining the sensitivity of laboratory standard microphones. The general principles for pressure- and free-field calibration are given in literatures 1 and 2. In both documents it is stated that the overall accuracy of a calibration will decrease at higher frequencies. It is the purpose of this paper to discuss the reasons for this and indicate methods to increase the accuracy.

PRESSURE CALIBRATION

In literature 1 two different approaches are given for calibration at higher frequencies, i.e. up to 10 kHz for 1 inch microphones. One method is based on using a relative large coupler volume of 20 cm$^3$ filled with hydrogen or helium combined with an empirically determined wave-motion correction. Various attempts on a theoretical determination of wave-motion are not convincing and, furthermore, the wave-motion is influenced by the acoustical impedance of the microphones. Basically, the use of a coupling gas other than air and the application of a wave-motion correction is not a sound principle.

The other method makes use of a plane-wave or transmission line coupler having a volume of about 3 cm$^3$, in which case the source- and load impedance, i.e. the acoustical impedance of the microphones plays an important role for the transfer characteristic of the system. The necessary accuracy for the determination of the acoustical impedance is discussed and various methods for obtaining these data are mentioned.

The precision of the two methods, the mutual agreement and the resulting accuracy are finally discussed.

FREE-FIELD CALIBRATION

Apart from the problems arising from the metrology used under the calibration, the quality of the anechoic room, cross talk and bad signal-to-noise ratio, the accuracy of a free-field calibration mainly depends on the determination of the distance between the acoustic centres of the microphones. The philosophy behind the concept 'acoustic centre', its dependence on frequency, direction and distance from the microphone will be discussed, as well as the overall accuracy of calibrations.

Literature 1: IEC publication 327: Precision method for pressure calibration of one-inch standard condenser microphones by the reciprocity technique.

Literature 2: IEC publication 486: Precision method for free-field calibration of one-inch standard condenser microphones by the reciprocity technique.
Reciprocity calibration of microphones depends on measuring the ratio of the output voltage from one microphone to the input current entering another, i.e. on measuring an electrical transfer impedance, and the reciprocity apparatus at NML embodies some of the principles of electrical impedance measurement to improve the practice of reciprocity calibration as traditionally carried out. The resolution is better than 0.005 dB and reproducibility is largely determined by the stability of the microphones themselves which, in favourable circumstances, can exceed 0.01 dB over periods of months.

In addition to the 3 cc and 20 cc couplers prescribed by the IEC, a novel 3-aperture coupler of approximately 10 cc volume has been constructed, enabling the rapid calibration of a group of three microphones with three measurements of electrical impedance. Measurements of equivalent volume are made by substituting a metal plug having a cavity of appropriate shape and volume for one of the microphones whilst monitoring the apparent change in sensitivity of the remaining pair. Calibration of unknown microphones may be carried out by a similar substitution technique without requiring knowledge of their equivalent volumes.

Since three microphones cannot be simultaneously mounted coaxially, no theory is available to predict the high frequency behaviour. Comparison with the 3 cc IEC coupler however reveals that up to 2 kHz, the corrections needed for "wave motion" are of the same order, i.e. less than 0.2 dB.

Measurements have been made at lower frequencies down to 31.5 Hz, and "heat conduction" corrections intermediate between those required for the 3 cc and 20 cc IEC couplers have been found adequate to reconcile results from the 3-aperture coupler with these. Routine monitoring of the primary microphone group is carried out at 250 Hz where the composite correction is less than 0.05 dB.

An interesting by-way disclosed by these measurements is the discovery that 1-inch capacitor microphones are 0.1 dB more sensitive at 31.5 Hz than they are at 250 Hz. This is confirmed by "electrostatic-actuator" measurement and is speculatively attributed to a "heat-conduction" phenomenon occurring in the air volume behind the diaphragm.
Vermeidung von akustischer Rückkopplung durch Frequenzverschiebung.

Verein Deutscher Ingenieure, Deutsche akustische Gesellschaft

Dr. Arns  Ulrich

Prof. Dr. Ing. U. Arns, Fachhochschule, D-7100 Heilbronn


Eine solche Begrenzung der zulässigen elektronischen Signalverstärkung tritt aber nicht ein, wenn eine synchrone Zeitverschiebung der Signale durchgeführt wird, mit der sich eine Untersuchung befaßt, über die dieses Referat berichtet.

CARBON MICROPHONE MEASUREMENTS BY AN ARTIFICIAL SPEECH SIGNAL

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The objective measurements that can be conducted on non-linear systems depend on the type of the excitation signal used.

Telephone carbon microphones are intrinsically non-linear and varying with the time. Thus, it would not be correct to talk about the sensitivity/frequency characteristic, mathematically established for linear systems, which do not vary with time. However, it may sometimes be interesting to make use of a measurement, based on the concept of the sensitivity/frequency characteristic.

If a system is per se non-linear, the data can be obtained by exciting the system through a signal the characteristics of which are as close as possible to those of the excitation signal used in practical cases.

The signal exciting telephone sets in common application is speech. Thus it would be expedient to conduct objective measurements by a signal having spectral and time characteristics similar to those of speech and mathematically defined. The essential characteristics of a voice signal for this purpose are:

1) the power spectrum density, averaged on a long-term basis over various persons saying different sentences;
2) the amplitude probability density (in which the ratio peak value/RMS value is equal to ± 4.5);
3) the periodicity of the signal when it is voiced;
4) the presence of intervals of quasi silence and signal.

A pseudo-speech signal with these characteristics as been implemented, using an ad-hoc designed generator.

Fig. 1 shows sensitivity/frequency characteristics obtained by 1) real voices, 2) sinusoidal signal 3) pseudo speech signal.

The artificial signal can also be used for determining the distortion introduced by carbon microphones.

The distortion of output signal depends from the input signal and may be considered as a summation of a linear component (related to the response characteristic) and the non-linear amplitude distortion.

The artificial signal mentioned above permits the identification of these components. In fig. 2 'e' represents the difference between the actual output 's' from the carbon microphone, when an excitation pseudo speech signal is applied to the input, and the output 's' from a linear filter having the same excitation signal at the input.

The linear filter is determined so as to minimize $e^2$

$$e^2 = \langle (s - \hat{s})^2 \rangle$$

where $e^2$ is the distortion.

The response characteristic of the linear filter can be considered as the sensitivity response of the carbon microphone.

Measurements on different types of carbon microphones have been carried out.

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**Fig. 1:** Sensitivity/frequency characteristics by

- real voices
- pseudo speech signal
- sinusoidal signal

**Fig. 2:** Schematic method to measure linear and distortion components of carbon microphone

* Patent pending
CORRELATION BETWEEN LOUDNESS AND QUALITY OF STEREOPHONIC LOUDSPEAKERS

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INTRODUCTION
Comparing the loudspeakers with objective and subjective measuring methods Staffeldt found /1/, that there is no correlation between the loudness calculated according to Zwicker’s method, and the subjective testing of the loudspeakers. This conclusion seems to be contradictory to the one observed in the practice of electroacoustical transmission. The authors have revised this conclusion using the results of an international stereo comparative loudspeaker test /2/.

METHODS
By means of five pairs of studio monitoring loudspeakers 30 records /25 sec each/ were listened to and tested /3/ by 17 sound engineers and musical producers. Correlation was calculated between the subjectively tested loudspeakers’ quality and the computed /4/ loudness of the records observed at the listening points. At the given listening point the loudness was determined from the power spectral density /P_H/ derived by /5/:

\[ 10 \log P_H/\Omega_n/ = 10 \log P/\Omega_n/ + 20 \log H/\Omega_n/ \]

where \( P \) is the power spectral density of the record and \( H \) is the transfer function of the loudspeaker. Indexes in /1,9/ represent the listening points and \( n \in /1,26/ \) indicate the number of 1/3 octave bands /\( \Omega_n/\). The records were of six categories. The computations with the aim of binaural summation /6/ were made for all variations of the listening points, records, and loudspeaker types.

RESULTS
1. Correlation has been found between the loudness of records and the results of the loudspeakers’ quality testing. In 3 out of 6 recording categories /symphonic-, pop-music and speech/ the correlation was: \( \eta > 0.5 \).
2. When evaluating the subjective quality judgements of loudspeakers’ testing we found that they were influenced by listening conditions; physical parameters and personal characteristics.
3. Loudness was taken into consideration at a greater extent by sound engineers than by musical producers.
4. When used pink-noise instead of listened records a high correlation was found between the calculated loudness /see term \( P \) in Eq. 1./ and the subjective quality judgements.

REFERENCES
EQUALIZATION IN ARTIFICIAL-HEAD RECORDING

FOR LOUDSPEAKER REPRODUCTION

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INTRODUCTION

The equalization of the frequency response of the artificial-head, considering the compatibility with the loudspeaker reproduction, and the quality of the headphone reproduction of the equalized artificial-head recordings are discussed.

EQUALIZATION OF ARTIFICIAL-HEAD

In general, a large peak exists in about 2 - 5 KHz range of the frequency response of the artificial-head and in case of the loudspeaker reproduction, timbre will be quite unnatural. Therefore, in the field of broadcasting or disc recording, the equalization is necessary in the recording process. In this case, it will be sufficient for practical use that the frequency response of the artificial-head for frontal incidence is equalized approximately flat. Fig.1 shows the equalized frequency response of the artificial-head, in which the microphones are mounted at the entrance of the blocked ear canals. However, on the other hand, there is a problem of whether the quality of the headphone reproduction, because of the equalization, will be impaired or not.

LISTENING TESTS IN HEADPHONE REPRODUCTION

Listening tests were carried out on five different types of headphones with and without the equalization. Three acoustic engineers were selected as listeners. The items evaluated were such as; localization in the horizontal plane, elevation and distance of sound image, and also timbre referred to the original sound source. The results of the tests with the equalization, compared to that without the equalization, are summarized as follows: (1) In all of the five headphones, the reproduction quality did not, at least, degraded. (2) In a few of headphones, especially the No.2 (open-air type, see Fig.2), timbre was improved close to that of the original sound source, and also elevation of sound image was suppressed even though slightly. Fig.2 shows the frequency response of the headphones, measured in the free field by means of loudness comparison with a frontal loudspeaker of flat frequency response as a reference. In a headphone of flat frequency response measured in this way, the most natural hearing sensation can be expected in case the frontal frequency response of the artificial-head is equalized flat, and this is endorsed by the above mentioned results.
PERCEIVED SOUND QUALITY OF SOUND-REPRODUCING SYSTEMS

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Equipment for sound reproduction is usually described in terms of various physical parameters. However, such descriptions are not enough to understand what the perceived sound quality will be. The perceived sound quality of sound-reproducing systems is probably multidimensional, that is, it is constituted by a number of separate perceptual dimensions. It may be possible to give a perceptual description of sound-reproducing systems by stating their positions in such dimensions. A research project has been performed with the following goals: (a) to find out and interpret the meaning of relevant dimensions within perceived sound quality, (b) to find out the positions of the systems in these dimensions, (c) to explore the relations between the perceptual dimensions and the physical characteristics of the systems, and (d) to explore the relations between separate perceptual dimensions and the overall evaluations of the systems.

Subjects of various categories judged loudspeakers, headphones, and hearing aids with regard to their reproduction of selected programs of music, speech, and daily life sounds. Either the systems were presented in pairs for judgments about the perceived similarity between the two reproductions (these data were analyzed by multidimensional scaling techniques), or each system was presented separately and judged on many selected adjective scales (subjected to factor analysis). Free verbal descriptions were added.

The combined results of eight experiments suggest the following dimensions to be constituents of perceived sound quality: Clarity/Sharpness/Harshness - Softness, Brightness - Darkness, Fullness - Thinness, Feeling of space, Nearness, Disturbing sounds, and Loudness. Approximate positions of the systems in these dimensions may be found by the solutions from multidimensional scaling and factor analysis. This makes it possible to explore the relations between the perceptual dimensions and the physical properties of the systems. Continued experiments dealing with the validation of the suggested dimensions and other aspects will be reported.

REFERENCES

MEASUREMENT OF THE PREISACH DISTRIBUTION IN MAGNETIC TAPE

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INTRODUCTION

Since the German scientist F. Preisach in 1935 presented his hypothesis on the distribution of domains in a magnetic material, the hypothesis has been used by several authors in the last three decades in a model describing the magnetic recording process. However, only a limited number of papers deal with an experimental determination of the distribution of domains - the Preisach distribution - in magnetic tape, partly because of the great number of tedious measurements involved. It is the purpose of this paper to describe a fast method of measurement using an on-line digital computer.

MEASUREMENTS

Various signals to be recorded on the tape were investigated, but the signal proposed by H. Völz already in 1966 seems most suitable. The signal consists of a single step sawtooth modulating a 40 kHz bias oscillator. The length of the sawtooth was chosen to 5 ms and the repetition rate about 1 s. The replayed signal was sampled digitally (128 samples) and an averaging over 16 signals was carried out. During recording the resulting peak value of the input signal was kept constant (well above saturation of the tape), while the ratio of the maximum of the sawtooth to the maximum amplitude of the AC-bias were changed stepwise from 0 to 1. The corresponding set of output signals each represents the total density below the corresponding straight line in the Preisach diagram. Thus, from the stored data the Preisach distribution can be calculated.

A variety of old and new tapes have been measured with an excellent reproducibility. Of greatest interest is the presence of one or two areas with a negative density for all the new tapes. Although a negative density in a strict sense cannot be present, according to the hypothesis, it has been known for high permeable materials for some time. The existence of the negative densities in the measured Preisach distribution has been further approved here by calculation of the output signal as function of bias current for a constant value of the low-frequency signal. Only when the areas with negative densities are taken into account the calculated curves agree with measured values.
OCCLUDED-EAR SIMULATOR FOR THE MEASUREMENT OF EARPHONES

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INTRODUCTION

For objective measurements of earphones coupled to the ear by ear inserts simulators might be used provided these simulators simulate the acoustical behaviour of the portion of the ear canal between the tip of the earmold and the eardrum, including the acoustic impedance of the eardrum of a median human ear.

For the exchange of specifications and physical data of the earphones the properties of the simulators should be standardized. In the IEC-Subcommittee 29 C, Measuring Devices, working group 6 worked on this problem since 1971. In an earlier paper details for the requirements of ear simulators are given (1). At the last meeting in Stockholm, in May 1979, a final proposal was made for an occluded-ear simulator in the frequency range from 100 Hz to 10 kHz.

DESCRIPTION AND SPECIFICATIONS

The simulator shall be constructed of non magnetic and stable material. It consists of a special cavity and a cylindric tube simulating the ear canal with an effective volume of 1.26 cm³ at 500 Hz. For measuring the sound pressure in the simulator a high-impedance half-inch condensor microphone with known sensitivity is used. The acoustic impedance of the human eardrum is simulated by special acoustic networks. The magnitude of the acoustic transfer impedance, i.e. the ratio between sound pressure at the microphone location to the volume velocity at the entrance of the simulator, is given.

PERFORMANCE TEST

The transfer impedance as function of the frequency can be simply controlled by measuring the sound pressure level at the microphone location when a source with constant volume displacement at the entrance is applied. As source a small condensor microphone (1/4-inch type) may be used. The resulting sound pressure level curve can easily be corrected by the frequency, according to the difference between velocity and displacement.

CONNECTIONS

For the coupling of the different kind of earphones, (f.i. insert type, behind-the-ear type, in-the-ear type, real ear moulds, real ear inserts) to the occluded-ear simulator examples are given.

REFERENCE

APPLICATION OF THE LPC SYNTHESIZER TO THE PUBLIC ADDRESS SYSTEMS
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INTRODUCTION

Automatic public address systems (PAS) are practically being used in the subway stations and the airports in Japan. These are the multichannel systems installed in the information centers, and rather expensive systems. Recently, economical and small-sized systems have been required. These systems will operate individually in each rail-way station, or in each story of the department stores. These systems require low price, high sound quality, and high reliability. But single output channel may be enough. To meet the requirements, the LPC speech synthesizer has been introduced and the functional model has been composed.

THE LPC SYNTHESIZER

The ten-stage lattice filter, whose 10-reflection coefficients represent the spectrum of a speech sound, has been introduced. The ten coefficients and excitation parameters (pitch period and energy) have to be extracted from the speech signals and stored into the ROM’s. Factors which exert a serious influence upon synthesized speech sound are the excitation waveform, the bit allocation of the reflection parameters, and the analysis frame shift. In this system, to get rich, bright, and comfortable sound, and not to produce buzzy speech sound, the excitation waveform of one pitch period of the prediction residual has been put to use, and also, the bit allocation of (10, 8, 6, 6, 6, 5, 5, 4, 4) and the frame shift of 5ms have been provided.

SYSTEM SET-UP AND BIT REDUCTION

The system set-up of the PAS is shown in fig. 1. The µ-CPU reads coded parameters from the ROM’s and decode them. The µ-CPU also controls the LPC synthesizer by transmitting the decoded parameters. To reduce the memory size, two technics have been developed. One is to vary frame shift, another is to vary frame rate according to the distance between the code vectors of the parameters. Consequently, the memory size has been reduced by 30% or 40% with almost no degradation in sound quality. The µ-CPU selects the frame shift (5ms or 10ms) and controls frame rate automatically.

CONCLUSIONS

The LPC speech synthesizer has been applied to the automatic PAS and the functional model has been composed. Frequency band-width was about 5kHz, data rate was about 10kb/s for Japanese female voice. The system with lower data rate could not be applied to the automatic PAS.

Fig. 1 System set-up of the automatic PAS.
THE THÉVENIN ACOUSTIC IMPEDANCE AND PRESSURE OF DUMMYHEADS

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INTRODUCTION We have studied the equalization for the dummyhead-headphone system capable of reproducing true directional information[1]. The two elements in the expression for the equalizer, i.e., the Thévenin acoustic impedance and Thévenin pressure of the dummyhead, are examined for the purpose of realizing a standard dummyhead for acoustic measurements.

THEORETICAL BASES The equivalent circuit of the dummyhead is shown in Fig.1. When two known impedances $Z_1$ and $Z_2$ at $a\alpha'$ in Fig.2 are connected successively, $Z_S$ and $P_S$ are obtained as follows:

1. $Z_S = \frac{(P_2/P_1) - 1}{(1/Z_1) - (P_2/P_1)(1/Z_2)}$  
2. $P_S = \frac{Z_1 - Z_2}{(Z_1/P_1) - (Z_2/P_2)}$

EXPERIMENTAL TECHNIQUES A dummyhead was provided whose ear canal was terminated with the variable impedance apparatus of realizing the two known impedances $Z_1$ and $Z_2$. $Z_S$ and $P_S$ were measured using the dummyhead in an anechoic room. The measurements were made at four different azimuthal angles of incidence, $\theta=0^\circ, 30^\circ, 60^\circ,$ and $90^\circ$.

RESULTS AND CONCLUSIONS Measurements on several dummyheads were made. Measured and evaluated values of $Z_S(x=0)$ for the spherical dummyhead are shown in Fig.3. The result obtained shows a good agreement with the evaluated value. It suggests that $Z_S$ obtained is reasonable. Results of other measurements will be given in details in the full written version of our presentation.


Fig.1 Thévenin equivalent circuit of the dummyhead.

Fig.2 Thévenin's equivalent circuit terminated in a known impedance $Z$

Fig.3 $Z_S(x=0)$ for the spherical dummyhead. The solid line indicates the computed value of the acoustic radiation impedance of a piston set in a sphere of the same radius as used here.
M. ACOUSTICAL MEASUREMENTS AND INSTRUMENTATION:

SIGNAL PROCESSING:

STATISTICAL METHODS IN ACOUSTICS
ANTENNES ACOUSTIQUES DE CHAMP PROCHE : REALISATION ET UTILISATION
D'ANTENNES DE FLUX -

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L'opération d'analyse des sources de bruit, rencontrées dans l'industrie, se heurte en général à des difficultés de deux ordres :

- Difficultés pratiques liées à l'environnement de la source considérée,
- Difficultés de principe concernant le choix du mode de représentation des champs acoustiques complexes rencontrés.

On présente ici une technique d'analyse permettant de surmonter ces difficultés.

Cette technique fondée sur l'utilisation d'antennes acoustiques planes permet d'effectuer des mesures à proximité immédiate des sources et de caractériser les champs observés en terme de flux d'intensité acoustique locaux, associés à des "paquets d'ondes" sélectionnés dans le cône de directivité de l'antenne.

Les informations locales et directives ainsi recueillies donnent accès :

- Par sommation, au champ acoustique lointain,
- Par projection géométrique des vecteurs d'onde moyens sur la source, aux composantes de surface responsables du bruit.

Les principes ainsi que les modes de réalisation de telles antennes sont explicités et illustrés par des exemples de construction.

On montre en outre à l'aide d'exemples de mise en application l'intérêt et le systématisme d'une telle procédure de caractérisation :

- Application à l'analyse du bruit émis par des moteurs diesel,
- Application à l'analyse du bruit émis par des presses.

Des cas d'utilisation sur des sites industriels, fruits de l'expérience de l'équipe acoustique de la Direction des Etudes et Recherches d'Electricité de France sont également évoqués.
THE APPLICATION OF CEPSTRAL TECHNIQUES TO THE MEASUREMENT OF COMPLEX REFLECTION COEFFICIENTS IN SITU

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Objectives

A measurement system has been developed which allows measurements of complex reflection coefficients of plane surfaces at oblique angles of incidence. The technique does not disturb the surface, relies on a single measurement chain only, and does not require precise knowledge of the geometrical configuration. A loudspeaker is used to emit a pulsed signal thus permitting the accurate definition of the active reflecting area, and allowing measurements to be made in non-anechoic environments. The technique has been designed so as to allow the field measurements to be made simply and with minimal equipment requirements.

Techniques

A single microphone senses the direct and reflected pulses and a recording is made for subsequent analysis. The pulses used are of relatively long duration (of order 10 ms) and have spectra which are essentially flat over the range 100 Hz - 10 kHz; both deterministic and random signals have been investigated. Long pulses have been chosen to minimize window effects and achieve adequate sound pressure levels in the field. As a consequence of the pulse duration, there is considerable overlap of the direct and reflected signals.

Complex cepstrum techniques were used to deconvolve the direct and reflected pulses; other techniques have also been investigated and will be described. In the initial stages of this study the acoustical system was modelled by an electronic system of known characteristics (delay, impedance) and this was used to test the validity of the algorithms developed. The simulation has also been used to test the sensitivity of cepstral techniques to the presence of additive noise as occurs in field acoustical measurements.

Experimental trials were conducted using fibrous absorbing materials and the pressure reflection coefficients obtained. From these, plane wave reflection coefficients have been derived and compared with existing data.

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The normal specific acoustic impedance of sound absorbing materials is conventionally evaluated from results of standing wave ratios and positions of minima in a Standing Wave Tube. The process of determining the impedance as a function of frequency can thus prove to be lengthy. Normal impedance information can also be derived from transients. If a transient sound wave is propagated towards an absorptive termination, then a measurement of the resulting reflected wave will provide sufficient information to allow the evaluation of the normal impedance of the termination.

Experiments using both techniques were conducted on three materials. They were i) Fiberglass (bulk density 48 kg/m³), ii) Estafoam EF 570 and iii) Meracell foam 080. A B & K Standing Wave Apparatus was used for the standing wave technique. Normal absorption and the normal specific acoustic impedance of the materials were evaluated for the frequency range of 0.2-5kHz. The transient technique involved the use of a spark generator, a metre long brass tube (0.3 1.1/4" and thickness 1/8"), and a B & K 1/8" microphone Type 4138. The spark impulses were digitally processed using Fourier Transforms to yield the normal impedance.

The values obtained for fiberglass from both techniques were identical. The absorption coefficient was practically uniform and the impedance was almost purely resistive for frequencies above 1kHz. Estafoam; a distinct absorption peak was evident with a corresponding change in the impedance characteristic at the frequency of 2.2kHz. A dip in the absorption coefficient manifested itself as a peak in the resistive part and a phase reversal in the reactive part of the normal impedance. The standing wave technique yielded more pronounced changes, i.e. the troughs and the peaks were more distinct. The presence of peaks and troughs highlighted the quarter wavelength resonance phenomena due to sample thickness. Meracell; for this material, a peak in the resistive part of the impedance occurred at 3kHz. A phase reversal in the reactive part followed suit. However, the absorption characteristic did not exhibit pronounced peaks and throughs. Values obtained through the transient technique were consistently larger throughout the frequency range.

The applicability of the spark technique was restricted to linear responses of the materials to the sound wave. It is feasible that nonlinear responses of the foams resulted in the differences between the techniques. However, excellent results were obtained for the fiberglass. The significant advantage that the transient technique offered was that values of absorption and impedance for a large range of frequencies were obtained from a single experiment.

AUTOMATED STANDING WAVE RATIO APPARATUS.
Scientist, D.S.I.R.

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INTRODUCTION.
A method has been developed which allows the measurement of standing wave ratio (SWR) at several frequencies with a single pass of the microphone down the tube. Measurement and calculation are under desktop computer control (HP 9825 A).

DESCRIPTION OF STANDING WAVE APPARATUS.
The tube has an internal square cross-section of 400mm and total length of 11.5 m. Four 200mm loudspeakers are mounted in a baffle at 650mm from the source end. The walls are two thicknesses of high density particle board separated by a layer of softboard. The tube is terminated with a damped thick steel plate sealed against a neoprene gasket. Total weight is 935kg. The inside of the tube has three coats of polyurethane varnish sanded smooth.

An electret foil microphone is mounted on a boom supported on a light trolley of small cross-section area. A vane protrudes from the trolley into a small groove in the lower surface of the inner tube to guide the microphone. The microphone position is incremented via a thin stainless steel cable wound around a drum driven by a stepping motor through a 20:1 reduction gearbox.

METHOD OF MEASUREMENT.
Ten frequencies are nominated. The SWR is sampled at 1/100th wavelength intervals, 120 samples are taken at each frequency. A D/A converter supplies the appropriate dc voltage to a B&K 2010 heterodyne analyser which generates the frequencies. The microphone output is filtered at 3.16Hz bandwidth and a B&K 2427 digital voltmeter measures the filtered microphone output after stable output is obtained. At the same time the pen of a B&K graphic level recorder dots the chart which is synchronised with the microphone travel, building up a graph of the SWR's for visual records as the measurements proceed. The measured pressure amplitude and microphone position are stored in two 10 x 120 arrays.

ANALYSIS OF MEASUREMENTS.
The pressure maximum and the first two minima for each frequency generated are established by a curve fitting program held in the computer. The complex impedance, reflection coefficient, normal and statistical absorption coefficients are calculated according to Standard procedures and printed on the internal printer. Measurement time is about 80 minutes for ten frequencies sampled at 1/100th wavelength intervals and calculation time, 3 minutes.

The method developed is fast, accurate, and relieves the tedium of manual measurement. This is especially so where systematic measurements are made in development of anechoic wedges, and energy absorption above cut-off frequency is greater than 99%, i.e., the SWR is less than 1.74dB. Measured data and calculated results together with sample data, date etc., are stored in a file on a digital cartridge and can be recalled for further analysis, for instance data of several samples can be displayed as a family of 3 dimensional graphs on an X-Y plotter.
A DEVICE FOR MEASURING ACOUSTIC ENERGY IN A TRANSIENT SOUND FIELD.

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INTRODUCTION

Instrumentation for measuring the acoustic intensity at a point in a steady state sound field has been available for almost five decades. Within the last 3 or 4 years there has been a renewed interest in this area due primarily to the availability of digital processing techniques. The device described here has been developed for measuring acoustic energy at a point in a transient sound field. It has been used to identify and rank the more significant noise sources on a punch press.

THE DEVICE

The acoustic energy measuring device consisted of two closely spaced B & K microphones which were connected to a digital computer via a dual precision phase matched preamplifier and a high speed analogue to digital converter. Typically sampling rates were about 50 kHz. The data was stored on disc for processing. The two microphone signals allowed both the acoustic pressure and pressure gradient to be measured from which the particle velocity and energy could be calculated. A plot of the acoustic energy as a function of time was obtained. When used for noise source identification, the device was placed successively at a large number of positions close to the surface of the source. It was assumed that high energy at a particular position was indicative that the surface in the immediate neighbourhood was a significant noise source. The change of acoustic energy as a function of time also gave insight into the significance of the various noise sources.

SOME APPLICATIONS

The acoustic energy device was evaluated initially in the far field by using a swept sine wave radiated by a loud speaker in an anechoic chamber. It was found that with due care, good results could be obtained when sweeping from 300 Hz to 4000 Hz in 10 milliseconds. Consistent results were also obtained when the device was placed in the near field.

The device was also used for measuring the acoustic energy radiated from a punch press during a blanking operation. Typically the acoustic energy at the various positions close to the surface of the press varied by some 5 to 15 dB. There were considerable differences in the shape of the energy-time plots. The significant noise source regions were easily identified.

CONCLUDING COMMENTS

Tests to date have indicated that, with digital processing, it is possible to measure with reasonable accuracy the acoustic energy at a point in a transient sound field. The technique, when applied to a punch press, gave a clear insight into the variation of acoustic energy as a function of time at various positions and identified and ranked the more significant sources of noise.
ACOUSTIC IMPEDANCE MEASUREMENT OF HUMAN EAR

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LIST OF SYMBOLS

- \( Z \) acoustic impedance
- \( H(j\omega) \) transfer function
- \( p \) sound pressure
- \( Z_0 \) characteristic impedance of a pipe
- \( k \) wave number

Subscripts

- \( t \) terminating
- \( i \) input

INTRODUCTION

An equalizing network on a dummyhead for free-field reproduction with two loudspeakers in a network configuration has been discussed previously [1]. In order to improve prefiltering of a digital audio signal provided by a dummyhead and PCM recorder, a more precise acoustic impedance description of a human ear is needed. A modified impedance measurement method which allows the digitized input impedance over the required wide frequency range is discussed.

METHOD

The method is presented here for a typical case, let us consider the propagation of plane wave through the air confined in a rigid pipe with a small diameter, see Fig. 1. Input impedance \( Z_{in} \) and terminating impedance \( Z_t \) of the pipe are given by

\[
Z_n = Z_0 \frac{Z_C \cos k_\ell + jZ_S \sin k_\ell}{Z_C \cos k_\ell + jZ_S \sin k_\ell} \quad Z_t = Z_0 \frac{Z_C \cos k_\ell - jZ_S \sin k_\ell}{Z_C \cos k_\ell - jZ_S \sin k_\ell} \quad (1)
\]

If \( H(j\omega) \) is the transfer function defined by \( p_t/p_{in} \) of two points one of which is \( \ell \) distance away from the pipe-end, and the other \( \ell' \) distance away from it, then the terminating impedance \( Z_t \) is

\[
Z_t = Z_0 \frac{H(j\omega) \sin k_\ell - j\sin k_\ell}{\cos k_\ell - jH(j\omega) \cos k_\ell} \quad (2)
\]

Input acoustic impedance of an ear canal entrance can be measured by a equipment as shown in Fig. 2. Impulse responses of the free-field to point A1 and A2 are measured respectively and \( H(j\omega) \) is obtained from their Fourier transform representations. Eq.(1) and Eq.(2) give an input impedance \( Z_t \) at the ear canal entrance.

LEVEL RECORDER WITH OPTIMUM RESPONSE FOR SOUND LEVEL RECORDING

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This level recorder is being put to a wide range of sound level recording applications. According to IEC-Pub. 651 regarding sound level meters, the precision of a RMS detector & indicator is divided into a number of classifications according to the crest factor. It is also desirable that the writing responses of the level recorder correspond to the indicating responses of the sound level meter, and Rion has thus developed an exponential type level recorder that fulfills the requirements of IEC-Pub. 651. This exponential type level record has been made possible through major improvement of the writing responses of the conventional, widely used constant velocity response type level recorder.

Level Recorder Model LR-04 offers a wide frequency range and a dynamic range of 50 dB in this exponential response type unit. As the result of the “true RMS” circuit and the high-speed pen driving mechanism, the writing response approaches the limits of the theoretical response, and the indicated values closely approximate theoretical responses.

Model LR-50 makes use of a discharge destruction recording system that adopts a multi-pin head consisting of 120 pins, and this feature means that the writing response is practically the same as the response characteristics of the electrical circuits, thus being close to the theoretical response for such a unit.

The appended figures illustrate the writing response of these level recorders, and both units are light and compact and can be operated on battery power.
SOUND INTENSITY MEASUREMENT OF MACHINERY NOISE

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When sound power emitted from a stationary machine must be measured in its natural surroundings, i.e. in a factory hall with other noisy machines around, a means of measuring directivity has to be used in order to separate wanted and unwanted signals. Likewise, for multiple noise sources on a complex machine, the contribution of each noise source must also be analyzed and evaluated separately.

Such directivity characteristics can be obtained from the sound intensity measurement. Sound intensity is defined as the time averaged vector product of sound pressure and particle velocity. The particle velocity is difficult to measure directly, but the value can be derived from the pressure gradient value by integration. The pressure gradient can be measured by two pressure-sensitive microphones, the separation of which is small compared to the wavelength of the highest frequency of sound to be measured.

Major problems in this type of measurements are that calculations are based on the difference between two signals of almost equal level, and that phase matching of the two measuring channels is essential. When third octave spectrum measurements are required, the phase matching requirements get even more difficult to fulfil.

Ways to overcome the phase problem have been suggested by several authors.

For the measurement of steady noise sources, F.J. Fahy has suggested the use of sum and difference circuits such that only one filter is needed. This makes possible a very simple and inexpensive sound intensity meter.

Real-time measurements are possible by the use of two identical digital filters, and fast data processing permits measurement of non-steady noise sources, as well as fast localizing of sources.

Results of practical measurements will be discussed.
AN IMPULSE MEASUREMENT OF THE ACOUSTICAL IMPEDANCE OF THE EAR FOR CLINICAL USE

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The most commonly used audiological impedance meters make use of sinusoidal probe signals around 220 Hz and 660 Hz. The extension of the measurement frequency range, though very useful, would require too long measurement sessions, and this could be impractical from a clinical point of view. The pulse technique here applied allows the measurement time to be cut back, so overcoming the before mentioned difficulties, traditionnally associated with the multifrequental impedance measurements. By introducing this instrument into the clinical use new wide-spread information about the ear impedance could be obtained. It could likely become very useful for the diagnostic use and for the middle-ear model development as well.

The measurement set-up is shown in Fig. 1. The driver T, fed by the pulse generator PG, sends an impulse train which travels up to the ear and back to the microphone M through the tubes connecting the transducers to the ear cavity. The ear impedance affects very strongly the global transfer function and consequently the received impulse shape. By running two preliminary tests on known impedances, for example cavities of well defined dimensions, the whole set-up can be calibrated and subsequently the unknown impedances can be simply calculated on the ground of their impulse responses. This processing is accomplished by a computer connected by means of an A/D converter to the microphone output. To enhance the S/N ratio the impulse response is averaged over some tens of pulses. The resulting signal is then transformed through the FFT and processed to obtain the real and imaginary parts of the unknown impedance.

By repeating the impedance measurement for various static pressure values 3-dimensional diagrams are obtained similar to the one shown in Fig. 2. On the ground of simple considerations the external ear dimensions can also be evaluated from the multifrequental input impedance measurement. The acoustical impedance near to the eardrum can then be calculated, so eliminating the need of risky measurement techniques. The impedance, shown in Fig. 3, is referred to the plane 2 mm apart from the eardrum of the ear the input impedence of which is reported in Fig. 2. The characteristic ear resonances are particularly evident as well as the typical tympanometric behaviour in the low-frequency region.

![Measurement set-up](image)

![Acoustical impedance at the measurement plane](image)

![Acoustical impedance calculated 2 mm apart from the eardrum](image)
A PRACTICAL SOUND INTENSITY METER

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A portable sound intensity meter with useful frequency and dynamic range has been developed. The instrument, which incorporates a set of dBA, linear and 8 octave band filters, compares fairly well with standard sound pressure level meters as regards size, weight and ease of use. An accuracy of ±2 dB is achieved in the frequency range 40 Hz to 10 kHz and the dynamic range on the A-scale is from 30 to 130 dB. Operation of the instrument is based on the principle of obtaining pressure and particle velocity from measurements of sound pressure at two closely spaced points in the sound field and applying the difference-integral method.

Experimental development was guided by computer simulation of existing techniques with the object of optimizing the measurement system with regard to accuracy and reliability on the one hand, and simple and compact circuitry on the other hand. The main problems associated with the difference-integral method and variations of it, are connected with the extremely small phase difference between the two pressure signals at low frequencies. It is shown that commutation of the signals eliminate phase matching problems in a similar way as reported for the cross-spectral method. Calculations, confirmed by experimental observation, show that measurement error due to amplitude mismatch in a difference-integral system is surprisingly small.

Particle velocity calibration may be simplified considerably by electronic simulation of the microphone signals for a free progressive plane wave and a particular microphone spacing. A pistonphone is used in conjunction with an electronic calibrator to perform sound pressure and particle velocity calibration respectively while the latter instrument also provides the facility to test and to compensate for amplitude and phase mismatch.

BISPECTRAL PASSIVE VELOCIMITER OF MOVING NOISY MACHINE

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A new method of velocity determination of a moving noisy machine by the analysis of the signal detected at a fixed point is proposed. Bispectra of machine noises have significant peaks which are related with the fundamental and its harmonics. On the other hand surrounding noises can be regarded as almost Gaussian, hence with vanishing bispectrum.

These special features are used to realize new velocity measuring system which is not suffered from disturbing surrounding noises.

In this system it is assumed that the trajectory of the moving object is known beforehand and the nonstationary change of the bispectral characteristics along with the movement of the object is detected from a set of short time bispectra in segmented intervals and the method of minimum mean square error fitting is applied to get the trace curve of the change of the doppler shift. Then the velocity is derived from the form of the fitted curve.

The algorithm used in the system is shown in Fig. 1.

A result of the velocity estimation is shown in Fig. 2, where the object is moving along a circle with a fixed speed and fairly large surrounding noises is assumed.

Comparison with the result by the conventional means which uses the power spectral analysis shows clearly the effectiveness of the new method.

The details of the principle, construction and results will be presented at the meeting.

Mini- and microcomputers are useful in computer-based acoustical measurement systems, and in analysis and design. In the first role, instruments and analogue to digital converters are interfaced to give signal processing or measurement and control facilities. In the second role standard and specially developed procedures are required. Software development can be expensive and time consuming and also very personal so that the output of the programmer is often not shared by his co-workers. This paper describes a package that has been developed in an attempt to provide an easily usable set of subroutines for the common acoustical calculations and to facilitate the quick and easy addition of new routines.

The package is written in BASIC. At its heart is an executive program which defines a common data area containing the working arrays and variables. All the subroutines it calls communicate through the common area. The executive has a routine for handling four character codes keyed in by the operator to specify operations to be performed. Each code corresponds to a subroutine stored in a disk file which is loaded as an overlay and called by the executive program. Apart from the common area the storage requirements are modest as all subroutines are loaded as overlays.

To use existing routines the user keys in four character codes for the desired operations. Parameters are entered interactively. An example of a sequence of events in the design of a room might be:

**BAND** Specify a range of 1/1 or 1/3 octave centre frequencies
**VOLU** Interactive determination volume of room of complicated geometry
**AREA** Interactive specification of set of surfaces in a room (area and O)
**REVT** Calculate the reverberation time
**REVP** Plot a graph of RT vs frequency on graphics terminal
**SAVR** Save results in a disk file

Routines are called in a logical sequence and the results of a subroutine (eg AREA) are left in common to be used by a subsequent subroutine REV T. The user need not be familiar with programming details and the internal data representation. Available routines are listed in the full paper.

Writing of new subroutines is performed subject to closely defined ground rules and documentation standards. Typical subroutines are 10 to 50 statements and it has been the general experience that it is possible to get a new subroutine installed and debugged in about an hour. The effort required to add a new feature is thus minimal. Strict ground rules ensure that the new subroutine is documented and immediately useful to other users.

The author's suite has graphics routines based on a Tektronix 4010 terminal. Adaptation to other terminals will require editing the various graphics calls. The version of BASIC used on the NOVA 3 computer facilitates the addition of peripheral drivers and high speed signal analysis programs written in assembly language. This is a machine dependent feature but could be modified for implementation on other computer types.
EFFICIENT TESTS OF INTEGRATING-AVERAGING SOUND LEVEL METERS

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An integrating-averaging sound level meter includes an omnidirectional microphone, at least the A-frequency weighting, a squarer, an exposure integrator, a time clock and time integrator, logarithm takers, an adder, displays of average sound level and sound exposure level. (Exposure is the time integral of squared A-weighted sound pressure.) Efficient tests can be made by manipulation of the external controls and insertion of electric signals in series with a dummy microphone. The signals are steady sine waves and one-cycle sinebursts of the 31 preferred frequencies at 1/3 octave intervals from 10 Hz to 10 kHz. Theoretical sound exposure levels of the one-cycle sinebursts, for A-, C-, and Flat frequency weightings are the comparison standards in tests of transient responses. With a steady input, the constancy of cumulative average levels during exponentially increasing time periods is an easy clue to the consistency of the exposure integrator, the time integrator, and the logarithm takers; the time period at which a non-constant average level appears may be a clue to a defective component. The steady non-linearity test is performed by observing the 10-second or other average levels that result from inserting a steady sine wave at levels varied say at 1-dB steps; it is a steady test of the squarer and exposure integrator; the measuring range is that between the levels at which a stated non-linearity occurs. The dynamic non-linearity test is performed by observing the sound exposure levels that result from inserting pulses, such as a one-cycle sineburst of 125 Hz(8 ms), at levels varied say at 1-dB steps; it is a dynamic test of the squarer and exposure integrator; the top of the dynamic range is the steady sound level corresponding to the greatest sound exposure level reported correctly. The ability of the squarer and exposure integrator to handle pulses of different durations is tested efficiently by observing the sound exposure levels of the one-cycle, constant-amplitude sinebursts, 10 Hz to 10 kHz. Information about the exposure integrator and logarithm taker is obtained by observing the cumulative sound exposure levels of a steady sine wave over exponentially increasing time periods as long as 24 hours. The declining average sound level of an initial short signal (e.g. 10 seconds) during increasing time periods yields information about the time integrator and logarithm taker.
Impulse Duration: A new instrument for its measurement

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IMPULSE DURATION METER

An instrument is described which, when used with a Precision Impulse Sound Level Meter, will measure the durations of acoustic impulses in accordance with the Atherley & Martin and CHABA criteria for hearing damage risk to impulsive noise. The instrument is small, lightweight, can be battery powered and is designed for field use. The Sound Level Meter is operated in the "peak hold" mode to obtain the peak sound pressure level of the impulse and the Impulse Duration Meter is connected to the electrical output of the Sound Level Meter. For each acoustic impulse the Impulse Duration Meter simultaneously measures the impulse duration for both the Atherley & Martin criterion and the CHABA criterion and stores these values for subsequent presentation on the liquid crystal display of the instrument.

Comparison tests show that the Impulse Duration Meter gives more accurate and repeatable results than the oscilloscope trace photograph method or the digital waveform recorder method of impulse duration measurement.

By using the instrument with a suitable impulse source reverberation time measurements may be conducted with results comparable to those of existing methods.
Improved Array Performance in the Presence of Localized Nearfield Noise Sources
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INTRODUCTION

The reception of desired acoustic signals by arrays of hydrophones is often degraded by the simultaneous reception of interfering noise. Previously, investigators had developed array element shading techniques to reduce the interference of noise arriving from farfield locations. These techniques are not applicable, however, when the noise sources are in the nearfield of the array. In this paper a new shading technique is described that reduces the interference from localized nearfield noise sources while retaining good farfield receiving characteristics.

ANALYSIS

Consider a receiving array of $N$ omnidirectional, point hydrophones, a nearfield volume $V$ containing noise sources, and a frequency range $\Omega$ over which element shading coefficients are desired to reduce interference. The response of the array to a point source located at $\mathbf{r}$ and radiating at the angular frequency $\omega$ can be expressed as the standard sum of the responses of the $N$ hydrophones. The desired array response for point sources located within $V$ is idealized to be zero. The desired array response for farfield locations is characterized by a beam pattern $f(\theta, \phi)$ that contains a specified main lobe and is idealized to be zero outside the main lobe. Choosing the point source to be either in the nearfield volume $V$ or in the farfield of the array and setting the sum equal to the desired response results in a linear equation in the unknown element shading coefficients $\beta_n$, $n=1,2,\ldots,N$. A large number of values for $\mathbf{r}$ both in the nearfield volume $V$ and in the farfield together with frequency values from within $\Omega$ are chosen randomly until $V$, $\Omega$, and the farfield are well represented. A numerical solution of the resulting overdetermined set of simultaneous equations produces a set of shading coefficients that is optimum in the least squares sense for $V$, $\Omega$, and $f(\theta, \phi)$.

RESULTS

The technique was used to obtain shading coefficients for a line array of 20 elements with inter-element spacing $d$. The resulting shading coefficients provide greater than 30 dB reduction in interference (relative to a uniformly shaded line) from sources located within a nearfield volume whose dimensions are $2d$ on a side, even when the nearfield volume is located in front of the center of the array. The corresponding farfield beam pattern possesses lower sidelobes than that for the uniformly shaded array, although the main lobe is somewhat broadened. The theoretical predictions were confirmed experimentally by measurements made using a specially constructed line.
Is The Sound Power Defined by ISO/TC43 Independent of Specific Environmental Measurement Conditions?

HÜBNER

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INTRODUCTION

For acoustical planning and for comparison purposes the sound power output of a technical noise source - machinery or equipment - determined according a noise measurement standard should be independent from specific environmental conditions such as: (1) background noise, (2) sound reflection/absorption and (3) varied static pressure and temperature. On principle the relevant ISO documents (ISO 3741...3748) agree with this requirement and define several corrections. However from newer investigations two questions arise:

- Cover the 3 parameters mentioned above all significant influences?
- Fullfill the correcting equations given in the relevant ISO documents its purpose?

CORRECTIONS IN RESPECT TO VARIATIONS OF STATIC PRESSURE AND TEMPERATURE

We compare 3 different descriptions of sound power:

- \[ W_{\text{meas}} = \frac{1}{\langle P_{\text{act}} \rangle} \langle P_{\text{act}}^2 \rangle S \] respectively \[ L_{W_{\text{meas}}} = L_{P_{\text{act}}} + 10 \log \frac{S}{\text{lm}^2} \] (1)

- \[ W_{\text{ISO}} = W_{\text{meas}} \frac{\langle P_{\text{c}} \rangle}{\langle P_{\text{c}} \rangle} \] respectively \[ L_{W_{\text{ISO}}} = L_{P_{\text{act}}} + 10 \log \frac{S}{\text{lm}^2} + \Delta_1 \] (2)

- \[ W_{\text{norm}} = \frac{1}{\langle P_{\text{norm}} \rangle} \langle P_{\text{norm}}^2 \rangle S = W_{\text{meas}} \frac{\langle P_{\text{act}}^2 \rangle}{\langle P_{\text{norm}} \rangle} \] respectively \[ L_{W_{\text{norm}}} = L_{P_{\text{act}}} + 10 \log \frac{S}{\text{lm}^2} + \Delta_2 \] (3a)

\[ \Delta_1 = -10 \log \frac{P_{\text{c}}}{\langle P_{\text{c}} \rangle} \] \[ \Delta_2 = -20 \log \frac{P_{\text{c}}}{\langle P_{\text{c}} \rangle} \] (3b)

For sources radiating structure borne noise only: \[ \Delta_2 = -20 \log \frac{P_{\text{c}}}{\langle P_{\text{c}} \rangle} \] (3c)

The physical content of \( W_{\text{meas}} \) according Eq. (1) is the "uncorrected power" calculated from measurements carried out under specific conditions: static pressure \( B \) and temperature \( \Theta \).

\( W_{\text{ISO}} \) is the result of ISO-Standard procedures and has the meaning of the actual sound power under \( B, \Theta \) conditions. Therefore this value is not independent from specific meteorological conditions.

\( W_{\text{norm}} \) according Eq. (3) is a true normalized sound power and is related to normal conditions \( B = B_0 = 1000 \text{~ab} \text{er}, \Theta = 180 \text{~C} \).

For differences in static pressure of e.g. \( B/B_0 = 0.88 \), corresponding \( B \)-ratio for sea level and 1 km height, ISO formula (Eq.2a) requires a correction of \( \Delta_2 = 0.54 \text{~dB} \) whereas the true normalized sound power level requires corrections \( \Delta_2 \) between 1.09 dB up to 2.06 dB.

ON THE MEASUREMENT FOR INSTANTANEOUS SEPARATING OF PROGRESSIVE AND RETROGRESSIVE SPATIAL SOUND WAVES BY A CORRELATION METHOD

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The cross correlation between transmitter and receiver signals in the sound field was used for the equivalent spatial sound waves composed of progressive and retrogressive sound waves expressed in terms of the space-dependent function, by regarding delay time as real time and parameter of the waves. However, in the case of reflective sound field close to the reflector, spatial sound waves are measured as interference sound waves, and it is not able to instantaneously derive the individual spatial sound waves.

In this paper the author presents a measuring method of instantaneous separating of progressive and retrogressive spatial sound waves under the presence of reflector using Shift pulse width control M-sequence signal analogue correlator. Separating measurement is carried out by scanning of receiver distance from the transmitter with constant velocity, and simultaneously scanning of increasing or decreasing delay time, ratio of which is corresponding to the scanning velocity with sufficient resolution. By this operating, the phase of the sound wave to be eliminated is fixed constant and eliminated, therefore, the other sound wave to be measured is separated instantaneously, controlled in the travelling distance by initial delay time and compressed superficially in space coordinate.

Fig 1 shows progressive spatial sound waves $P_1(t_0, x)$ measured by means of increasing delay time and receiver distance x scanning. These waves travel backwards the transmitter and attenuate in amplitude with increasing of initial delay time $t_0$. Fig 2 shows retrogressive spatial sound waves $P_r(t_0, x)$ measured by means of decreasing delay time and receiver distance scanning. These waves travel towards the transmitter and attenuate in amplitude similarly. In this experiment, frequency is 2 KHz, scanning velocity of receiver is 0.217 cm/s, scanning delay time ratio is 6.24 μs/s, and distance of reflector from transmitter is 220 cm.
SOURCE SOUND RESTORATION IN MULTI-SOURCE ENVIRONMENT

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INTRODUCTION

A statistical method to restore source sounds in a multi-source environment is presented. The method is based on the generalized inverse(g-inv)[1,2] of matrices in the frequency domain. The g-inv is useful for obtaining an unique solution of a given set of equations with any number of unknowns. We have already proposed several applications of g-inv to acoustic signal processing[3,4,5].

FORMULATION OF SOURCE SOUND RESTORATION

Suppose there are $n$ point sources $S_j (j \in \mathbb{N} = \{1, 2, \ldots, n\})$ generating sound waves $s_j(t)$ in the free field. Let $f_i^*(t)$ ($i \in \mathbb{N} = \{1, 2, \ldots, m\}$) be the wave received by the $i$'th microphone $M_i$, and $d_{ij}$ be the distance between $M_i$ and $S_j$, and the following equation holds:

$$f_i^*(t) = \sum_{j=1}^{n} \frac{1}{d_{ij}} s_j(t - \frac{d_{ij}}{c}) \quad i \in \mathbb{N} \quad (1)$$

where $c$ denotes the sound velocity in the free field. Taking the Fourier transform of eq.(1), we have

$$F_i^*(\omega) = \sum_{j=1}^{n} \frac{d_{ij}}{d_{ij}} e^{-j\omega \frac{d_{ij}}{c}} S_j(\omega) \quad i \in \mathbb{N} \quad (2)$$

where $F_i^*(\omega) = \mathcal{F}[f_i^*(t)]$ and $S_j(\omega) = \mathcal{F}[s_j(t)]$.

Eq.(2) can be rewritten in matrix form as

$$F(\omega) = A(\omega) S(\omega), \quad \text{where } a_{ij} = e^{-j\omega \frac{d_{ij}}{c}}, \quad (3)$$

and its least-squares solution is

$$\hat{S}(\omega) = \hat{A}(\omega)^{+} F(\omega) \quad (4)$$

where $+$ denotes the Moore-Penrose g-inv. The time domain estimation can be obtained as the inverse Fourier transform of $\hat{S}(\omega)$. This method yields the optimal restoration of the source sounds in the least-squares sense.

RESULTS

This method achieves perfect restoration under the ideal conditions (free field, point sources, distances exactly known). However, it presents some difficulties in practical situations. Investigations of such difficulties and perceptual effects of the method are being carried out.

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Aircraft Noise Identification System by Correlation Technique

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One of the important measurements for noise countermeasures is the automatic long-term measurement of specific noise. There are wide applications for monitoring aircraft noise within every country where continuous long-term monitoring is necessary. One of the most essential problems in automatic monitor design is to discriminate aircraft noise from other offensive noise, in other words, to extract noise created by aircraft alone. Up to now this has been done by a mixture of aircraft take-off and landing schedules with artificial judgement.

This new Aircraft Noise Identification System is for extracting aircraft noise alone by discriminating the noise of aircraft in flight from other noises. As shown in Fig. 1 the system has two microphones fixed at a predetermined intervals. From the desired pattern of change in the mutual correlation coefficient, aircraft noise is discriminated from unwanted noise.

For example there is no difference in the arrival distance and arrival sound from automobiles and the like on the surface of the earth. As shown in Fig. 2 the best value of the correlation coefficient is always $\tau = 0$. However, as shown in Fig. 2 the arrival distance and arrival sound to the two microphones in the case of an aircraft in flight have differences. The greatest value of the correlation coefficient shifts and $\tau = 0$ becomes the smallest value. Also for airborne noise from sources not moving such as chimes and sirens there is no change in the correlation coefficient pattern. Because aircraft are moving the pattern changes with respect to time, and if this change is detected, aircraft noise can be discriminated from other unwanted noise. Also one application of this system is change from the greatest value of correlation coefficient of noise (aircraft noise) seen by the microphone on the ground is in proportion to the angle (angle of elevation). Accordingly, as this system is installed on either side of a flight course the angle of elevation can be determined from two locations at the same time discrimination is done. From these two elevations it is possible to project the aircraft position on the earth and calculate its altitude.

In this report we wish to introduce the wide uses and give actual examples of the application of the Aircraft Noise Identification System by Correlation Technique.

---

**Figure 1**

- Microphone 1
  - Characteristic Compensation
  - Correlation
  - Pattern Comparator
  - Noise Level Memory
- Measurement Diagram for New Aircraft Noise Measurement System

---

**Figure 2**

(Aircraft) (Automobile)

$\tau_1$

$\tau_2$ (At Time of Passing)

$\tau_3$

$\tau_4$

Cross-correlation function
Observation of Absorbing Materials by Acoustical Holography

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In this paper, we propose a new field measurement method of
the sound absorption coefficient of acoustic materials by
acoustical holography. The principle of the measurement is
based on recording three sheets of holograms as follows:

\[ H_A = H_1 + RH_2 \] : with the sample material to be under
test. \((R;\)Reflection coefficient of
the sample.\)

\[ H_B = H_1 + H_2 \] : with such a condition as the sample
space is replaced by rigid wall \((R=1).\)

\[ H_C = H_1 \] : with such a condition as the sample
space is replaced by the material
having \(R=0.\)

The absorption coefficient of the sample is given by

\[ \alpha_{h0} = 1 - |R|^2 = 1 - \left| \frac{H_{AC}}{H_{BC}} \right|^2, \]

where

\[ H_{AC} = H_A - H_C, \quad H_{BC} = H_B - H_C. \]

As well known, in pure tone experiments in a room, sound
fields are generally made complicated by interference.
Therefore, a sort of average technique should be demanded.
Fortunately, an average of data is calculated automatically
in space in the process of image reconstruction. In this
case, the dullness of resolution curves caused by long
wavelength holography is not a serious problem. It is
confirmed that our experimental results of normal incident
absorption coefficient agree fairly well with those of the
standing wave method by an acoustical tube.
DIRECTIVE HARMONIC WAVE DETECTOR BY THE USE OF LINEAR MICROPHONE ARRAY

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Various methods have so far been proposed to find out directions and spectra of sound waves from their sources. The methods were conventionally based, such as on the good use of a single microphone with super directivity, on the system of a moving microphone, and on the use of a multi-microphone system. In case of multimicrophone system the power spectrum and the directive power spectrum have been computed by means of the first Fourier transform for the spectral analysis.

Here we introduce a procedure that the monochromatic spectral analysis and the synthesis of directivities are able to be done simultaneously without computation of the fast Fourier transform by using a linear array of many microphones. The output of each microphone is multiplied by an appropriate weighting coefficient for beamforming. The multiplication is easily carried out using potentiometer. The sum of those products for all microphones is sampled with an adequate sampling frequency after passing through a low pass filter and then taken average for obtaining the power spectrum.

This method is considerably significant for detection of harmonic wave from specified direction by using only analog circuits in spite of involving several of frequency dependant variates, such as the weighting coefficient, the sampling frequency, and the cut off frequency of filter. The applications of the proposed method will be given to provision of noise control and the detection of underwater sound waves.
L'IMAGERIE ACOUSTIQUE ET SON APPLICATION DANS L'ENVIRONNEMENT

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Il arrive fréquemment, dans un environnement industriel complexe, de ne pas savoir apprécier la part respective de chaque installation, dans le bruit total que subit le voisinage. Le procédé d'imagerie acoustique a été conçu précisément pour remédier à cette impuissance.

Il permet, en un point donné, de repérer en fréquence et en direction les différentes sources sonores, responsables du bruit en ce point. Le panorama acoustique est ainsi décrit par une énumération des sources bruyantes et une estimation de leur importance relative, octave par octave.

Après avoir brièvement rappelé la méthode et décrit le matériel utilisé, on présente les résultats de quelques essais, ayant eu lieu au voisinage de centrales de production d'électricité.

SPECIFICATIONS TECHNIQUES

- 8 bandes d'octave : 63 à 8000 Hz.
- exploration horizontale avec ouverture de 180° (90° de chaque côté de la normale à la base de mesure).
- possibilités d'exploration dans un plan vertical (barre verticale ou horizontale).
- précision : 1°
- pouvoir de résolution (aptitude à séparer 2 sources de même fréquence) 5 à 10°.
  - résolution fréquentielle : 1/128 fréquence max. de l'octave.

APPLICATIONS

- Panorama acoustique d'un site géographique.
- Identification des différentes sources de bruit contribuant au bruit total régnant au point d'observation.
- Recomposition du bruit en ce point d'observation après suppression d'une source puissante (après insonorisation par exemple).
ANALYSIS OF NONLINEAR SYSTEMS USING WALSH FUNCTIONS

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Transmission properties of an electroacoustical system treated generally as a nonlinear system with memory are described by a set of multi-dimensional transfer functions \( \{ H_n \} \) [1, 3]:

\[ y(t) = \sum_{n=0}^{\infty} G_n[H_n, x(t)] \]  

(1)

where the functionals \( G_n[H_n, x(t)] \) are orthogonal and are defined by Wiener [1].

In the paper [3] it is assumed that the output \( y(t) \) for any input \( x(t) \) to an unknown nonlinear system may be expressed as formula (1), where \( \{ G_n \} \) are the functionals similar to Wiener’s except that instead of being expressed as convolution integrals, it is used the operator \( \otimes \) /addition modulo 2 without carry/.

It is shown that [3]:

\[ b_\nu = a_\nu c_\nu + \sum \sum \sum d_{ij} e_{ijl} + \cdots \]

where \( a, b \) are Walsh coefficients of input and output series, respectively, and \( a_\nu, d_{ij}, e_{ijl} \) are Walsh coefficients of transfer functions \( H_\nu, H_1, H_2, \ldots \).

Referring to relations given in the papers [2, 4] it becomes possible to evaluate Walsh coefficients applying the chain of transformations:

\[
\begin{align*}
\text{"arithmetic" autocorrelation function} & \quad T_{A-L} \quad \text{"logical" autocorrelation } \\
& \quad \text{function} \quad W \quad \text{Walsh power spectrum}
\end{align*}
\]

Thus, evaluation of Walsh coefficients of transfer functions \( H_1, H_2, H_3, \ldots \) necessitates the performance of the following steps:

1. Measurement of "arithmetic" autocorrelation functions of signals \( x(t) \) and \( y(t) \) : \( R_{xx}(\tau) \) and \( R_{yy}(\tau) \);
2. Transformation of "arithmetic" autocorrelation functions to "logical" autocorrelation functions:
   \[ L_{xx} = T_{A-L} R_{xx} \]
   \[ L_{yy} = T_{A-L} R_{yy} \]
3. Evaluation of Walsh transforms of \( L_{xx} \) and \( L_{yy} \);
4. Evaluation of Walsh coefficients of transfer functions \( H_1, H_2, H_3, \ldots \) according to the formulas given in the paper [3].

References

Les alternateurs des Centrales Electriques sont de plus en plus puissants et les coûts d'immobilisation forte sont considérables. Aussi est-il souhaitable de mettre en œuvre tous les moyens possibles pour prévenir d'éventuels incidents et localiser les organes défectueux.

Dans le cas d'un stator d'alternateur, les barres conductrices sont positionnées dans les encoches au moyen de cales : un alternateur de 900 MW comporte 2700 cales. Le décalage des barres entraîne des vibrations, responsables d'usure, pouvant créer des défauts d'isolement.

Pour vérifier le calage, le constructeur fait usuellement appel à des spécialistes expérimentés qui frappent chaque cale à l'aise d'une masse spéciale et déterminent à l'oreille, d'après le bruit émis, si l'état de calage est bon ou mauvais.

Il semble utile d'objectiver cette technique en se servant des méthodes de traitement du signal, et si possible de l'automatiser.

Pour cela, il est indispensable, en premier lieu de calibrer l'impact : on frappe la cale, au même endroit au moyen d'un marteau spécial équipé d'une jauge de force, ce qui permet de contrôler l'intensité de percussion.

En deuxième lieu, pour l'analyse proprement dite du bruit de choc, deux méthodes de traitement ont été mises au point :

- l'une consiste à représenter le spectre de fréquence du bruit de choc en 3 dimensions.

Les 3 variables sont : la fréquence, l'amplitude et la position d'une fenêtre temporelle glissant sur la durée du signal du bruit.

- l'autre méthode consiste à faire une analyse statistique des différents signaux de bruits de chocs représentés par un nombre donné de paramètres caractéristiques.

- Ces deux méthodes permettent de détecter les cales défaillantes, après une période "d'apprentissage".
A CONSTRUCTION OF A DUMMY-HEAD-LOUDSPEAKERS SYSTEM

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A modified version of an equalizing network on a dummyhead system for free-field reproduction with two loudspeakers, in a network configured as a bubble memory and a micro-computer controlled digital arithmetic system is presented in this paper. The equalizing network can be applied to an asymmetric condition of head-related transfer function, and is found to be particularly useful in the hardwear construction method or accurate simulation of an original sound field using a digital system. In order to test the validity of the equalizer, a prototype circuit is developed, utilizing analog devices which consist of a programmable active filter, a BBD system, and digital delay system. Then, a digital equalizer is developed in order to improve the precision of prefiltering and to allow easier control of the characteristics of the system. The 16-bit digital audio signal provided by PCM recorder at 44.0559 KHz sampling rate, is stored into the bubble memory whose capacity is about 10 sec 2Ch audio signal. Micro-computer controlled digital arithmetic system allows 1024 to 4092 points convolution sum between digital audio signal in the bubble memory and impulse responses of the equalizing network, and sends filtered audio signal to PCM recorder, as shown in Fig. 1.

Fig. 1. System structure of equalizing network utilizing a bubble memory and a micro-computer controlled digital arithmetic system.
ACOUSTIC DATA PROCESSING SYSTEM USING A MAGNETIC BUBBLE MEMORY

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An equalizing network on a dummyhead for free-field reproduction with two loudspeakers in a network configuration has been discussed previously [1]. In order to investigate such an equalizing network, a specially designed system is required for recording and reproducing acoustic data and also analyzing them with a computer. A acoustic data processing system was, therefore, developed in which magnetic bubble memory units with large storage capacity and reliability are utilized.

This system consists of a magnetic bubble memory (MBM), a PCM audio unit (PCM), a video tape recorder (VTR), a computer (CMP), and a X-Y plotter (XYP) (Fig. 1). Acoustic signals are recorded by VTR through PCM at 44.056 KHz sampling rate with 16-bit resolution and transferred to MBM with 32-bit length (16-bit x 2 channel). Then, digitalized data from MBM are fed to CMP. After computing and analyzing the data, CMP transfer them to MBM or XYP. Digital audio signal in MBM can be reproduced with PCM and VTR, if required.

MBM consisting of four magnetic bubble memory units (MBMU), IC buffer memory (BUF), and control circuit has 2 Mbyte storage capacity as a whole. MBMU have 80 chips each 64 Kbit. Memory structure in each chip is a conventional major-minor loop configuration. Access time and cycle time of MBMU is 5ms and 10ms respectively. MBMU transfer 128 words (one word is constructed 32 bits plus one parity bit) in each one cycle (Fig. 2).

Fig. 1 Structure of system  Fig. 2 Bubble memory system

EFFECT OF ASYMMETRY OF ACOUSTIC POWER FLUX FROM THE LFM-IDT* ON THE AUTO-CORRELATION FUNCTION
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When the SAW is excited by IDT on LiNbO$_3$, because the superposition of the original effect with the secondary effect caused by the acousto-correlation interaction, the acoustic power flux is asymmetrical. In this article, the computations of the transfer function at two ports of the single unapodized LFM IDT, and of the distortion of auto-correlation function of LFM-DL* introduced by the asymmetry of transfer function are reported. The main results are as follows:

1. When the impedance discontinuity is ignored, the acoustic power flux and amplitude of transfer function is asymmetrical. When the impedance discontinuity is considered, they all are asymmetrical. 2. The forms of the auto-correlation functions are different for considering and ignoring impedance discontinuity. The distortion of the side lobes of auto-correlation is asymmetrical and its increment depends on the fingers pairs of LFM ID. For larger fingers, it would increase above 5db.

* LFM IDT is means the linear frequency modulation interdigital transducer.
** DL is means dispertion delay line.
COMPARATIVE EFFECTIVENESS OF VARIOUS TIME DELAY PROCESSING TECHNIQUES

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Time delay is a basic measurement in diverse applications. Despite the varied applications, two basic types of time delay estimation problems are encountered. In one basic type at least one propagation path becomes scattered by an inhomogeneity in the channel, which leads to its intersection with another path at a spatial point. At that point, a single sensor receives the composite signal which may be processed by autocorrelation, cepstrum or complex demodulation to estimate the time delay parameter. In the other basic type, the propagation paths do not intersect at the sensing point. Then at least two spatially separated sensors are used to realize the diversity condition. Generalized cross-correlation techniques with a pre-detection filter by Hannan-Thomson, Eckart, Carter-Nuttall-Cable, Roth, or Hassab-Boucher may be applied to estimate the time delay parameter.

When implementing a time delay measuring system for either type of problem, a decision must be made as to the relative merits of the various alternatives (1-13).

In this regard, this paper explores two issues to determine the relative effectiveness of the processing techniques within each type of problem, and to identify the common areas within and across the two types of problems. As a result of studying the first issue, a matrix is presented that compares the performance of several variations to the processing techniques within each type of problem in the presence of various physical constraints i.e. signal to noise ratio, relative bandwidths, distortion, dispersion. As a result of studying the second issue, a shared hardware and software structure is developed for both types of problems. Since both types of problems may alternately or simultaneously occur in practice, obvious benefits are derived in treating them jointly. It turns out that much of the structure and concepts can be shared. Those include smoothing of spectral peaks, pre-detection filtering, time delay peak detection and tracking, peak search and gating, time delay filtering and data processing. The concepts have been implemented in a laboratory model on a PDP 11-70. Results and implications of those issues to the total measuring system will be presented. Finally, as noted in the introduction, the time delay parameter is not the ultimate quantity of interest but rather it is the source location, its motion, its identification, or the channel response, its distortion, its dispersion. Using the laboratory model, new results on three time delay application problems are reported.

INTRODUCTION

A computer based measuring system (1) has been used to determine the direction, bearing and relative intensity of individual reflections in an enclosure excited by a sound source emitting impulsive noise. The system samples the sound pressure at 125 equally spaced points in an imaginary cube and then uses time domain beamforming techniques to generate the focused data. However, simultaneous sampling of the microphone signals requires a large data acquisition system. This paper describes the design and use of a sparse volume array of microphones.

SPARSE ARRAYS

Sparse arrays (2) are formed by deleting certain redundant elements and substituting self- and cross-power terms from the remaining elements to preserve the beam pattern of the full array. Consider a general n-element full array with output signals $p_1, p_2, p_3, \ldots, p_n$. These signals have been delayed appropriately to focus the beam in a particular direction. The output power signal

$$P_{\text{full}} = \frac{(p_1 + p_2 + p_3 + \ldots + p_n)^2}{(p_1^2 + p_2^2 + p_3^2 + \ldots + p_n^2 + 2p_1p_2 + 2p_1p_3 + \ldots \text{ etc})}$$

is formed by adding these signals and then determining the mean square value. Alternatively the equation may be expanded to yield

$$P_{\text{full}} = p_1^2 + p_2^2 + p_3^2 + \ldots + p_n^2 + 2p_1p_2 + 2p_1p_3 + \ldots \text{ etc}$$

If the sound field is homogeneous the value of the self-power terms will be the identical. Cross-power terms with the same spacing and orientation are also equal and large reductions may therefore be made in the number of elements required. A computer study of a 5 x 5 x 5 equispaced volume array shows that only 46 of the 125 elements are essential. In the diagram below the 5 levels of the array are shown with "1" indicating a microphone.

| 1 1 1 1 0 | 1 1 1 1 0 | 1 1 0 0 0 | 1 1 0 0 0 | 1 1 0 0 0 |
| 0 1 0 0 0 | 1 0 0 0 0 | 0 0 0 0 0 | 0 0 0 0 0 | 0 0 0 0 1 |
| 0 1 1 0 0 | 1 1 0 0 0 | 1 0 0 0 0 | 0 0 0 0 0 | 1 0 0 0 1 |
| 0 0 1 1 0 | 1 1 0 0 0 | 1 0 0 0 0 | 1 0 0 0 0 | 1 1 0 0 1 |
| 1 1 0 0 0 | 1 1 1 0 0 | 1 1 0 0 0 | 1 1 0 0 0 | 0 1 1 0 1 |

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FREQUENCY SHIFTING OF AUDIO SIGNALS

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A number of sound-signal processing and architectural acoustics devices become realisable when low-distortion frequency shifting of sounds is achievable. These devices include a proposed system for artificial reverberation, ultrasonic delay lines, howlback suppressors for sound reinforcement systems, sound effects etc.

A number of methods of frequency shifting have been investigated and because of their apparent advantages economically and ease of variability two have been singled out for development and experimental verification. Both methods -

(i) the phasing method of single side-band generation

and (ii) a new method that we are proposing and refer to as the phase-increment method

require accurate and flat-frequency-response 90° shifted or 90° split program signals.

The work so far has been concerned with producing the optimum method for 90° phase-splitting wide bandwidth signals.

Details will be presented of the study of five different methods for 90° phase-splitting and their application in the two types of frequency shifter.
USE OF PULSE CODE MODULATION ENCODERS IN ACOUSTICAL MEASUREMENTS

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Signals encountered in acoustical measurements often have large crest factors, (e.g. impulses), or have fluctuating levels, (e.g. community noise). In instrumentation terms, acoustic pressure and its rms level have wide dynamic ranges. An ideal measurement system should maintain an acceptably small error over a large proportion of the dynamic range. In a digital measurement system the error at low signal levels will be due to quantisation noise when the signal does not occupy sufficient intervals. This paper describes the use of currently available electronic circuit modules in the measurement of equivalent sound level in the case of signals having substantial dynamic range.

The measurement system fig 1 consists essentially of such filters as may be required, an analogue to digital converter and a microprocessor. The converter is a standard type (μ 255) used for encoding signals in pulse code modulation systems. This has an amplitude compression characteristic built in and a dynamic range of 72 dB is covered with ± 128 intervals and 8 output bits which is ideally suited to simple microprocessor interfaces. Non uniform quantisation of amplitude samples is inherent in the system and intervals are approximately 0,5 dB wide. The signal is sampled at an adequate rate. As each sample is converted to digital form an integer, \( I_m \), signifying the interval into which the signal is classified is fed to the computer.

The program for calculating the mean square value of the system from the compressed and quantised sample values is based on the algorithm

\[
\overline{V^2} = \frac{1}{M} \sum_{m=1}^{M} S(|I_m|)
\]

\( I \) is defined above, \( M \) is the number of samples, \( S(r) \) is a stored table

\[
S(r) = \sum_{i=1}^{r} h_i V_{\frac{i}{2}} \quad i = 0 \ldots \ldots 127
\]

where \( h_i \) is the width of the \( i \)th interval and \( V_{\frac{i}{2}} \) is its mid value.

![Diagram](image)

FIG 1. MEASUREMENT SYSTEM

FIG 2.

(1) H E HANRAHAN, (1974) ICA 8 LONDON, 690
A NEW PROCEDURE FOR AUTOMATIC DETECTION OF UNDERWATER SOUND SIGNAL

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In this paper, we suggest a new method of CFAR processing and signal extraction for automatic detection in locally non-stationary background.

In the ordinary cell-averaging CFAR procedure, a mean value of background is estimated from the independent samples of background. In the procedure we suggest, an estimation of background fluctuation is obtained from the correlated samples of the background.

The correlated samples of the background is denoted by \( \{ x_i \} \). A new data set \( \{ z_i \} \) is formed from \( \{ x_i \} \) :

\[
z_i = [x_i - x_{i-1} + x_{i+1} - x_{i+2}]
\]

Then the estimation of background fluctuation is given by the following equation:

\[
\hat{\eta} = \frac{1}{N} \sum_{i=1}^{N} z_i
\]

The influences of the variation of the background mean value and the external signals on \( z \) can be effectively reduced by appropriately selecting the sampling period of \( \{ x_i \} \).

A signal \( x_j \) is declared to be present when

\[
x_j - x_{j-k} \geq T \hat{\eta} \quad \text{and} \quad x_{j+1} - x_{j+1+k} \geq T \hat{\eta}
\]

where \( T \) is the threshold, \( k \) is an integer corresponding to the signal duration.

By using this signal extracting procedure the false-alarm arising from the edges of locally strong background can be greatly reduced. Under the Gaussian assumptions the false-alarm and detection probability of this procedure can be computed from the following formulas:

\[
\begin{align*}
\mathcal{P} &= 1 - erf(-\frac{\bar{\eta}}{-\frac{\sigma}{\sqrt{2}}}) - \frac{1}{\sqrt{\pi}} \int_{\frac{\bar{\eta}}{-\frac{\sigma}{\sqrt{2}}}}^{\infty} e^{-x^2} dx \frac{1}{\sqrt{1 + \frac{\sigma^2}{\bar{\eta}^2}}} \frac{1}{\sqrt{1 + \frac{1}{\bar{\eta}^2} \frac{\sigma^2}{\bar{\eta}^2}}} \cdot dt \\
\gamma &= \left[ 1 + 2(1-p_s^*) \frac{T}{\bar{\eta}} \right]^{\frac{1}{2}} \left[ (SNR)^{\frac{1}{2}} - p_s^* \frac{T}{\bar{\eta}} \right] \\
\alpha &= \left[ \frac{\frac{\bar{\eta}}{\sigma}}{1 + \frac{\bar{\eta}}{\sigma} \frac{1}{\frac{\bar{\eta}}{\sigma}}} \right]^{\frac{1}{2}} \\
p_u &= \left( \gamma^2 - 2 \gamma (1 - p_s^*) / (\gamma - 1) \right) \left( \gamma - 1 \right) \left( \gamma - 2 \right) / \left( \gamma - 1 \right) \\
p_b &= 2 \left( \gamma - 1 \right) \left( \gamma - 2 \right) / \left( \gamma - 1 \right)
\end{align*}
\]

where \( \bar{\eta} \) - the normalized covariance between \( x_i \) and \( x_{i+1} \); \( \alpha \) - a parameter depending on \( \{ \eta \} \) and \( N \), usually \( \ll 1 \).

The results of theoretical analysis and computer simulation have shown that the performance of this procedure is superior to that of the ordinary cell-averaging CFAR processing in nonstationary background and they are the same in stationary situations.
SIGNAL PROCESSING TECHNIQUE FOR REDUCING TRACING DISTORTION IN TWO-DIMENSIONAL RECOPDING SYSTEMS

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Because of the possibility of high density recording and very short access time (less than 0.1 sec) for $10^4$-$10^5$ audio program sources of duration about 10 sec, a microfische system which records audio signals as two-dimensional patterns is thought to be suitable one. The two dimensional audio patterns are recoded on the fishe as pairs with corresponding image patterns.

Simulations are carried out on several candidates of signal processing method which transforms one-dimensional audio signals into two-dimensional patterns and is able to overcome serious distortion produced by mistracing of the two-dimensionally transformed audio patterns.

Among them, recording of pitch or noise and impulse response of vocal tract, extracted by PARCOR method, shown in Fig.1 is found to be wonderfully excellent and allows oblique mistracing over up to 16 lines of recorded two-dimensional signals without distortion for speech.

For general audio signals, especially music for typical example, the recording of successive amplitude and phase spectra shown in Fig.2 shows fairly good characteristics and allows oblique mistracing up to 8 lines with slight degradation of tonal quality. Simulation results will be heard by the audiences at the meeting.

Fig.1 Two-dimensionally Arranged Impulse Response of Vocal Tract.

Fig.2 Two-dimensionally Arranged Amplitude(left) and Phase(right) Spectra.
Sampling Statistics for Vibrating Rectangular Plates

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A significant part of the noise output of machines, such as electric generators, automobiles and ships is radiated by plates vibrating in flexure. It is often desired to estimate the radiated power from measurements of the vibrational amplitudes or accelerations on the plate surfaces. To make such estimates one must know or judge the relative contributions of travelling and standing waves to the measured values. Cases arise in which standing waves predominate, because the plated bodywork of the machine consists of plates stiffened by the attachment of ribs, beams, etc., which reflect part of the travelling wave energy. In the following we assume that all the vibrational energy is associated with standing waves, i.e., that the travelling wave component is nil.

For plates having a rectangular module the standing waves, i.e., modes, are of two different types, axial and tangential. For the axial modes, the amplitude of the bending waves varies sinusoidally in the x direction and is constant in the y direction; for the tangential modes it varies sinusoidally in both x and y directions. The probability density functions (pdfs) for sampling such axial and tangential modes are known, so that the precision of an estimate of the mean square value, based on a specified number of sample values, can be calculated.

Here an extension is made to the cases where two overlapping modes are excited at a given frequency. The corresponding pdfs are given by convolutions of the pdfs for each mode; however, as the latter involve the complete elliptic integral, the convolution integrals are intractable, and the required results are found from Monte Carlo calculations done on a computer.
"Application of Fast Chirp Waveforms to Signal Processing"
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ABSTRACT

A major area among the applications of surface acoustic waves has been the application to signal processing of linear chirp waveforms, i.e. waveforms where the carrier frequency varies linearly with time. Linear chirp waveforms have been employed in such applications as Fourier Transform generation, electronically variable delay lines and systems for arbitrary compression/dilation of signals in time, with or without time reversal, and no distortion of signal waveform.

It has recently been shown (1) that the implementation of these signal processing operations is much simpler, if chirp waveforms with fast rates of chirp are used. If the rate of chirp is not fast enough, higher order terms appear, which are proportional to powers of the rate of chirp $\mu$ and which in general introduce errors. The form of the error terms is such that in a straightforward application of fast chirp waveforms we are limited to narrow band signals. However, there is a number of applications in which some or all of the higher order terms either cause no error, or can be cancelled out by using an appropriate arrangement.

For example, in the case of the variable time delay device, if the time delay is suitably measured, the higher order terms produce no error at all and the arrangement can then be used as a very sensitive frequency discriminator. For Fourier Transformation, the terms containing odd powers of $\mu$ can be cancelled by appropriately averaging the outputs in a scheme, in which an up-chirp and a down-chirp with identical magnitudes of $\mu$ are produced. The Fourier Transformer is also capable of asynchronous operation, in that it will readily work for non-repetitive or randomly arriving signals.

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REFERENCE

MEASUREMENT OF TRANSFER CHARACTERISTICS OF ACOUSTIC SYSTEM WITH MULTIPLE PATHS

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This paper describes estimation methods of the transfer function of an acoustic system by using cross spectral technique especially for the system composed of multiple paths. In the cross spectral analysis, the time window for the discrete Fourier transform (DFT) should be longer than the reverbation time. On the other hand, the time window duration should be shorter than the propagation time difference of any two paths when the system is composed of two or more paths. The frequency resolution of transfer function is low when the system is composed of some paths having resonances. This paper presents the solutions of this problem applying residual spectra and cepstra.

Fig.1 shows the block diagram of a system having two different paths where the transmission time of the second path is longer than that of the first path by d. The transfer characteristics of the combined system \( H_1 + H_2 \exp(-j\omega d) \) will be estimated using the time window of long duration. \( H_1 \) and \( H_2 \) can be separately estimated using the time window of duration d and the delay d. However, the frequency resolution is less than 1/d in this case, and the estimated transfer function is not reliable if d is short. The time window of longer duration can be used for the separate estimation of \( H_1 \) and \( H_2 \). The theoretical explanation and the experimental verification will be presented in the lecture. Figs. 2, 3, 4, 5 and 6 show the experimental results where \( d=4.5 \) ms. Fig.2 shows the combined transfer function. Fig.3 shows the \( H_1 \) measured individually. Fig.4 shows \( H_2 \) using the window of short duration where \( \omega=4.5 \) ms. Fig.5 shows that \( H_2 \) obtained by residual spectra, where \( \omega=32 \) ms. Fig.6 shows the \( H_2 \) estimated by complex cepstral technique using comb lifter of 25ms duration.
In modern signal detection, the signals with different FM characteristics are usually used for adapting various requirements. Technically a signal with arbitrary FM mode can be approached by a frequency stepping (FS) signal

\[ s(t) = \sum_{n=0}^{N-1} \text{rect} \left( \frac{t - T_n}{T_0} \right) \cos \left( 2\pi f_n (t - nT_0) + \phi_n \right) . \]

When \( N \gg 1 \), by changing the values and/or the arrangements of \( \{ f_n \} \), a variety of FM modes will be obtained, but the \( f_n \) and \( \phi_n \) are changed step by step rather than smoothly.

The detection performance of the FS signal has been studied in detail. For ACD (All Coherence Detection), it is decided by the ambiguity function \( x(\tau, \beta) \) which has been derived strictly and analysed for 4 typical FS signals (single, double, linear and random stepping). A new processor, FDMF (Post Detected Matching Processor), is introduced. In FDMF, the FS signal \( s(t) \) passes a set of parallel channels (narrow-band filter \( \{ f_n \} \) - envelope detector - lowpass filter - matching delay line), then the sum of the outputs is used for detection. The performance function \( y(\tau, \beta) \) of FDMF also has been obtained and discussed for 4 typical FS signals.

It will be seen that theoretically the processing gain of FDMF is less than that of ACD (in general, 2db) and the resolution in FDMF is about \( 1/N \) of that in ACD, but practically the detection performance of FDMF is more preferable since the \( y(\tau, \beta) \) is independent of \( \phi_n \) which may be varied stochastically.

A FS signal generator consisted of a number of digital shift register chains, a D/A converter and a linear v-f converter has been presented. By choosing a set of codes, it can generate the expected FS signal to approach a needed FM mode.
COMPARISON OF SIMULATED AND ANALYTICAL RESULTS OF THE EFFECTS OF RANDOM SHADING, PHASING, AND ELEMENTS FAILURES ON BEAM GAIN AND BEAMPATTERN OF A PLANAR ARRAY, NAVAL UNDERWATER SYSTEMS CENTER

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The effects of random shading, phasing and element failures on the beampattern and beam gain of a planar array are investigated via simulation and analysis. The simulated results are compared with theoretical value. The simulated results and analytical value indicate that there is a loss of beam gain and rise of sidelobe level due to random errors. The loss of beam gain and rise of sidelobe depend on the amount of random errors and element failures and location of element failures. The effect of element failures is more pronounced on beam gain and on deep sidelobe level compared to the effect of combined shading and phasing errors. The simulated results and theoretical value are compared and found to agree within a few (2-3) dB.
The use of spectrum analysis to diagnose problems with rotational or reciprocating machinery is well established. Time integration is often used to detect spectral line components in a noisy broadband background. However for non-stationary sources integration will often not produce any increase in signal detectability and often the frequency variation itself is of fundamental importance in diagnosing the problem. This paper discusses digital techniques for tracking weak spectral lines in random noise. In particular an optimum solution for randomly fluctuating (in frequency) lines is discussed together with a method to imbed constraints in the tracking algorithm.
PROCEDURES FOR AUTOMATIC PURE-TONE AUDIOMETRY

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The use of digital techniques has an obvious place in the development of audiological testing procedures. One area in which it has been applied is that of automatic pure-tone testing to establish hearing levels for individuals. Unfortunately the technology itself does not imply which is the most effective means by which to test an individual's hearing, particularly in pathological groups.

This paper discusses the difficulties of programming a minicomputer to make the same decisions that an experienced clinician makes in routine audimetric testing. Reliability data is presented for five normal and five hearing impaired subjects for a testing algorithm that attempts to simulate the decisions of the clinician. These results are contrasted with the performance of the same groups using a number of other techniques including a random interleaved staircase method, a Bekesy tracking procedure, an audiometric ROC curve derived from signal detection theory and the PEST procedure. The results of reliability of each of the procedures is discussed relative to the accuracy and cross validity for each of the techniques and the requirements of the test situation.
N. OTHER ACOUSTIC TOPICS
Acoustics is an interdisciplinary subject and, for this reason, it is often underestimated in high education teaching, especially in developing countries. But the suitable position of acoustics in higher education is of great importance for its prestige and prospective applications.

At the mathematical-physical faculties of universities, acoustics should be well established in the curriculae of physics; physical acoustics and, in particular, molecular and quantum acoustics should be lectured to the students of higher terms with taking into account the specific features of work in the research laboratory /some examples of curriculae at universities/.

At the faculties of electronics and the like in technical universities /generally at engineering faculties/, it is necessary to maintain the equilibrium between theory and applications to meet the specific requirements of a given country following from its developing stage. Even in less developed countries such needs exist and demand to be analyzed in all details. Orientation of researches in acoustics should be - to some extent - the function of these needs of the country although the fundamental researches of cognitive nature should be conducted as well.

The most typical problems of applied acoustics are:
1/ Noise control in large cities and industrial plants. Developing countries are now at stage of urbanization and building new industrial plants and they should avoid the errors committed in time of the dynamic expansion of industry in the highly developed countries /examples of some problems of noise control/.
2/ In many developing countries, too little attention is devoted to the quality control of products and the proper exploitation of technical equipment. In this field, valuable services can be rendered by ultrasonic methods of materials non-destructive testing /Examples of application of ultrasonic testing/.
3/ In view of easy and safe operation, ultrasonic diagnostic apparatus employed in medicine, and, particularly in ophthalmology and gynaecology can be prove to extremely useful in developing countries /examples of diagnostics/.

To implement these postulates, it necessary to expand - to much larger extent - the exchange of the current information gained results as well the exchange of scholars and technicians in acoustics between developing and developed countries.
The highly interdisciplinary nature of the science of sound usually occludes its being a branch of physics, notwithstanding the fact that its position in modern physics is somewhat dubious and mostly unjustified. The purpose of this paper is to explore with illustrative examples that from the unity of physics, the interaction of acoustics with other branches of physics etc., acoustics is as it has always been an important and indispensable branch of physics though sometimes hidden. The relation between macroscopic acoustics and microscopic physics, the mutual enrichment of acoustics and other branches of modern physics are discussed in some length.
Competence and reliability of testing laboratories has been for many years of concern to governments, commercial interests, industrial enterprises and consumers.

Since the early 1960’s, however, there has been considerable discussion and agreement at national and international levels on procedures that will lead to formal recognition of the competence of laboratories. Such procedures are described by the term "laboratory accreditation."

This rapid increase in interest in formal accreditation has been brought about by an increase in awareness of and concern for health and safety in the community and by other factors such as increases in:

- government regulations
- use of technical barriers to trade
- technical sophistication in purchasing specifications
- costs of testing.

This paper outlines the concept and development of the world's first laboratory accreditation program which is operated by National Association of Testing Authorities, Australia (NATA).

NATA provides Australia with a service which accredits laboratories undertaking testing in all scientific disciplines and areas of industry and technology. Its systems have been developed and refined over a period of thirty years to a point where all similar bodies now being established around the world (in countries such as Canada, Denmark, France, Korea, New Zealand, United Kingdom and United States) look to NATA for advice and assistance.

The paper also discusses international relationships in accreditation and the way in which accreditation of laboratories dovetails into the broader international technical scene.